ELEKTOR ELECTRONICS

THE INTERNATIONAL ELECTRONICS MAGAZINE
MAY 1993
UK £2.25

WORKBENCH POWER SUPPLY UNIT

Electromagnetic Compatibility

Making Sense of Measurements

FM Stereo Signal Generator

VHF/UHF Receiver

Philips Preamplifier

RDS Decoder

9 770268 451036 05

RADIO DATA SYSTEM (RDS) DECODER

This article describes a radio data system (RDS) decoder based on Motorola's MC68HC05E0 microprocessor. The present decoder boasts a number of features that were not available on an earlier design published in *Elektor Electronics*.

Based on Motorola Application Note AN460, by Peter Topping

MAIN SPECIFICATIONS

- Permanent display of PS name and time or date and time (depending on mode).
- Optional display of PI code and secondary RDS features (RT, PTY, PIN, MS, DI, TA, TP, MJD and EON) on demand, including the principal frequencies of up to 11 other networks.
- Full use of CT providing auto-setting, accurate clock with automatic date and summertime adjustment.
- Implemented as an alarm clock which can control the power to the radio and/or sound an alarm.
- Sleep timer.
- TA=TP=1 (traffic announcement taking place) output.

THE Radio Data System (RDS) adds digital data capability to the VHF FM broadcasts on band III (87.5 to 108 MHz), and is in use in most of Western Europe. The specification is defined in EBU technical document number 3244 (see Ref. 1). A description of the capabilities of RDS can be found in previous *Elektor Electronics* articles (Refs. 2 and 3). A brief recap is appropriate here.

To transmit the data, a subcarrier is added at 57 kHz, three times the stereo pilot tone. This subcarrier is amplitude-modulated with a bi-phase coded signal. The subcarrier itself is suppressed to avoid data-modulated cross-talk in phase-locked loop stereo decoders, and to maintain compatibility with the German ARI (Autofahrer Rundfunk Information) system, which uses the same subcarrier frequency. Information is sent in groups of four 26-bit blocks. Each group of 104 bits is one of several types containing different information. It is up to the broadcaster which features are transmitted as long as the specified format is adhered to, and PI, PTY and TP are included. Each group contains a different sub-set of the RDS features. A list of all currently defined RDS features is shown in Table 1.

The retrieval of data is carried out by demodulation hardware which generates clock and data signals that can be used by a microprocessor. Suitable devices that can perform this function include the SAF7579T, SAA6579, TDA7330 and LA2231. The *Elektor Electronics* RDS demodulator board (Ref. 2) is based on the SAF7579T. The SAA6579 is a similar device that integrates the filtering, and thus requires fewer external components (Ref. 4).

The block diagram of a typical application is shown in Fig. 1. The microprocessor, in this case a Motorola MC68HC05E0, decodes the RDS data using the clock and data signals from the demodulator, and sends selected data to dot-matrix modules.

The decoder incorporates an alarm clock which, if permanently powered, can be used to switch on the radio supplying the RDS data at the required alarm time. There is a second output intended to sound an alarm. This output is cancelled when any key is pressed, leaving the control output active. This could control the power supply of the radio, or only the audio stage. If an audio mute is used, RDS data can be updated even when the radio is 'off'. Alternatively, the decoder can be used to simply display RDS data with its power being supplied from the radio, and manually switched on and off.

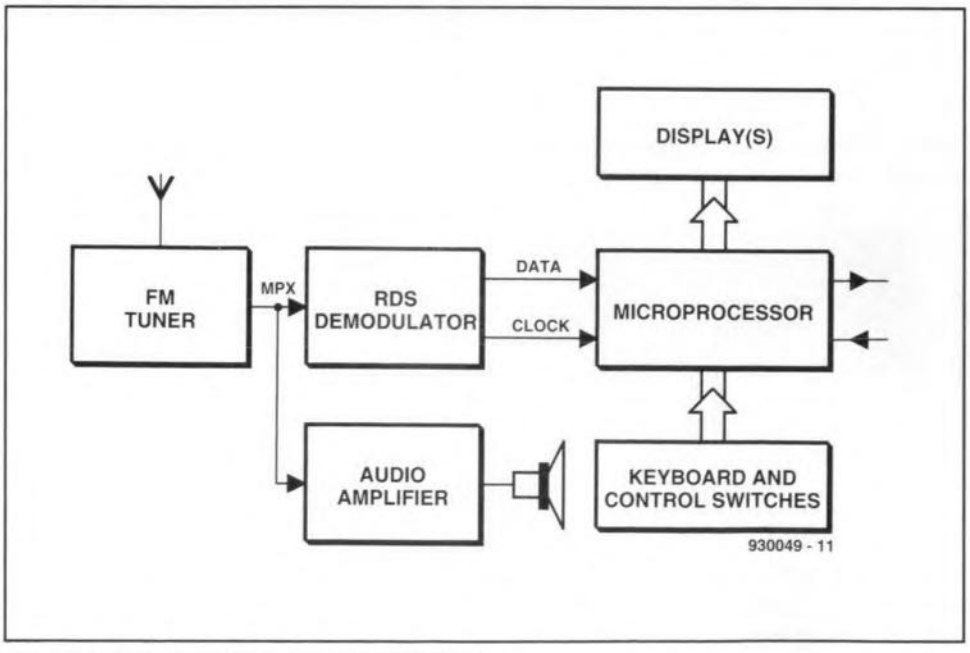


Fig. 1. Typical layout of an RDS receiver system.

Feature	Information
PI	Program identification
PTY	Program type
PS	Program service name
RT	Radiotext
CT	Clock time and date
AF	Alternative frequencies
TA	Traffic announcement
TP	Traffic program
MS	Music/speech switch
DI	Decoder identification
PIN	Programme item number
EON	Enhanced other networks
TDC	Transparent data channel
INH	In-house data

Table 1. RDS features.

RDS features supported

The present decoder supports PI, PTY, PS, RT, CT, TP, TA, MS, DI, PIN and EON. It facilitates permanent display of the 8-digit station name (PS) and time (CT) and, on request, can display program type (PTY), radiotext data (RT) and the status of the other RDS features. EON data can be displayed, but the retuning features associated with AF and EON are not supported as there is no capability to control the tuned frequency. In a car radio, EON data is used to switch the radio to a station which is broadcasting local traffic information, while AF data is used to tune the radio to the strongest signal carrying the selected device.

Program identification (PI) is a two-byte number which identifies the country, coverage area and service. It can be used by the control microprocessor, but is normally intended for display. A change in PI code causes the initialization of all RDS data as it indicates that the radio has been detuned. This decoder facilitates the display of the current PI code on request.

Program type (PTY) is a 5-bit number which indicates the type of program being broadcast. At present, 16 of these types are defined. Examples include 'no programme type', 'current affairs' and 'pop music', although the actual syntax which is displayed is determined by the software of the controlling microprocessor. In this example, PTY can be displayed on request — Table 2 shows the display used for each PTY code.

Program Service Name (PS) is the eight character name of the station, and is permanently displayed (except in standby mode).

Radiotext (RT) constitutes a string of up to 64 characters which give additional information regarding the service or programme being transmitted. In this application, RT is displayed on request on the 16-digit dot matrix using scrolling. The data often contains extra spaces to centre the text on a 2×32 character display. As these are not appropriate for a 16-character scrolling display, the software reduces all sequences of two or more spaces to a single space.

Clock Time (CT) data is transmitted every minute on the minute, and provides a very accurate clock, traceable to national standards. The (modified Julian) date and local time variation is also transmitted. Time is permanently displayed. In standby mode (see below), the date is displayed instead of the PS name. The MJD number, which is the form in which the date is transmitted, can also be displayed. The microprocessor converts this number into day-of-week, day-of-month, month-of-year.

Alternative Frequencies (AF) would be used by a car radio to retune to the strongest signal carrying the selected service. AF data, along with TDC and INH, is not used in this application.

Traffic Announcement (TA) and Traffic program (TP) are flags. TP is set if the transmitter normally carries traffic information, and TA is set if a traffic announcement is in progress. The combination TA=1 and TP=0 is used to indicate that EON data is being used to supply information to other networks, including traffic announcements. The status of these flags can be displayed, and the combination TA=TP=1 is brought out to a pin, and can be used to control an LED or external hardware. This could demute the radio, or switch from cassette when a traffic announcement is taking place.

Music-Speech (MS) is a single bit indicating either music or speech, and is intended to be used to make a tone or volume adjustment to a radio's audio stage. The MS bit is displayed on request.

Decoder Information (DI) constitutes four bits indicating the type of transmission (mono, stereo, binaural, etc.). It is not essentially in use in the UK, but can be displayed as a number between 0 and 15.

The Programme Item Number (PIN) is used to identify the programme currently being broadcast. The format is a 2-byte number which includes the scheduled time and date (day-of-month) of the start of the programme. It can be displayed as four hexadecimal digits, or fully decoded to day-of-month and time. Once it is fully implemented, PIN information will be able to facilitate automatic switch-on or recording of a pre-selected programme when it actually starts, even if this is not at the scheduled time.

Enhanced Other Networks (EON) replaces the older Other Network (ON) format. If type-14 groups are used to provide EON data, then type-3 groups (ON) will not be used (Table 5 shows the currently defined group types). Type-14A groups are used to send information about other networks. The PS name and principal frequency of up to 11 other networks can be displayed on request. Type-14B groups are used to switch to traffic announcements on other stations in a radio in which the microprocessor can control the tuned frequency.

Circuit

Figure 2 shows the circuit diagram. As different demodulator devices can be used, the circuitry for the demodulator is not shown. The complete RDS decoder/clock can be built using the decoder described here, the *Elektor Electronics* demodulator board (Ref. 2), and a power supply. The circuit of the decoder is very simple as the MC68HC05E0 uses a non-multiplexed bus, and includes its own chip selects. The only other chip required is thus the EPROM, an 8-KByte 27C64 of which 4½ K is used.

In order to facilitate a choice of display technology, the decoder drives both a parallel LCD module (based on a HD44780 driver, with or without an HD44100) and a serial VFD module (based on an MSC7128 driver). The displays show the same data (within the limitations of their character ROMs). Either or both modules can be connected.

LCD modules using only the HD44780 use divide-by-16 multiplexing. The software is written for this type of display, and will also work with modules incorporating the additional HD44100. Modules with both chips are capable of higher contrast, by employing divide-by-8 multiplexing. To use this capability the data at EPROM address $1FF3 should be changed from $38 (÷16 multiplexing) to $30 (÷8 multiplexing). Software modified in this way will not operate correctly on a display which does not include an HD44100. If the LCD module is not connected, a pull-down resistor should be connected to bit 7 of port C. This bit is read by the microprocessor to check that the controller in the module is ready to receive a command, and may cause the software to hang up if it is left open-circuit.

The serially driven VFD module shows the same data as the LCD module. The display driver used has a different character set from the standard ASCII set used by the LCD module,

PTY	Display
0	no programme type
1	News
2	Current affairs
3	Information
4	Sport
5	Education
6	Drama
7	Culture
8	Science
9	Varied
10	Pop music
11	Rock music
12	Easy listening
13	Light classics
14	Serious classics
15	Other music
16-31	No programme type

Table 2. PTY types.

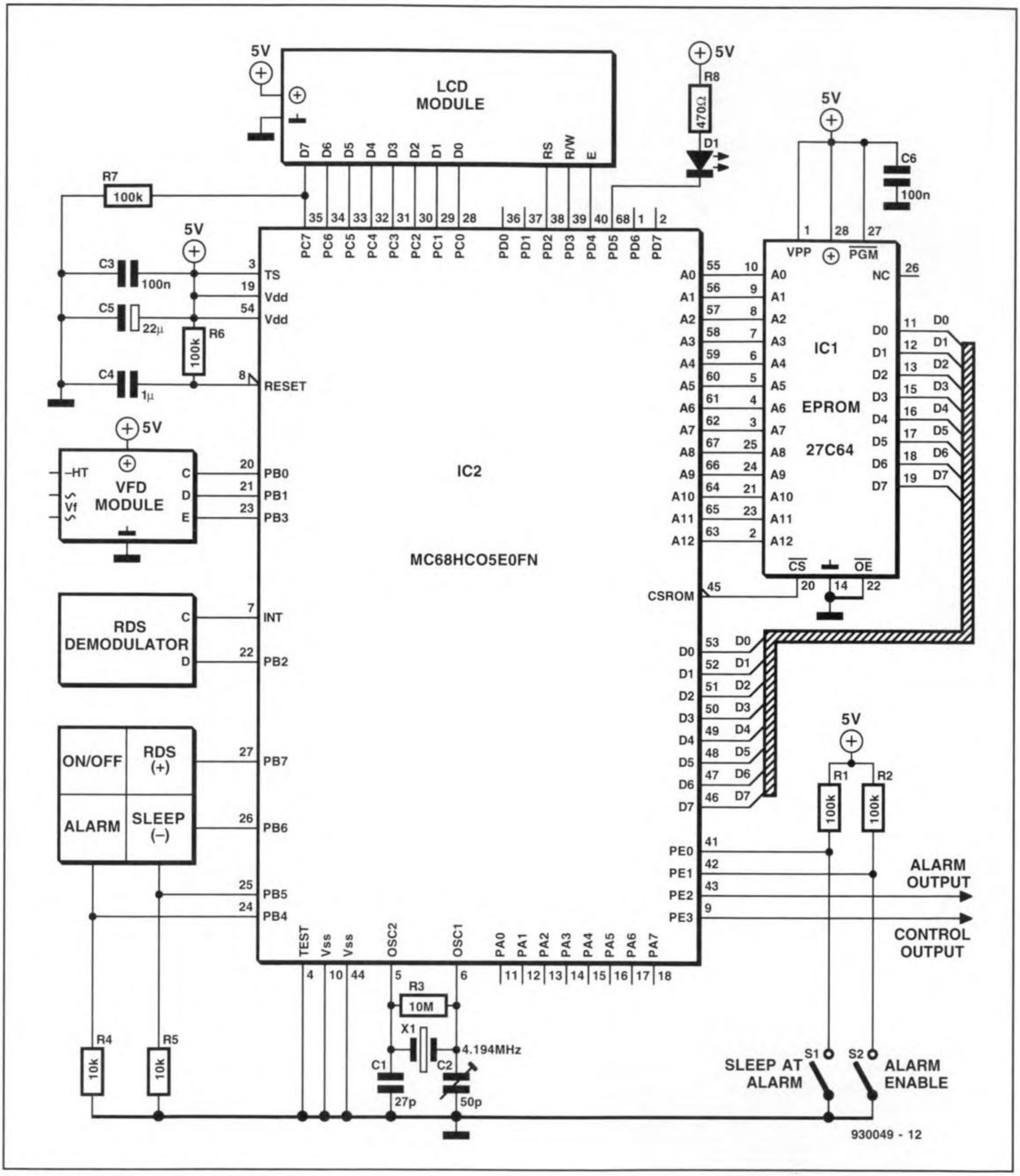


Fig. 2. Circuit diagram of the RDS decoder.

and a table is used to convert ASCII data into the required characters on the VFD module.

The only other components required are a crystal, the 4-key keyboard and a few passive components. Correct operation of the clock in the absence of an RDS signal requires that a 4.194-MHz crystal be used (the trimmer on pin 6 should be adjusted for accurate timekeeping).

Principle of operation

On power-up, the software initializes the display modules (the displays show 'Mon 0 inv 00:00' until a valid group 4A is received) and an idle loop regularly checks the local keyboard for a key press, compares the current time with the alarm time, and performs other time-dependent functions related to the display modules and the sleep timer.

The keyboard software scans the 4-key matrix for a key press every 16 ms. If the same key is pressed on three successive scans, it acts on this key function by calling the relevant subroutine. These routines also control the repeat rate of the SLEEP and RDS keys. This rate is set at 6 Hz (after an

<table>
<tr><th rowspan="2">MODE</th><th colspan="4">KEY</th></tr>
<tr><th>On/Off</th><th>Sleep</th><th>Alarm</th><th>RDS</th></tr>
<tr><td>Standby (Off)</td><td>mode normal (On)</td><td rowspan="4">mode sleep (On)</td><td rowspan="2">mode alarm</td><td>—</td></tr>
<tr><td>Normal (On)</td><td rowspan="2">mode stndby (Off)</td><td rowspan="3">RT
PTY
PI
TA/TP
PIN(h)
PIN(d)
MJD
MS/DI
EON 1
:
:
EON 11</td></tr>
<tr><td>Alarm OFF</td><td>mode alarm ON</td></tr>
<tr><td>Alarm ON</td><td>mode alarm set-up</td><td rowspan="2">mode alarm OFF</td></tr>
<tr><td>Alarm SET UP</td><td>toggle hr/min</td><td>dec. hr/min</td><td>inc. hr/min</td></tr>
</table>

Table 3. Key functions.

initial 750-ms delay) when the keys are used to change the alarm time, and 1 Hz for their normal function. The other keys do not repeat if held down. Table 4 shows the various functions available in each mode.

The On/Off key toggles between ON and standby modes. A port pin (3, Port E) can be used to control the power to the VHF radio and/or other external hardware. It is active high (low in stand-by). In stand-by mode, with the alarm disabled, the time and date are displayed. If the alarm is enabled, the alarm time is displayed along with the current RDS PS-name. Table 5 shows these display formats.

The Alarm key facilitates the display of the current alarm status. A second press changes the alarm armed status.

Display mode		Format
Standby (Off)	Alarm off	Thu 30 Apr 18:05
	Alarm off, no CT	Mon 0 inv 0:00
	Alarm on	0659 ALARM 18:05
Normal (On)	With RDS PS name	BBC R4 18:05
	Without RDS	-------- 18:05
Alarm	Alarm off	Alarm - OFF
	Alarm on	Alarm - 6:59
Sleep		Sleep 60 min.
RDS	RT	BBC Radio 4 ....
	PTY	News
	PI	PI code - C204
	TA & TP	TP - 0 TA - 1
	PIN(hex)	PIN no. - F480
	PIN(decod)	30th at 18:00
	MJD	MJ day - 48742
	MS & DI	M/S M DI 15
	EON 1	BBC R3 92.10
	2	BBC R.Sc 103.60
	3	BBC Nwcl 96.00
	4	BBC Scot 94.30
	5	BBC Mtme 92.50
	6	BBC Twed 93.50
	7	BBC R5 909kHz
	8	BBC Eng. 100.00
	9	BBC R1 99.50
	10	BBC R2 89.90
	11	--------

Table 4. Display formats.

When armed, the alarm time is displayed. In this mode, the On/Off key can be used to select either hours or minutes (indicated by flashing), and the Sleep and RDS keys to increase and decrease the settings. If the alarm has triggered, the first press of any key cancels it. The alarm display has one of the two alarm formats shown in Table 4, according to whether or not the alarm is armed. As all the keys have a special function in the alarm set-up mode, the only way to exit this mode is to wait for a time-out. If no keys are pressed for five seconds, the mode returns to normal.

The Sleep key controls the sleep timer. If the decoder is in the standby mode, the first press of SLEEP switches it on, and initializes the sleep time to 60 minutes. When the sleep timer is running, this is indicated by a flashing decimal point in the right-most character of the display modules. Subsequent presses of the SLEEP key decrease the time remaining by 5 minutes. When the sleep time has elapsed, the decoder returns to stand-by. In the alarm set-up mode, this key decreases the alarm time.

The RDS key steps through the various RDS data displays. Holding down this key steps through the displays at 1 Hz. The displays are RT (scrolling), PTY, PI, TA/TP, PIN (hex), PIN (decoded), MJD, MS/DI and EON (11 networks) as shown in Table 4. In the alarm set-up mode, this key increments the alarm time.

Alarm functions

The alarm time can be entered as described above. If the alarm is enabled (alarm time displayed on the first press of the ALARM key, and permanently displayed in stand-by mode) then, at the alarm time, the auxiliary control line will go high. This can be used to control external hardware, for example, to switch on the VHF radio supplying the RDS data. If the auxiliary line is already high (decoder fully on, or on via the sleep timer), then it stays high. The operation of the sleep timer is not affected if bit 0 of Port E is high. If this I/O line is low at the alarm time, the sleep timer is actuated for an hour. This takes place whether the decoder was previously on, off, or running the sleep timer, and has the effect of switching the auxiliary line low an hour after the alarm time, regardless of its condition prior to the alarm.

At the alarm time, the alarm output will also be actuated (active low) as long as it is enabled by bit 1 of port E being held low. This is intended to drive an alarm sounder. When this output is active, a press of any key cancels it until the next alarm. This cancellation does not affect the auxiliary output.

Groups handled

If a complete group has been received, the data can be processed. The PI code is checked to see if it has changed. If it has, the displays are initialized. In an application using the AF capability of RDS, more use would be made of the PI code. All RDS data, except date and time, is cleared if no valid RDS data is detected for a period of 10-seconds.

next, PTY and TP are updated, and the group type identified. Group types 0A, 0B, 1A, 1B, 2A, 4A, 14A and 15B are handled. Table 6 shows the type of information contained in each group, and Table 7 shows the detailed structure of these groups.

Group	Features
All	PI, PTY, TP
0	TA, DI, MS, PS, AF
1	PIN
2	RT
3	ON (replaced by EON)
4A	CT
5	TDC
6	INH
14	EON
15B	TA, DI, MS

Table 5. RDS groups

Groups 0 and 15B

As AF data is not handled, there is no difference in the treatment of groups 0A and 0B. PS data is extracted and placed in RAM according to the address bits in block 2 (see Table 6). TA, DI and MS data are then read, DI is

	Block 1		Block 2		Block 3		Block 4	
Group 0 and 15B	PI code	chck A	bit(s) use 15–12 : group no. 11 : group type 10 : TP flag 9–5 : PTY code 4 : TA flag 3 : M/S bit 2 : DI bit 1–0 : PS/DI address	chck B	AF (PI code in type 0B and 15B)	chck C or C'	PS name (as block 2 for 15B)	chck D
Group 1	PI code	chck A	15–12 : 0001 11 : group type 10 : TP flag 9–5 : PTY code 4–0 : not used	chck B	not used (PI code in type 1B)	chck C or C'	PIN data 15–11 : day-of-month 10–6 : hour 5–0 : minute	chck D
Group 2A	PI code	chck A	15–12 : 0010 11 : 0 10 : TP flag 9–5 : PTY code 4 : text A/B flag 3–0 : text address	chck B	RT 2 ASCII characters	chck C	RT 2 ASCII characters	chck D
Group 4A	PI code	chck A	15–12 : 0100 11 : 0 10 : TP flag 9–5 : PTY code 4–2 : not used 1–0 : MJD (16–15)	chck B	CT 15–1 : MJD (14–0) 0 : hour (4)	chck C	CT 15–12 : hour (3–0) 11–6 : minute (5–0) 5 : offset sense 4–0 : offset (4–0)	chck D
Group 14A	PI code	chck A	15–12 : 1110 11 : 0 10 : TP flag 9–5 : PTY code 4 : TP (On) flag 3–0 : usage code	chck B	EON information code: 0–3 : PS 4 : AF 5–9 : AF (map) 10–11 : not used 12–15 : not imp.	chck C	PI (On)	chck D

Table 6. Detailed structure of RDS groups handled.

sent a single bit at a time and uses the same address bits as the PS name to determine which of the four bits is being updated. Groups of type 15B also contain all this switching information. They are used to increase the repetition rate of this data, but contain no PS or AF information.

Group 1

Group types 1A and 1B are again treated identically as they contain the same data except for the repetition of the PI code in type 1B. The PIN data is recovered and saved in RAM.

At present the decoder simply allows the display of PIN data both in its raw hexadecimal form, and fully decoded to day-of-month and time. Full use of PIN data would require continuously comparing the PIN day-of-month and time with the current day-of-month and time, and switching on external hardware (radio and/or cassette recorder) when there is a match.

Group 2A

RT data from blocks 3 and 4 is written to RAM according to the address included in block 2. There are four address bits and four ASCII encoded bytes, giving the possibility of 64 characters. If the Text A/B flag changes state, the RT area in RAM is cleared as this indicates that the message has changed. Group 2B is not handled as it is rarely if ever used.

Group 4A

Two of the more complex tasks to be performed are required by the CT calculations for group 4A. These are for the local time difference, and the conversion of the MJD number into a recognizable date.

The broadcast time is Universal Coordinated Time (UTC, effectively the same as GMT). Time differences from UTC, including summer (daylight saving) time, are sent as an offset of up to ±12 hours in half-hour increments.

The software includes 4-function 9-digit integral BCD arithmetic which is used to decode the date from the MJD number using the formulae:

Y' = int [(MJD-15078.2)/365.25]
M' = int [(MJD-14956.1-int(Y'×365.25))/30.6001]
Day = MJD–14956–int(Y'×365.25)–int(M'×30.6001)
If M'=14 or M'=15 then K=1; else K=0
Year = Y'+K
Month = M'–1–12K

Group 14A

This group contains EON data. A large amount of information can be sent using this group, and it can take up to two minutes for all the data to arrive after the radio has been retuned. This application saves the PI code, PS name and principal frequency of up to 11 networks, although more networks, each with many frequencies, and other data (e.g., PTY(ON), PIN(ON), TA(ON), etc., may be sent. Table 4 shows the format of the EON display. All the information shown in Table 4 is real data from the Black Hill transmitter in Central Scotland. ■

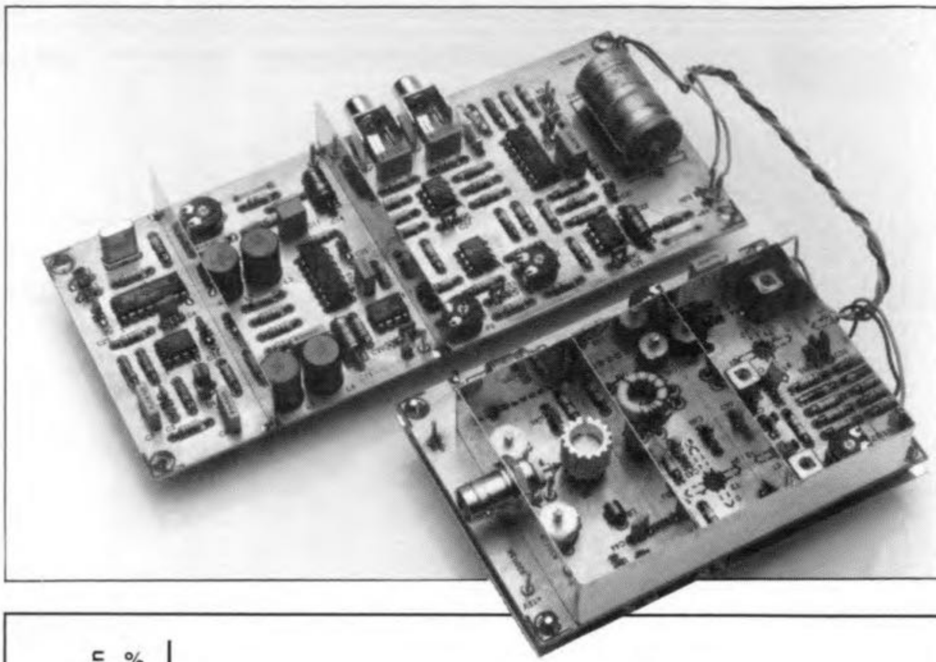
References:

1. EBU technical document 3244: Specifications of the Radio Data System, RDS, for VHF/FM Sound Broadcasting.
2. Radio Data System (RDS) demodulator. *Elektor Electronics* May 1989.
3. Radio Data System (RDS) decoder. *Elektor Electronics* February 1991.
4. RDS demodulator with integrated filter. *Elektor Electronics* October 1992.

Note: the assembly code listing of the control program contained in the MC68HC05E0 is printed in Application Note AN460/D, Available from Motorola Ltd., European Literature Centre, 88 Tanners Drive, Blakelands, Milton Keynes MK14 5BP.

FM STEREO SIGNAL GENERATOR

The construction project described here consists of two units: an FM stereo multiplex generator and a three-stage VHF FM exciter. Together, they form an FM stereo signal generator intended for testing and aligning FM stereo receivers.



Design by J. Barendrecht

GENERATING an FM stereo signal with a clean spectrum, and suitable for test purposes is not simple. However, it can be done with relatively simple means, as demonstrated here, provided you have some experience in working with high-frequency circuits. The signal generator described has an output frequency range that covers the VHF FM broadcast band (87-108 MHz), and supplies an output power of about 150 mW into a 50-Ω load. Owing to the absence of an output filter, harmonics are insufficiently suppressed to enable the generator to be connected to an antenna. Fortunately, that is not a problem for the main application of the signal generator: testing and aligning FM receivers, where a dummy load will be used.

The FM stereo multiplex signal

Figure 1 shows the theoretical frequency spectrum of the stereo multiplex (MPX) signal applied to the modulation input of a VHF FM broadcast transmitter. The two stereo signals L (left) and R (right) are added as well as subtracted to give the corresponding sum (L+R) and difference (L–R) components. The L+R component occupies the lowest part of the spectrum, up to about 15 kHz, and affords compatibility with mono receivers. The L–R component is converted into a double-sideband signal with a suppressed carrier at 38 kHz. This type of modulation is called DSCC (double-sideband, suppressed carrier). It causes an upper sideband (USB) and a lower sideband (LSB), mirrored against a suppressed ('invisible') carrier (here, 38 kHz). The carrier is suppressed to

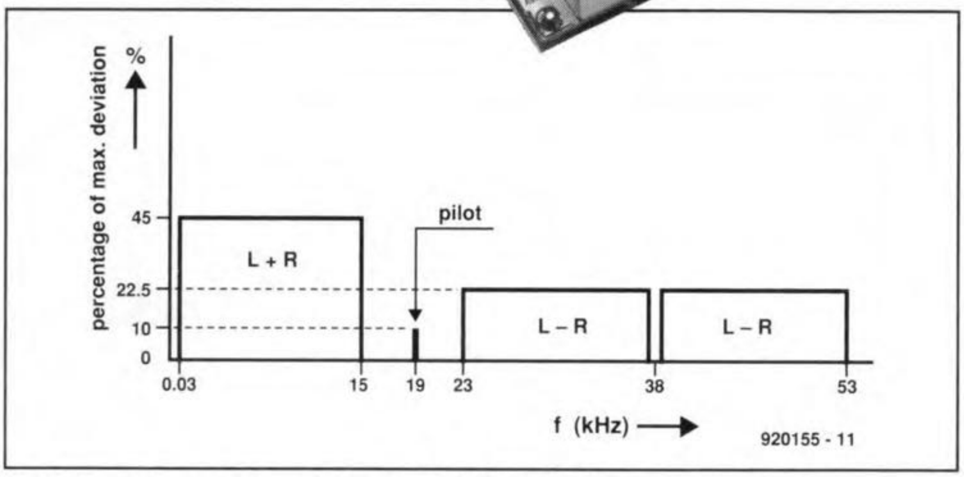


Fig. 1. Theoretical spectrum of the stereo MPX signal (EBU recommendation).

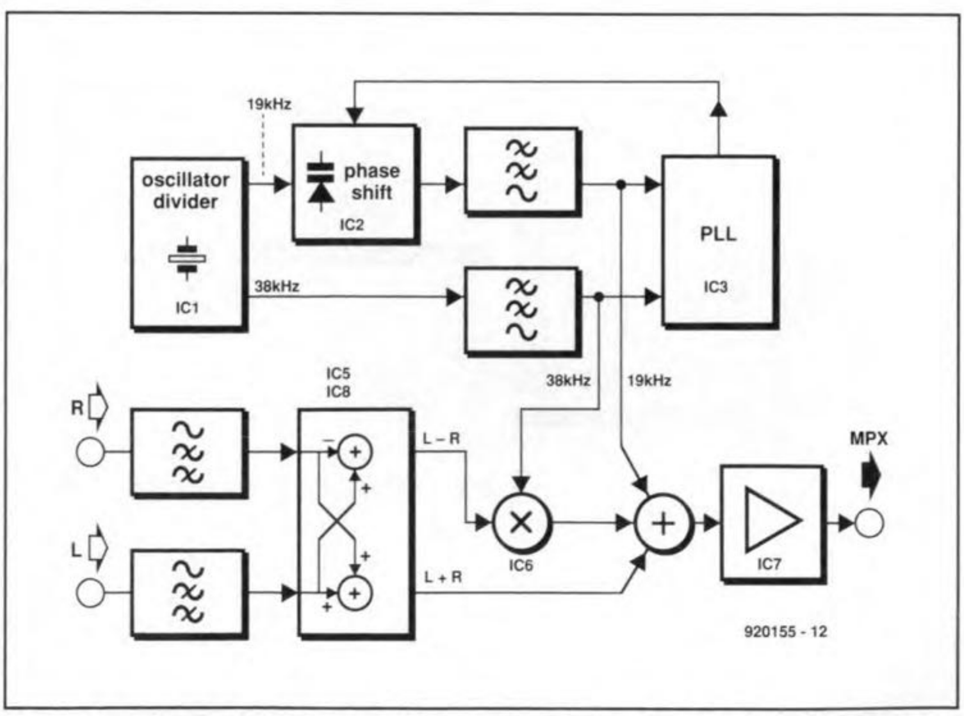


Fig. 2. Block diagram of the stereo MPX generator. A PLL is used to synchronize the 19-kHz (pilot) and 38-kHz signals.

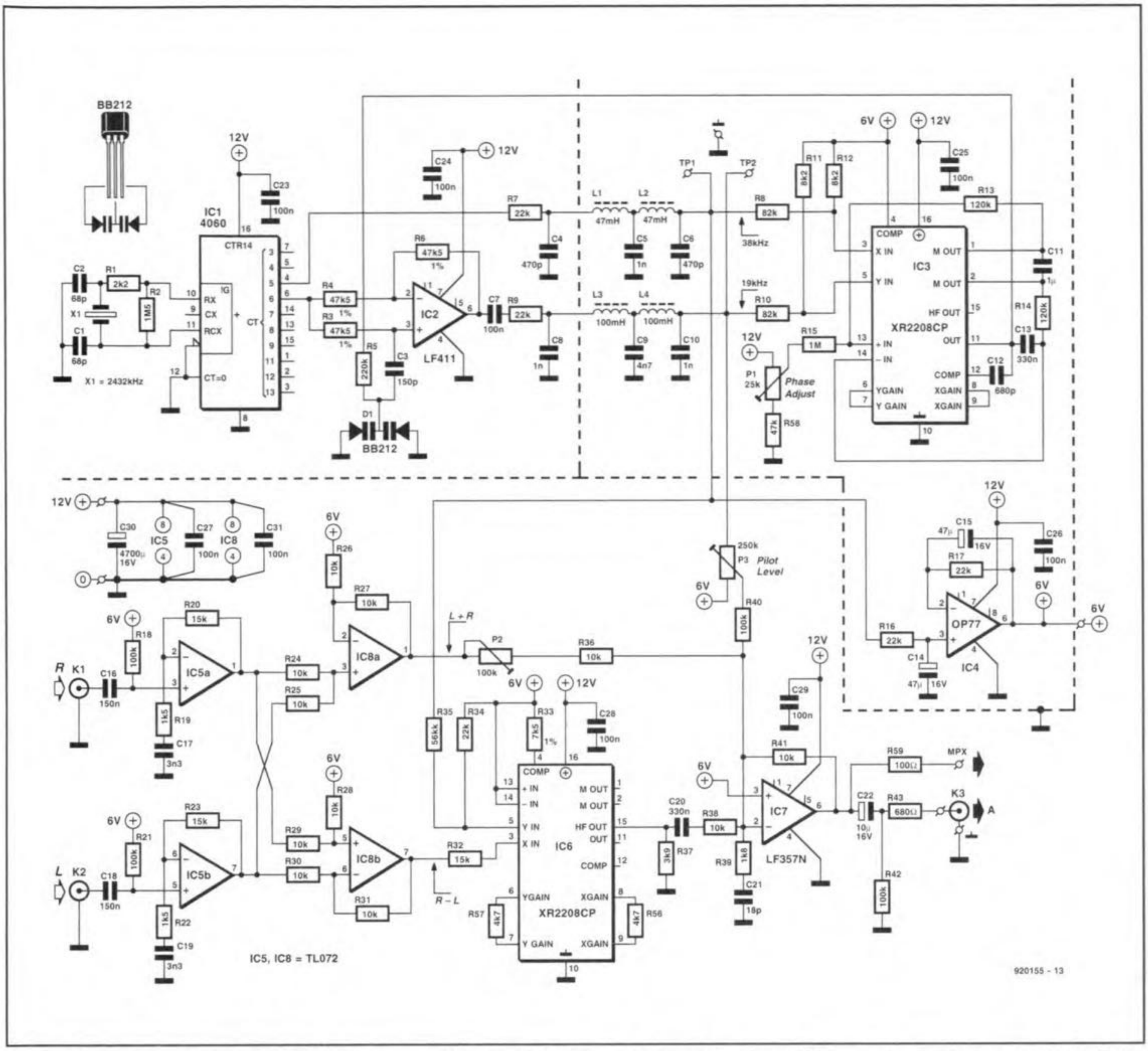


Fig. 3. Circuit diagram of the stereo MPX generator. The two XR2208s have different functions: IC3 is wired as a phase-locked loop (PLL), while IC6 works as an analogue multiplier.

keep the total deviation of the transmitter within limits.

At the receiver side, the 38-kHz carrier is recovered with the aid of the 19-kHz pilot tone contained in the MPX signal (relative level: 10%), which also serves to indicate a 'stereo' transmission. The 19-kHz pilot tone is doubled to give 38 KHz, and enables coherent demodulation of the L–R information. Next, a matrix is used to 'distill' the L and R signals from the components L+R and L–R.

Returning to the transmitter side, the levels of the components in the MPX signal are fixed to optimize the channel separation given the available bandwidth for the FM signal, and also to ensure that the sound quality on a mono receiver is not impaired.

MPX generator block diagram

The block diagram of the stereo MPX generator, or stereo coder, is given in Fig. 2. A quartz-controlled digital clock oscillator/divider supplies 19-kHz and 38-kHz signals to low-pass filters that turn the rectangular waves into sine-waves. In between the 19-kHz clock output and the input of the associated low-pass filter sits a phase shifting circuit controlled by a PLL (phase locked loop). The PLL compares the phase of the 19-kHz and the 38-kHz sine-waves, and supplies an error signal to the phase shifter ahead of the 19-kHz low-pass filter. Since the PLL will attempt to achieve an error signal of nought, the 19-kHz and 38-kHz sine-waves will have a fixed phase relationship.

The left and right audio signals are taken through pre-emphasis networks before they are summed and subtracted to give the L+R and L–R components. The L–R component is multiplied with the 38-kHz signal to give the DSSC bands in the MPX spectrum. A summing circuit combines the 19-kHz pilot tone, the L+R component, and the DSSC signal. The result is the stereo multiplex signal, which is fed to the modulation input of the FM transmitter.

MPX generator circuit description

The circuit diagram of the MPX generator, Fig. 3, clearly reflects the block diagram discussed above. The central oscillator is built around a 2.432-MHz

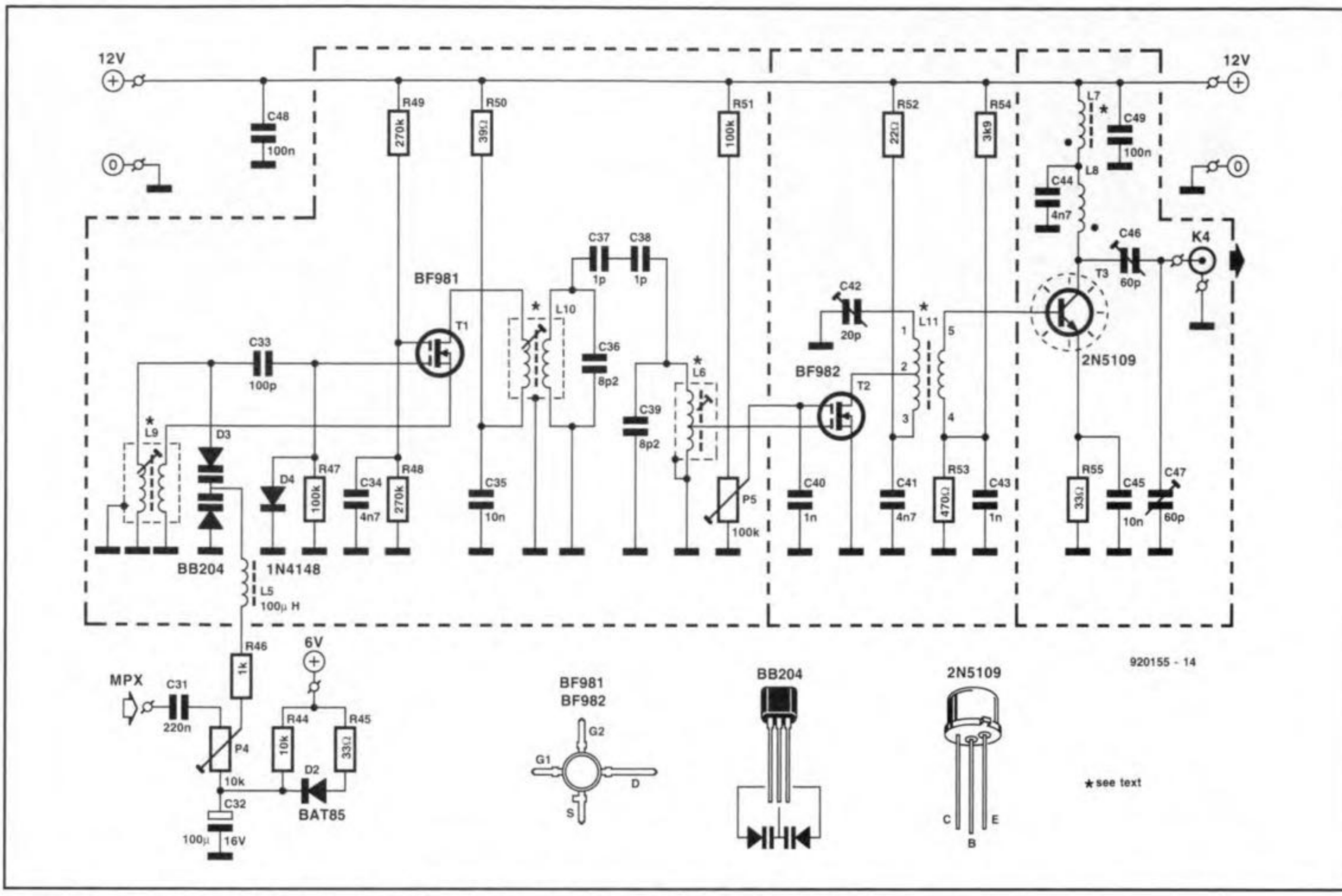


Fig. 4. Circuit diagram of the VHF FM exciter. The output power is adjusted with preset P5.

quartz crystal and the familiar CMOS Type CD4060 oscillator/divider (IC1). The phase shifter at the 19-kHz output is realized with the aid of a dual varicap (D1) and an opamp (IC2). The *L-C* pi-type low-pass filters convert the rectangular signals produced by the divider and the opamp into sine-waves that can be applied to the inputs of the analogue multiplier, IC8, an XR2208 from Exar, which functions as a PLL here. The PLL error signal controls the capacitance of D1, and thus the phase relation between the 19-kHz and the 38-kHz sine-wave. The phase is accurately adjusted with the aid of preset P1.

The left (L) and right (R) audio signals are first given a 50-µs pre-emphasis by the networks at the –inputs of opamps IC5a and IC5b. Next, the opamps contained in IC5 and IC8 turn the audio input signals into L+R and R–L components. The second XR2208 in the circuit, IC6, multiplies the 38-kHz sinewave with the R–L component. The multiplier also inverts the R–L component, so that its DSCC output signal has the L–R component as required.

The 19-kHz pilot tone, the DSCC signal and the L+R signal are summed by resistors at the –input of opamp IC7. The levels of the pilot tone and the L+R component are adjusted with presets P3 and P2 respectively. The opamp functions as a buffer for the multiplex signal. A separate AC-coupled output, marked 'A', is available for test purposes.

Finally, opamp IC4 serves as a 6-V (half the supply voltage) reference source.

VHF FM exciter

The VHF FM signal generator (exciter) is a straightforward design based on three transistors — see Fig. 4. The first, T1, functions as an FM modulator, and operates at half the generator output frequency, i.e., 44-54 MHz. The MPX signal arrives on dual varicap D3 via a modulation level control, P4, and DC and RF decoupling parts C31, R46 and L5. Note that the varicap forms the only capacitance across the oscillator inductor, L9. This is done to keep FM distortion (caused by *U-f* non-linearity) to a minimum. The modulator is pretty sensitive: the maximum deviation is achieved at an MPX level of 80 mV_{pp} already. The feedback in the oscillator is inductive via L9.

The second transistor, T2, works as an amplifier, bandfilter L10-L6 being tuned to the first harmonic of the oscillator (100 MHz). The bandfilter is a critically coupled type. It is very selective, and gives a suppression of about 50 dB at the oscillator frequency (50 MHz). Preset P5 serves to adjust the gain of T2, and thus to control the generator's output power. Power level control is must where two of these generators are used, for instance, for intermodulation distortion (IMD) measurements, which require equal power levels for the two test tones.

The power amplifier transistor, T3, is operated in class A, and supplies a maximum output power of about 150 mW into a 50-Ω load. Inductor L8 in the collector line of T3 is tuned to the output frequency by trimmers C46 and C47, which also take care of the output impedance matching.

The RF exciter and the MPX generator are powered by a 12-V supply, which must be regulated, and capable of supplying about 200 mA.

Construction

The FM stereo signal generator is built on two printed circuit boards, which have to be separated. One board contains the MPX generator, the other, the RF exciter. The artwork of the double-sided, though-plated, board is given in Fig. 6. Before you start cutting and populating the boards, however, we recommend that you make the inductors.

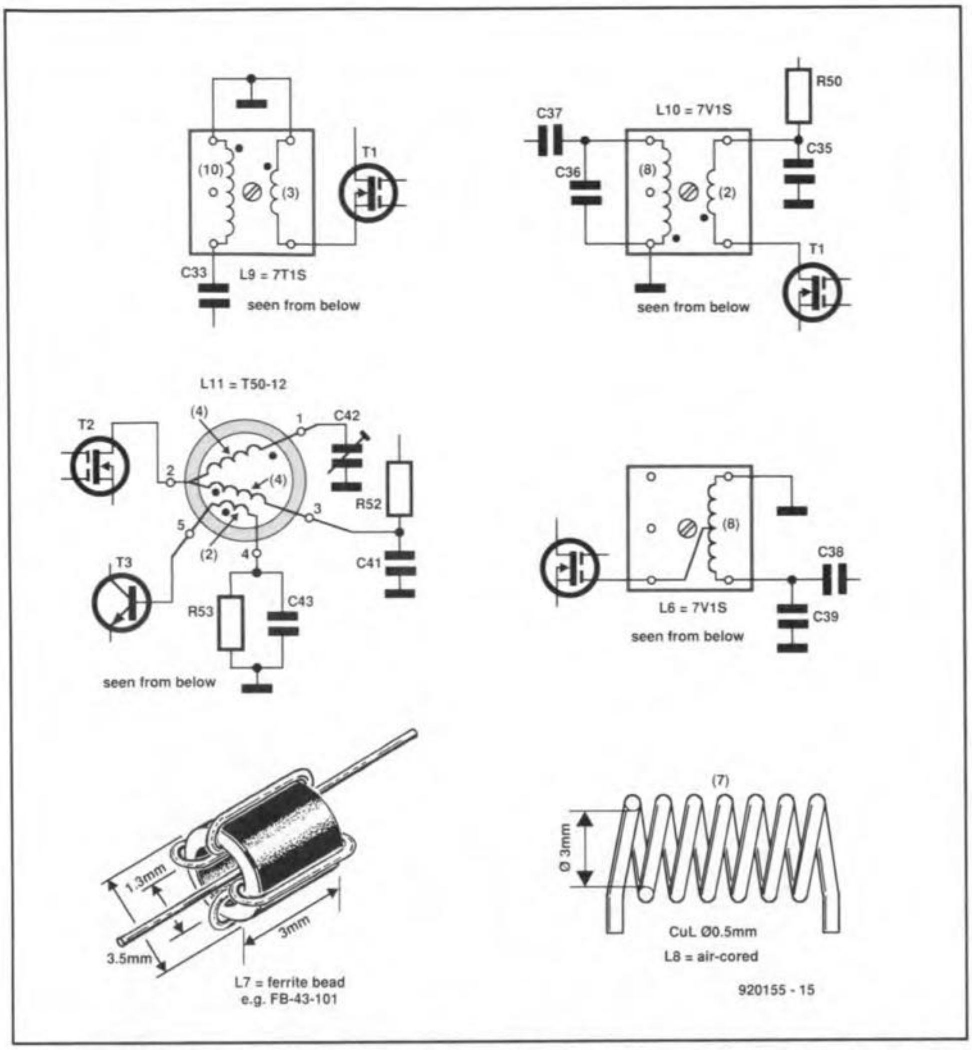


Fig. 5. Illustrating the construction and connection of the home-made inductors in the VHF FM exciter. The numbers in brackets indicate the number of turns of a particular winding, while the dots indicate the start of a winding.

Inductor construction

Since the success of the present project will depend to a large extent on the construction of the inductors, we strongly recommend that you follow this section closely. The basic construction of the home-made inductors is given in Fig. 5. The figures in brackets indicate the number of turns of a particular winding. In the description below, 'CuL' stands for enamelled copper wire. The dots in the drawings indicate the start of the winding.

L9 is wound on a Neosid 7T1S inductor assembly (yellow core). Primary winding: 10 turns 0.3 mm dia. CuL. Secondary winding: 3 turns 0.2 mm dia. CuL at the 'cold' (earthy) side of the primary winding.

L10 is wound on a Neosid 7V1S inductor assembly (green core). Primary winding: 2 turns 0.2 mm dia. CuL at the 'cold' (earthy) side of the secondary winding. Secondary winding: 8 turns 0.4 mm dia. CuL.

L11 is wound on an Amidon Associates (Micrometals) T50-12 ferrite ring core. Primary winding (1-3): 8 turns 0.7 mm dia. CuL, tap (2) at 4 turns. Secondary winding (4-5): 2 turns 0.5 mm dia. CuL at the 'cold' (earthy) side of the primary winding.

L6 is wound on a Neosid 7V1S inductor assembly (green core). Eight turns 0.4 mm dia. CuL, tap at 4 turns.

L7 is wound on a small ferrite bead, for instance, an FB-43-101 from Amidon Associates (MicroMetals). It consists of 4 turns of 0.2 mm dia. CuL.

L8 is an air-cored inductor. It consists of 7 closewound turns of 0.5 mm dia. CuL. The inside diameter is 3 mm.

Before fitting the Neosid inductors on to the board, check the continuity of the windings at the base pins. Also check for short-circuits between the windings and the screening can.

MPX generator board construction

This part of the circuit is simple to build. The two audio input sockets, K1 and K2, are mounted direct on to the board. After fitting all the components, solder 20-mm high tin plate screens over the dashed lines on the overlay. Set all three presets on the board to mid-travel.

RF exciter board construction

Some remarks are in order here, particularly for those with little experience in building RF circuits. Start the construction by fitting the three Neosid inductor assemblies. Do not mount the screening cans as yet. Next, mount the resistors, the diodes, the presets and the capacitors. The latter are fitted with the shortest possible lead length.

Proceed with the trimmers and the transistors. Attention: the type print on MOSFET T1 is legible from the component side, while that of T2 is legible from the track side of the board. Both MOSFETs are fitted at the track side of the board.

The RF power transistor, T3, is mounted at a height of about 2 mm above the board surface. It is fitted with a small clamp-on (TO-5) heat-sink.

Solder 20-mm high brass or tin plate screens on to the board at the locations marked by the dashed lines. Alternatively, bend a 'frame' from a 20-mm wide strip of tin plate, and solder it on to the board, making sure that nearby components are not damaged by overheating. The screen at the side of trimmers C46 and C47 is drilled to accept a BNC socket for single hole mounting. The centre pin of the BNC socket is soldered straight to the RF output solder pin. Finally, cut and bend a cover plate from the same material as used for making the screens. Drill holes in the cover to enable the adjustment points (trimmers, preset and inductors) to be accessed.

MPX generator adjustment

Connect a dual-channel oscilloscope to test points TP1 and TP2. Apply power, and adjust preset P1 until the zero-crossings of the 19-kHz signal and the 38-kHz signal coincide. This is illustrated in Fig. 7.

Next, adjust the L+R and pilot levels. Connect the scope to the MPX output. Apply a 1-kHz, 1-V_{pp} sine-wave to the L input, and a 300-Hz, 1-V_{pp} sine-wave to the R input. Turn the wiper of P3 (pilot level) fully counter-clockwise (to the +6 V side). Next, adjust P2 until the levels of the two components at the MPX output are equal (Fig. 8). Remove the L and R input signals, and adjust P3 for a pilot level of 100 mV_{pp} at the MPX output.

VHF exciter adjustment

Start by connecting a 50-Ω dummy

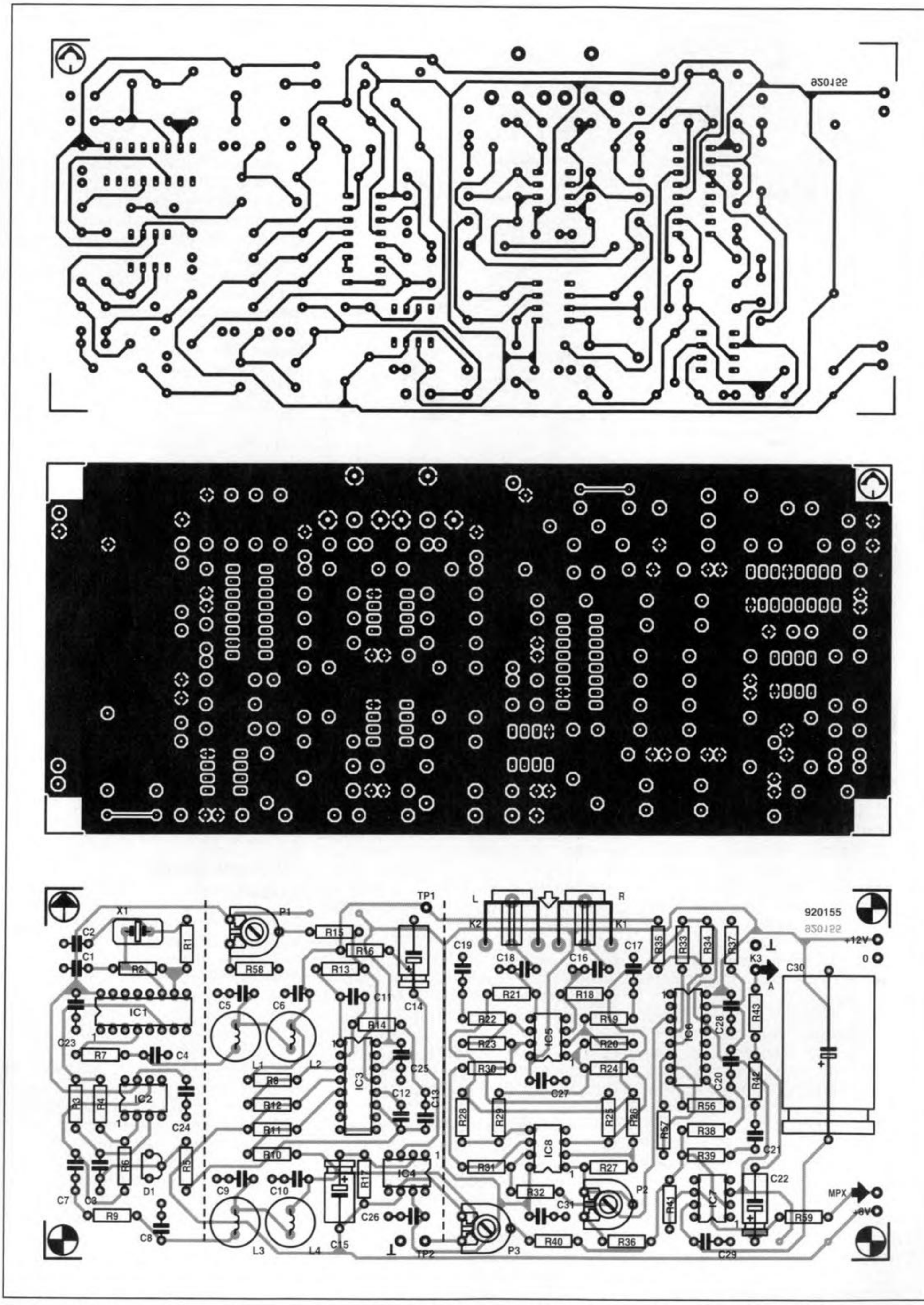


Fig. 6a. Component and solder side track layouts (mirror images) and component mounting plan of the MPX generator board.

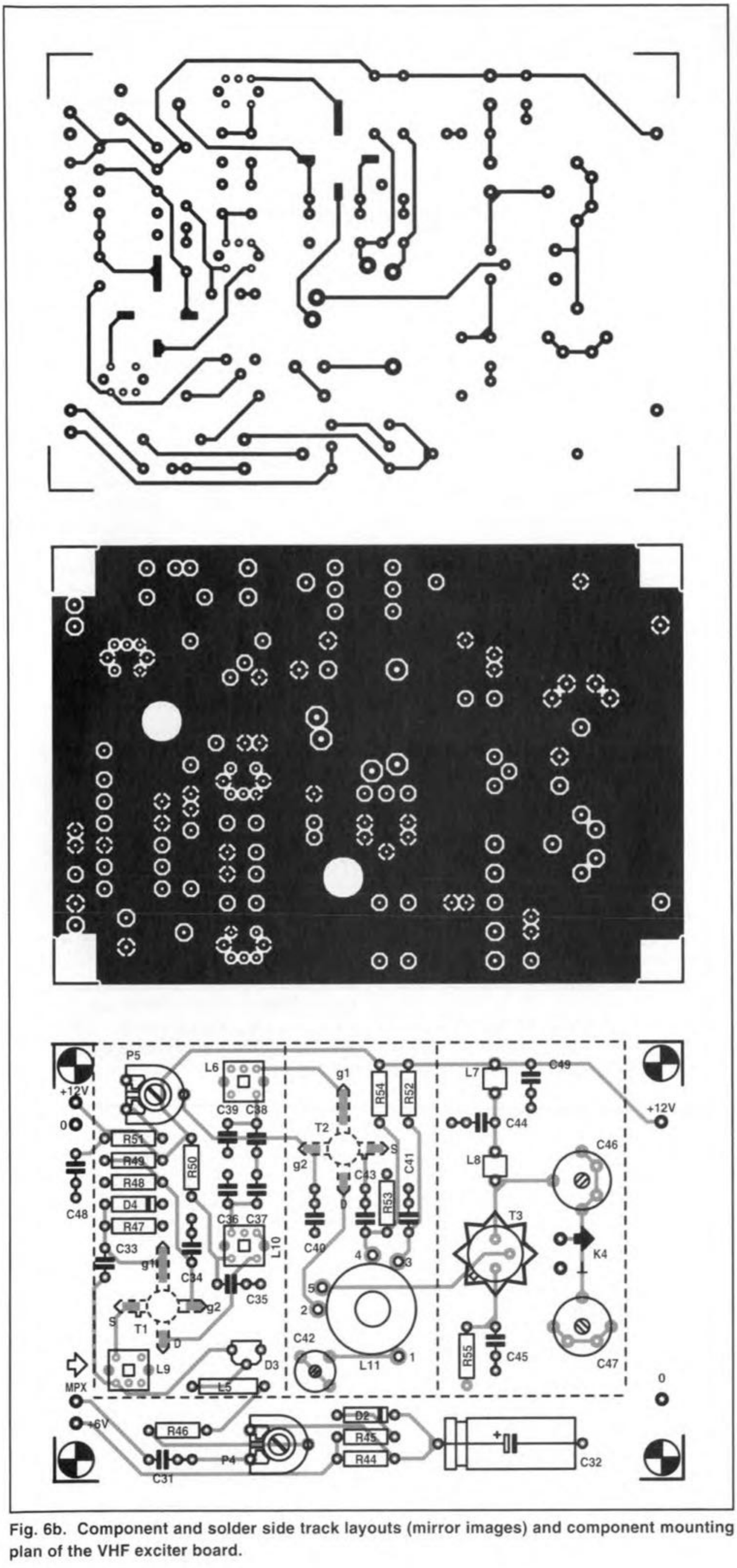


Fig. 6b. Component and solder side track layouts (mirror images) and component mounting plan of the VHF exciter board.

COMPONENTS LIST

VHF FM EXCITER

Resistors:

1	10kΩ	R44
2	33Ω	R45;R55
1	1kΩ	R46
2	100kΩ	R47;R51
2	270kΩ	R48;R49
1	39Ω	R50
1	22Ω	R52
1	470Ω	R53
1	3kΩ9	R54
1	10kΩ preset H	P4
1	100kΩ preset H	P5

Capacitors:

1	220nF	C31
1	100µF 16V	C32
1	100pF ceramic	C33
3	4nF7 ceramic	C34;C41;C44
2	10nF ceramic	C35;C45
2	8pF2 ceramic	C36;C39
2	1pF ceramic	C37;C38
2	1nF ceramic	C40;C43
1	20pF foil trimmer	C42
2	60pF foil trimmer	C46;C47
2	100nF ceramic	C48;C49

Inductors:

Winding details are given in the text

1	100µH choke	L5
2	7V1S assembly (Neosid*)	L6;L10
1	7T1S assembly (Neosid*)	L9
1	T50-12 (Micrometals**)	L11
1	3-mm ferrite bead	L7

Enamelled copper wire:
dia.'s 0.2mm, 0.3mm, 0.4mm, 0.5mm, 0.7mm

Semiconductors:

1	BAT85	D2
1	BB204	D3
1	1N4148	D4
1	BF981	T1
1	BF982	T2
1	2N5109	T3

Miscellaneous:

1	BNC socket	K4
1	TO5 heat-sink	
1	Printed circuit board 920155 (see page 70)	

* Neosid Ltd., Icknield Way West, Letchworth, Herts SG6 4AS. Telephone: (0462) 481000. Fax: (0462) 481008.
** Cirkit PLC (0992) 444111.

load to the generator output. Next, turn the wiper of P4 (modulation) fully counter-clockwise (towards the screen). Couple a frequency meter inductively to L1, and adjust the core for **half** the desired generator output frequency. Lacking a frequency meter, use an FM radio to find the first harmonic of the oscillator.

COMPONENTS LIST

MPX GENERATOR

Resistors:

1	2kΩ2	R1
1	1MΩ5	R2
3	47kΩ5 1%	R3;R4;R6
1	220kΩ	R5
5	22kΩ	R7;R9;R16; R17;R34
2	82kΩ	R8;R10
2	8kΩ2	R11;R12
2	120kΩ	R13;R14
1	1MΩ	R15
4	100kΩ	R18;R21; R40;R42
2	1kΩ5	R19;R22
3	15kΩ	R20;R23;R32
11	10kΩ	R24-R31;R36; R38;R41
1	7kΩ5 1%	R33
1	56kΩ	R35
1	3kΩ9	R37
1	1kΩ8	R39
1	680Ω	R43
2	4kΩ7	R56;R57
1	47kΩ	R58
1	100Ω	R59
1	25kΩ preset H	P1
1	100kΩ preset H	P2
1	250kΩ preset H	P3

Capacitors:

2	68pF ceramic	C1;C2
2	470pF ceramic	C4;C6
3	1nF	C5;C8;C10
9	100nF	C7;C23-C29; C31
1	4nF7	C9
1	1μF MKT	C11
1	680pF ceramic	C12
2	330nF	C13;C20
2	47μF 16V	C14;C15
2	150nF	C16;C18
2	3nF3	C17;C19
1	18pF ceramic	C21
1	10μF 16V	C22
1	4700μF 16V	C30
1	150pF ceramic	C3

Inductors:

2	47mH (Toko 181LY473J)	L1;L2
2	100mH (Toko 181LY104J)	L3;L4

Semiconductors:

1	BB212	D1
1	CD4060	IC1
1	LF411CN	IC2
2	XR2208CP*	IC3;IC6
1	OP77	IC4
1	LF357N	IC7
2	TL072	IC5;IC8

Miscellaneous:

1	2.432MHz quartz crystal; 30pF parallel resonance	X1
2	PCB-mount cinch socket	K1;K2

* Rohm Electronics (UK) Ltd., Whitehall Avenue, Kingston, Milton Keynes MK10 0AD. Telephone: (0908) 282666. Fax: (0908) 282528.

Attention:
although the artwork for the MPX generator board and the RF exciter board are shown separately in this article, these boards are supplied as ONE unit, order code 920155.This board has to be cut as described in the text to separate the MPX generator and the RF exciter sections.

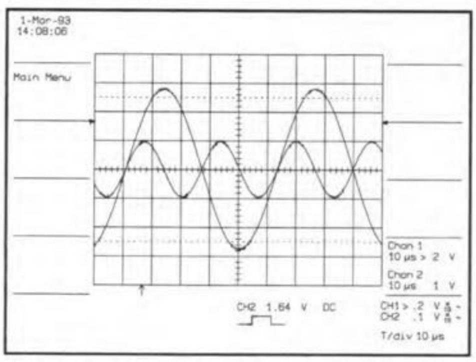


Fig. 7. The zero-crossings of the 19-kHz and 38-kHz sine-waves can be made to coincide by adjusting preset P1 in the PLL.

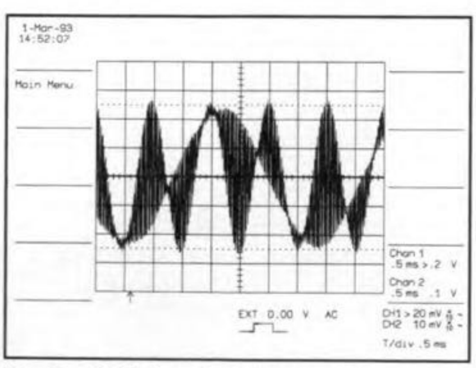


Fig. 8. Multiplex signal with L+R level correctly adjusted (no pilot).

Unless you have a 100-MHz scope with a low-capacitance probe, you will have to build the simple RF signal detector drawn in Fig. 9. Connect the scope or the detector to the drain of T1, and adjust the core of L10 for maximum RF signal amplitude. Similarly, measure the RF signal level at the tap on L6, and peak the inductor. Measure the DC voltage across R52, and adjust preset P5 for a reading of about 0.44 V. This corresponds to a drain current of about 20 mA through T2.

Move the probe to the secondary of L11, and adjust C42 for maximum RF signal. Next, carefully re-adjust L10 and L6. This is necessary to correct the effect of the load capacitance introduced by the scope probe or the RF detector input.

Connect the probe or the detector to the output of the generator. Adjust trimmers C46 and C47 for maximum output power delivered to the dummy load. Note that the trimmer settings will interact slightly.

Connect the L and R input signals (1 V_{pp} typ.), and use an FM radio to listen to the stereo signal. Carefully advance the modulation preset, P4, until the desired deviation is achieved. This is best done by comparing the sound level to that of a couple of stereo broadcast stations in the FM band. Typically, the MPX modulation voltage will be between 40 mV_{pp} and 80 mV_{pp} at the wiper of P4.

If you have not done so already, fit the ferrite cups and the screening cans on the Neosid formers. This will cause some detuning of the inductors, so small re-adjustments may be required.

The final remarks in this article concern the power supply of the generator. Because of the high sensitivity of

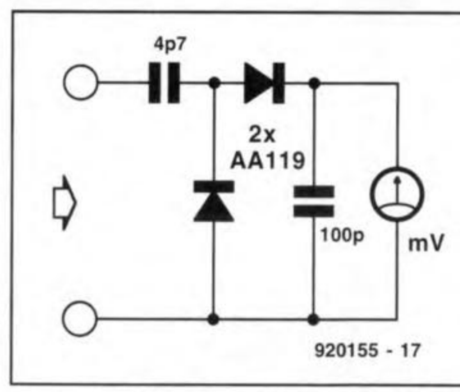
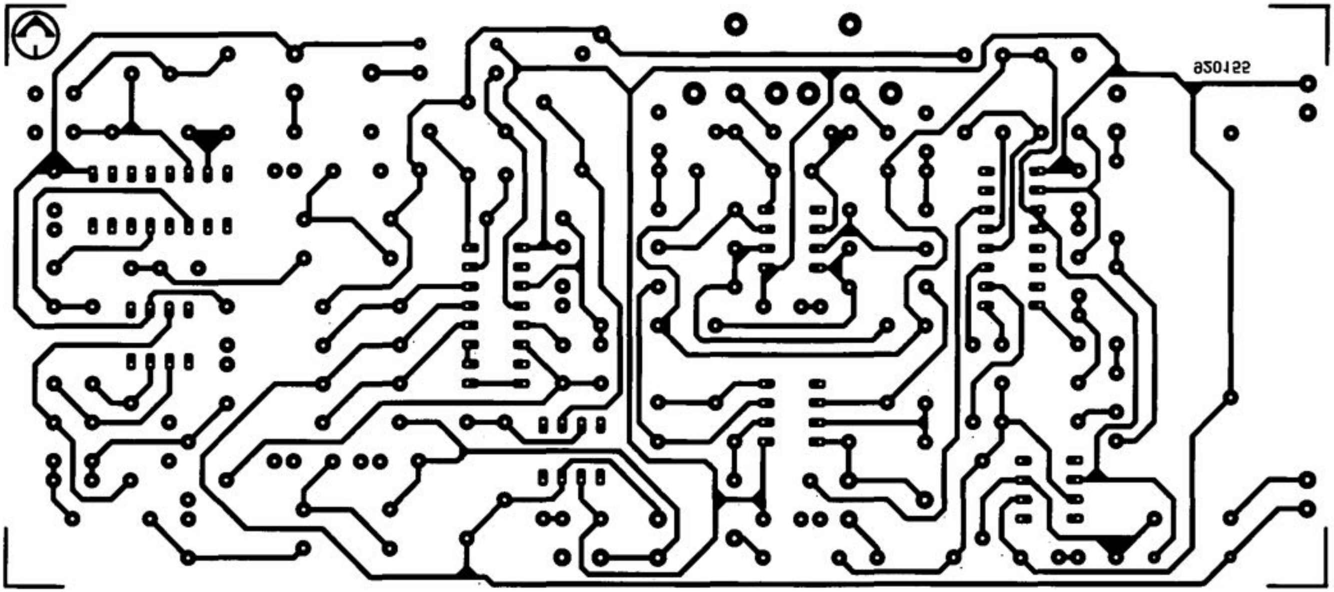
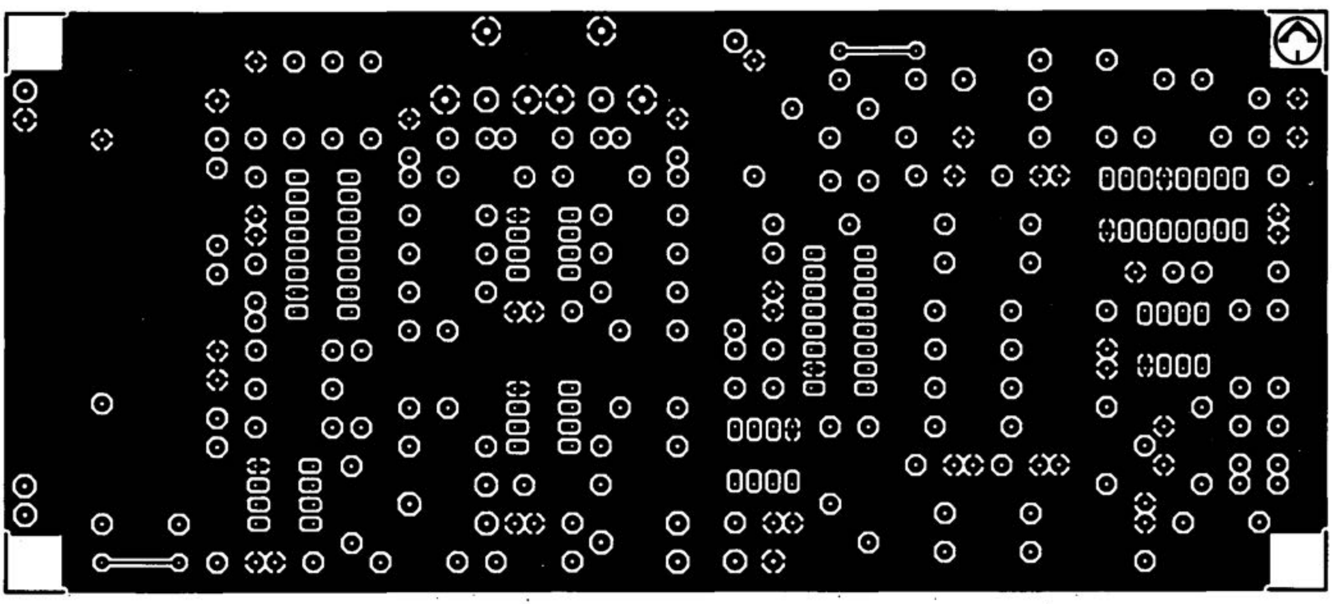


Fig. 9. Build this RF sniffer to adjust the exciter. The millivoltmeter can be a DMM set to its most sensitive direct voltage range (usually 200 mV).

the modulator, mains transformers should be kept as far as possible from the VHF exciter. If at all possible, the mains transformer should be left out of the signal generator case. ■





CORRECTIONS AND UPDATES

Digital Audio/visual system (Multi-purpose Z80 card)

May and June 1992
An extensive description of a modification to the memory backup circuit on the Multi-purpose Z80 card is available free of charge through our Technical Queries service.

FM stereo signal generator

May 1993
Capacitors C17 and C19 should have a value of 33nF, not 3nF3 as indicated in the circuit diagram and the parts list of the multiplex generator.

Workbench PSU

May 1993
The polarity of capacitor C15 is incorrectly indicated on the PCB component overlay (Fig. 5a), and should be reversed. The circuit diagram (Fig. 2) is correct.

Transformer TR2 is incorrectly specified in the circuit diagram (Fig. 2) and in the parts list. The correct rating of the secondary is 2 × 12V/5A. Also note that the secondary windings are connected in series to give 24 V.

Audio DAC

September 1992
The polarity of capacitors C25 and C58 is incorrectly indicated on the component overlay of the D-A board (order code 920062-2), and should be reversed.

U2400B NiCd battery charger

February 1993
The value of resistors R17 through R27 should be 2.7kΩ, not 12.7kΩ as stated in the parts list.

VHF/UHF receiver

May 1993
In Fig. 4, the connections to ground of the AF amplifier outputs, pins 5 and 8, should be removed. The amplifier outputs are connected to the loudspeaker only. The relevant printed circuit board is all right.

ELECTROMAGNETIC COMPATIBILITY (1)

By J. Ruiters

Electromagnetic compatibility (EMC) is the property of an electronic/electrical apparatus to function satisfactorily without radiating unacceptable interference into its electromagnetic environment. At the same time, it should be immune to interference from other electronic/electrical equipment. A non-EMC-compatible situation is shown in Fig. 1.

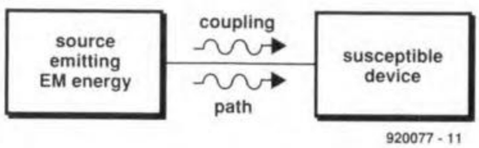


Fig. 1. The three elements that make electromagnetic interference noticeable.

Electromagnetic interference (EMI) indicates an interference problem, but is *not* the interfering signal. In practice, it may well be that an apparatus is subject to interference from more than one source or that an equipment adversely affects the operation of a number of other apparatuses.

It is clear from Fig. 1 that if one of the three elements in the drawing is missing, there is no interference or, rather, it is not noticed. In practice, the coupling path usually consists of a combination of conduction (cable, dielectric) and radiation (atmosphere).

The member states of the European Community (EC) have differing requirements as regards products that are sold and imported. This means that electrical/electronic products of a manufacturer made according to the laws of his/her country sometimes cannot be sold in other member states. However, one of the goals of the Treaty of Rome (1957) is that there shall be free movement of goods and services within the Community. In other words, a manufacturer in any of the member states must be allowed to sell his/her goods/services to any of the other member states.

As for many other products, the European Commission has issued, and the European Council of Ministers has approved, a Directive (89/336/EEC) on EMC. Such a directive, which is a law at European level, combines the requirements agreed by the member states as regards Public Health, Consumer Protection, Safety and the Environment.

To indicate that a product complies with the stringent requirements of the directive, it must carry the CE (Conformité Européenne) mark—an example of which can be seen on the toy in Fig. 2. Note, however, that this certification is *not* intended as proof of quality to the consumer. An electric coffee grinder that is EC-certified may not be very good at grinding coffee, but it can be sold in all countries of the European Community.

EMC Directive & standards

The EMC Directive is a law resulting from political endeavours to open up the European markets. It requires all electronic/electrical equipment that can cause interference or be affected by interference to comply with the EMC guide lines. From 1 January 1996, all manufacturers must be able to show that their equipment complies with the guide lines. At the same time, all relevant electrical/electronic equipment may, from 1 January 1996, be marketed in the Community only if they are CE-certified (and also comply with any other directives, of course).

It is important to note that manufacturers can already incorporate the EMC guide lines in their products to ensure that they can be marketed anywhere in the Community.

To facilitate proving and controlling conformity with the EMC guide lines, the European Electrotechnical Standardization Committee, CENELEC (Comité Européen Normalisation Electrotechnique), has formulated EMC standards. In contrast to the Directive, these standards do give a detailed technical specification In other words, the standards contain methods of measuring and concrete requirements on the basis of which manufacturers can define and test their products for compliance with the specification. Principally, compliance with the standards is not a requirement, but for large manufacturers it is probably the least expensive way of indicating that their products satisfy the guide lines. This is because manufacturers who apply the full standards need only give their own solemn declaration that their products satisfy the guide lines. National governments are obliged to accept this declaration as proof of compliance with the guide lines.

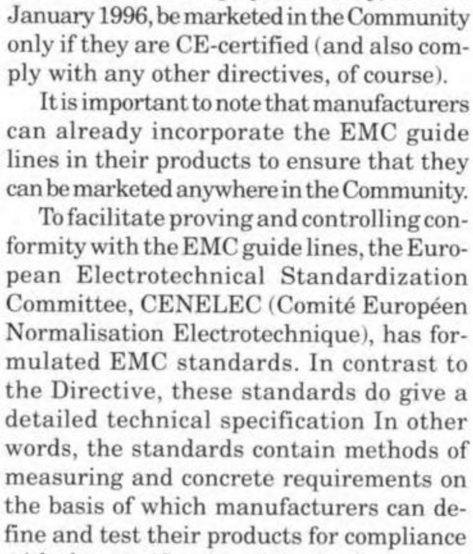


Fig. 2. This toy is CE-certified as shown at its lower rim.

If an equipment has been developed/designed/manufactured not in accordance with the standards, the manufacturer or importer must, at his/her own cost, have it tested and certified by a competent test organization. A manufacturer who applies the standards only partly saves on the cost of external testing, because the test procedure is much simpler if only a check is required to see which standards have been applied than when a complete test has to be carried out.

There are four types of standard: basic standards; general standards, product family standards, and dedicated product standards. It is noteworthy that in this sequence the range of electrical/electronic products to which the standards apply becomes smaller and smaller.

The general standards provide EMC specifications for those instances where no product group standard is available (as yet). For example, Standard EN50081-1 lays down limits of emission for equipment intended for domestic use. That is, generally, equipment for operation from the 240 V mains supply. The standard requires, for instance, that all apparatuses provided with a process or control circuit operating at a clock frequency higher than 9 kHz must be tested for interfering radiation. Legal limits have been set for the electric field: 30 dB $\mu V\ m^{-1}$ in the frequency range of 30–230 MHz and 37 dB $\mu V\ m^{-1}$ for the range 230–1000 MHz. The specified distance from the apparatus is 10 m (33 ft). These limits have been chosen to give a high probability of non-interference with radio and TV reception 10 m (33 ft) away. The limits are shown in graphic form in Fig. 3; Fig. 4 shows a possible test setup.

There are exceptions for domestic equipment, fluorescent lights and apparatus intended for communication via the mains supply. Domestic equipment is covered by Standard EN55014, while fluorescent lights and, for example, baby alarms via the mains fall under different product standards.

Specification EN50065, covering equipment for operation over mains (power) lines, deals with EMC as well as the transmission specification (operating frequency, transmit power). This shows that product standards cover EMC as well as operating requirements. Also, it explains why product

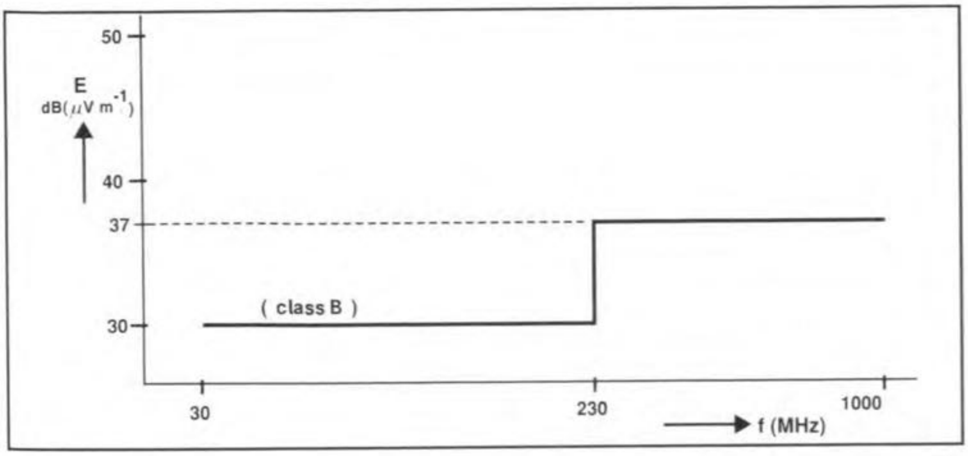


Fig. 3. Characteristic of the permissible electric field at a distance of 10 m (33 ft).

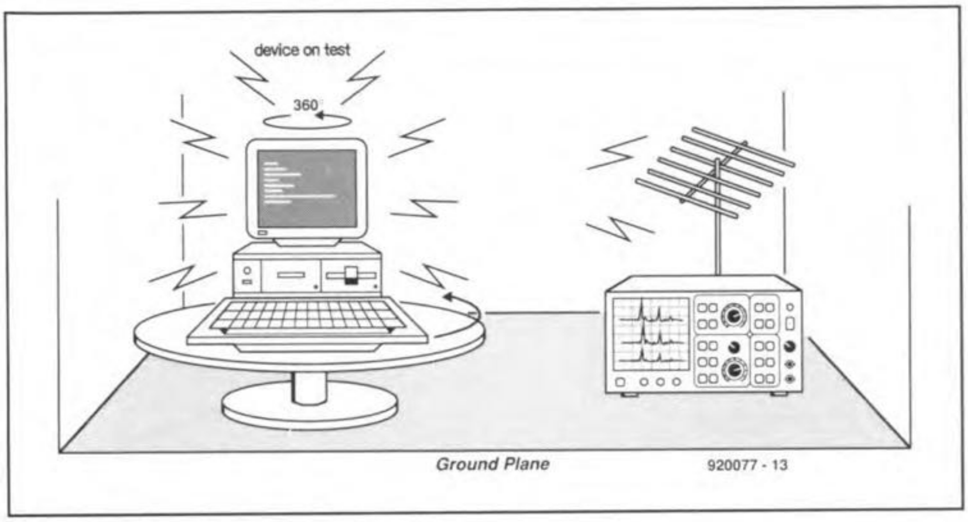


Fig. 4. Typical setup for measuring electromagnetic radiation.

standards have priority over general standards. As far as radiation limits are concerned, this does not make much difference, however.

Furthermore, EN60555-2 puts limits on the generation of harmonics on to the mains supply (power) lines. Such harmonics arise, for instance, from non-linear operations with the mains voltage—something every mains rectifier does.

In contrast to radiation standards, there are as yet few specific product standards dealing with immunity to interference, other than General Immunity Standard EN50082-1.

Legal considerations

EMC guide lines affect not only engineering, but also management and legal departments. Final responsibility for compliance with the EMC guide lines rests with the person or organization who markets the product or who takes it into use. In other words, the guide lines are of importance to producer as well as consumer. The liability of the user is limited to strict observance of the instructions in the user handbook, provided the equipment has been type-approved and EC-certified.

Marketing aspects

EMC is a problem that cannot be tackled by one discipline or one person: it requires the entire organization. Those who are acquainted with integrated quality assurance (BS9000) will know that EMC is more than just a technical quality aspect or a form of product quality. Briefly, EMC must be considered right from the research/development stage of the product through production engineering, manufacture, to quality assurance and quality control—see Fig. 5.

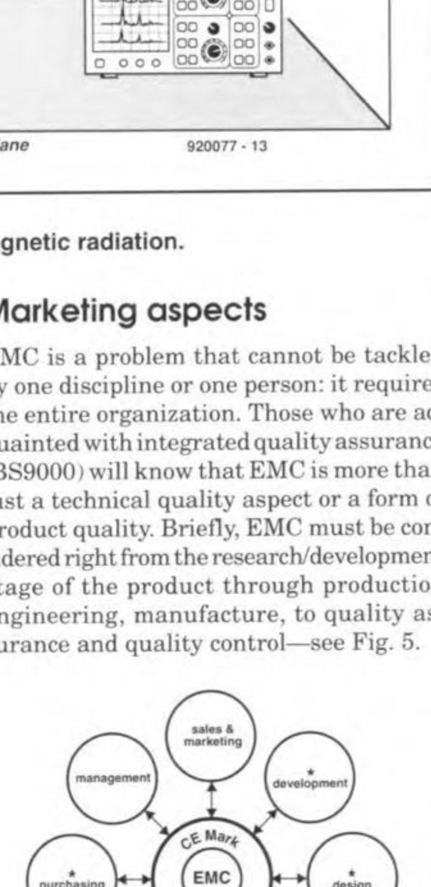


Fig. 5. Schematic representation of the departments that are (or should be) involved in ensuring compliance with EMC guides.

Quality assurance, and thus EMC, is very much a marketing responsibility. In its market analysis, marketing management must ask the question which EMC specifications are mandatory or desirable or both for the sectors of the market in which the organization is active. From such an analysis, it is possible to formulate an EMC strategy to ensure that any EMC specifications and guide lines can be incorporated in the product(s) in the most economical way.

It should be noted here that compliance with the EMC guide lines does not necessarily mean that no interference problems can occur. It may well be that designers and manufacturers have to work to even more stringent specifications to keep the number of complaints within reason.

Circuit engineering

The circuit designer has the special responsibility for ensuring that the equipment meets the marketing specification, which consists, among others, of the technical requirements and the EMC guide lines (there are also limits as to cost, the time scale for the design, after which it must be handed over to production engineering, and others).

Many circuit engineers were trained when EMC was not yet thought of and they must, therefore, acquire the necessary knowledge and experience in practical work. Younger designers were taught the fundamentals of EMC, but do not yet know how to apply these. This aspect will be reverted to in Part 2.

Suffice it for now to say that it is not enough to design an electrically well-functioning circuit. It is of real importance to look also at the side effects of the design, such as the generating of interference or its immunity thereto. Again, all electrical/electronic circuits must be designed with regard to the predicted electromagnetic environment in which they will operate. Only by incorporating the necessary safeguards in the design will it be possible to present production engineers with a prototype from which a marketable and economically viable product can be manufactured.

Production engineering

The production engineering department makes the design suitable for full-scale manufacture. Its task is, for instance, to specify a proper enclosure for the apparatus. The constituent parts of metal enclosures should not be painted or anodized before they are assembled to ensure that their junctions remain low impedance (which reduces any interference).

Another frequent cause of problems is the location of a mains filter. This filter is often placed in any odd available corner, where it may not be able to perform its task properly.

The layout of a wiring loom requires an experienced engineer, as does the feeding through of cables. If a cable, irrespective of whether it carries the mains supply, the power lines, or signals, has to pass through a screen, it must be filtered at the screen.

Purchasing

The purchaser or purchasing department is a vital link in the production chain. It stands to reason that he/she/they must ensure that all components and other electrical parts comply with the EMC guide lines. If, for instance, an IC is required, the purchaser must realize that the (faster) HCT version of this device radiates an electrical field that is three times as strong as that of an LS version. This illustrates the need for good communications between the various departments of a manufacturing organization.

The properties of components and other electrical/electronic parts play a vital role in the decision by the designer whether or not to incorporate it. For instance, a good rule of thumb is to assume that each and every solder joint, including those on a PCB, has a self-inductance of 1 nH mm^{-1}.

It may well be necessary, in view of EMC requirements, to use or add components to the design that are not essential for the electrical performance. The layout of the PCB normally needs to be varied quite a few times during the design stage. In this context, it may well be that a design based on a double-sided PCB (with copper earth planes) may be less costly to manufacture than one using a single-sided PCB, because the additional measures needed to restrict interference from the latter design often more than nullify the cost advantages of the board—see Fig. 6.

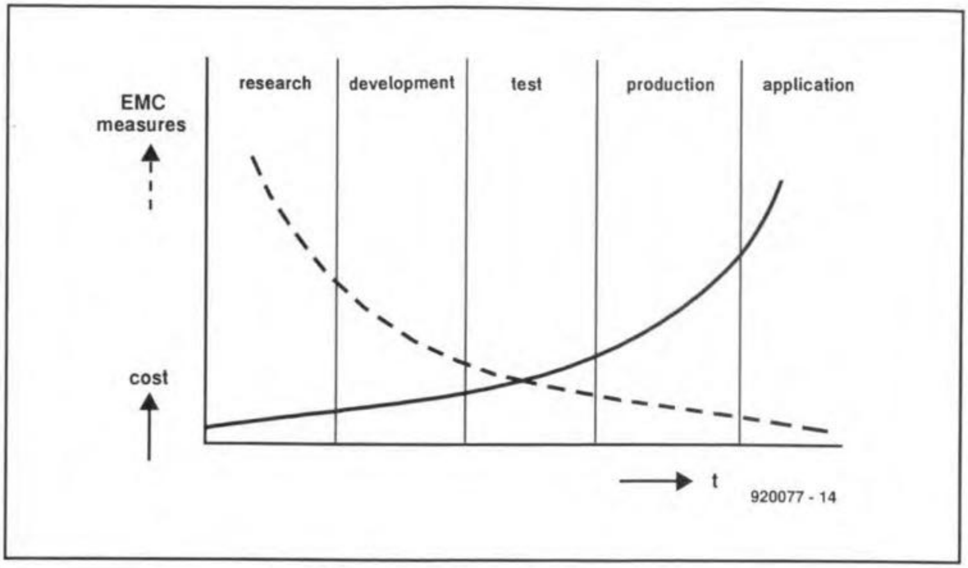


Fig. 6. EMC measures and their cost as a function of time.

Are you up to the Mark on the EEC EMC Directive?

A 50-page report from ERA Technology, entitled 'The EEC EMC Directive – Status as at 1 January 1993' provides an up-to-date picture of the European EMC regulations now the single market has come into force.

The report gives the background to the Directive and its requirements, deals with the EN (European) standards which are in place or in preparation, details the routes to compliance, and looks at the current position regarding the CE mark. Reference is also made to the UK legal regulations in the Statutory Instrument 1992 No. 2372 with respect to radio interference and immunity.

The report, 92-0768R 'The EEC EMC Directive – Status as at 1 January 1993' is available at a cost of £30 from Publication Sales, ERA Technology Ltd, Cleeves Road, Leatherhead, Surrey KT22 7SA. Telephone (0372) 374 151; Fax (0372) 377 927.

Further information

Further information may be obtained:

In the United Kingdom

General from
Technical Information Unit
Institution of Electrical Engineers
Savoy Place
London WC2R 0BL
Telephone 071 240 1871
Fax 071 497 3557

On the Directive from
The Department of Trade and Industry
151 Buckingham Palace Road
London SW1W 9SS
Telephone 071 215 5000

On radio frequency issues from
The Radiocommunications Agency
Waterloo Bridge House
Waterloo Road
London SE1 8UA
Telephone 071 215 5000

On standards from
British Standards Institution
Linford Wood
Milton Keynes MK14 6LE
Telephone (0908) 221 166

Outside the UK in the EC from
European Committee for Electrotechnical Standardization (CENELEC)
rue Brederode 2, Bte 5
B-1000 Brussels
Telephone +32 2519 6811
Fax +32 2 519 6819

Outside the European Community from
International Electrotechnical Commission (IEC)
3, rue de Varembe
CH-1211 Geneva 20
Telephone +41 22 734 0150
Fax +41 22 733 3883

Part 2, which will be published next month, will deal with a number of fundamental design considerations.

MAKING SENSE OF MEASUREMENTS (PART 1)

By Joseph J. Carr

"EVER wondered how to weigh a bull?" John's question took me by surprise. When he's not being an electronic hobbiest, my friend is the agriculture aide to a well-known U.S. Senator from a western state. John doesn't exactly look like the Outlaw Josie Wales, or even the Marlboro Man for that matter. His boyish face belies the fact that he had spent the first 22 years of his life riding horses to rope, brand and curse cattle on his father's western U.S.A. ranch (yes, there are still real cowboys, and they ride real horses). "No," I averred, "weighing bulls hasn't weighed very heavily on my mind lately." "No really, I'm serious," he retorted, "...there's some interesting measurement principles found in weighing cattle for market".

John had just returned to Washington from his home state, where he'd seen a new electronic scale at the cattle market. The problem is simple: until after they are slaughtered for beef, cattle are weighed alive, and it is nearly impossible to get them to stand still long enough for a conventional scale to read a steady value. As a result, the scale makers turn to a method that is based on simple statistics — and these methods are just as applicable to other electronic measurements and to the calibration of electronic instruments.

Before looking at the specific method used for weighing a cow, let us take a look at some basics.

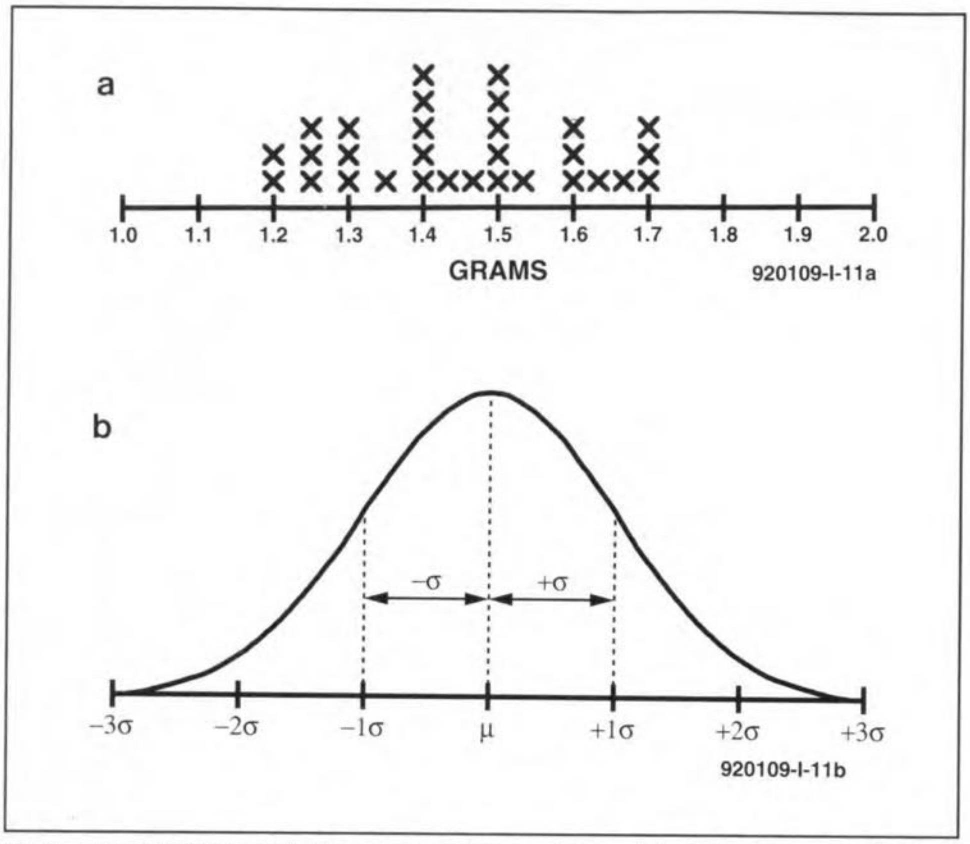


Fig. 1. a) Data points of actual capacitor weights are dispersed (note that none actually are at the design value of 1.42 grams; b) normal distribution curve.

The basics

Measurements are "...the assignment of numerals to represent [physical] properties."[1] Measurements are made to fulfill one or more of several different goals: obtain information about a physical phenomenon, assign a value to some fundamental constant, record trends, control some process, correlate behaviour with other parameters in order to obtain insight into their relationships, or figure out how much to pay for a beef cow. A measurement is an act that is designed to "derive quantitative information about..." some physical phenomenon "...by comparison to a reference..."[2] or standard. The physical quantity being measured is called the **measurand**.

All measurements are subject to a certain amount of **variation** caused by small **errors** in the measurement process, and by actual variation in the measured parameter. In this context, the idea of 'error' does not mean 'mistake', but rather a normal random variation due to inherent limitations of the system. There are many different causes of random variation. Some of them are dependent on the particular type of measurement being made (and are a function of the type of meter being used), while other error is due to inherent variation in the process being measured.

Consider an ordinary meter stick as an analogy. It is divided into 100 large divisions of 1 cm each, and 1000 small divisions of 1 mm each. Or is it? In truth, the distance from 0 to 100 cm is not exactly 100.0000 cm, but rather $100 \pm \varepsilon$ cm, where ε is some small error. In addition, the spaces between divisions (either 1 mm or 1 cm) are not exactly the same size, but vary somewhat from mark to mark. As a result, when you measure the size of, say, a printed circuit board there will be some error.

Random variation causes the data obtained when making measurements to **disperse**. Consider a practical situation. Suppose we want to measure the mass of an electronic component, say a capacitor. The design specification weight is supposed to be 1.42 grams, but when a sample of thirty identical capacitors are weighed on a very good scale the results shown in the crude bar graph of Fig. 1, and Table 1, were obtained. Even though the capacitors are all identical, there was a dispersion of the data points. Not one capacitor in the sample lot actually met the 1.42-grams specification exactly, but rather were dispersed around 1.42 grams.

If you examine a large number of capacitors, and plot their weights as in Fig. 1a. you will find that a pattern emerges. As

Table 1. Sorted data points (low to high sort)

1.2, 1.2, 1.26, 1.26, 1.26, 1.3, 1.3, 1.3, 1.35, 1.4, 1.4, 1.4, 1.4, 1.4, 1.43, 1.48, 1.5, 1.5, 1.5, 1.5, 1.53, 1.6, 1.6, 1.6, 1.62, 1.65, 1.7, 1.7, 1.7

long as certain conditions are met, the various weights will form into the familiar 'bell-shaped' curve (Fig. 1b) that is called, variously depending on country and context, the **normal distribution curve**, the **Gaussian curve** or the **Laplacean**. Regardless of what it is called, the normal distribution pattern of Fig. 1b is extremely common in all of science and technology.

The normal distribution curve plots frequency of occurence against some other parameter. Note in Fig. 1a that, even with only a few data points, some values are beginning to stack up more 'X' check marks than other values. When thousands are measured, it is likely that the result is a kind of bar graph. The **mean** is usually designated μ when the entire population of values is plotted, or $\overline{X}$ when only a sample of the population is taken.

One of the uses of the normal distribution curve is that it offers us a measure of the dispersion of the data. This property of the data can be summed up as the **variance** and **standard deviation** of the data. For the entire population, variance is denoted by σ^2 and standard deviation by σ. Variance is defined by:

$$\sigma^2 = \frac{\sum_{i=1}^{N}(X_i - \overline{X})^2}{N} \quad , \qquad [1]$$

and standard deviation, which is the square root of variance, by:

$$\sigma = \sqrt{\frac{\sum_{i=1}^{N}(X_i - \overline{X})^2}{N}} \quad . \qquad [2]$$

Equations [1] and [2] define the variance and standard deviation for the entire population of data. If a small sample is taken, then replace σ^2 with s^2, σ with s, and N in the denominators with N–*1*.

If the process is truly random, then the value that one would report in making the measurement is the mean value, μ. But it is also necessary to specify either the variance or standard deviation. We know that any particular measurement may or may not be μ. We also know that 68.27% of all values lay between ±σ (see Fig. 1b); 98.45% between ±2σ and 99.73% between ±3σ.

Categories of measurement

There are three general categories of measurement: **direct**, **indirect** and **null**.

Direct measurements are made by holding the measurand up to some calibrated standard and comparing the two. A good example is the meter stick used to cut a piece of cable to the correct length.

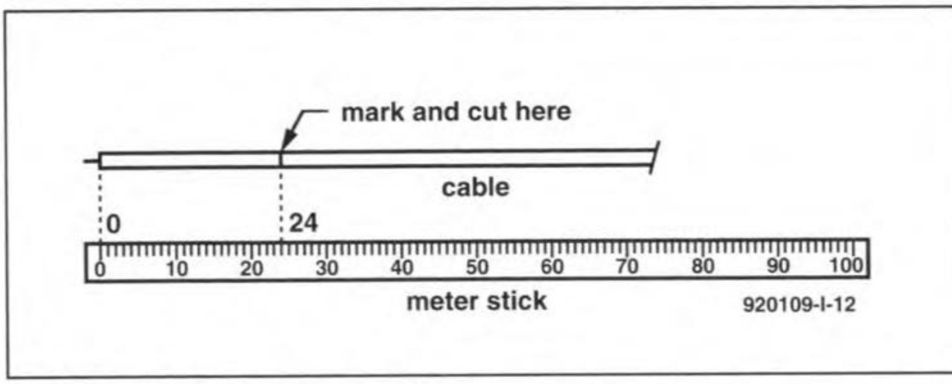


Fig. 2. Example of a direct measurement.

You know that the cable must be cut to a length of 24 cm, so you hold a meter stick (the standard or reference) up to the piece of cable (Fig. 2), set the zero centimeters point at one end, and make a mark on the cable adjacent to the '24' mark on the meter stick, and then make your cut.

Indirect measurements are made by measuring something other than the actual measurand. Although frequently considered 'second best' from the perspective of measurement accuracy, indirect methods are often used when direct measurements are either difficult or dangerous. For example, one might measure the temperature of a point on the wall of a furnace that is melting metal (Fig. 3a), knowing that it is related to the interior temperature by a certain factor (Fig. 3b).

There was once a minicomputer manufacturer who used an indirect temperature measurement to ease the job of the service technicians. They created a small hole at the top of the rack mounted cabinet where the temperature would be <39 °C when the temperature on the electronic circuit boards was within specification. They used this method for two reasons: a) the measurement point was available to the outside world, and thus did not require any disassembly; and b) the service technician could use an ordinary household medical fever thermometer (30 °C to 42 °C) as the measurement instrument. No special laboratory thermometers were needed.

Perhaps the most common example of an indirect measurement is the human blood pressure. It is measured by measuring the pressure in an occluding cuff placed around the arm; a process called **sphygmomanometry** (Fig. 4). Research showed that the cuff pressures at two easily heard events (onset and cessation of 'Korotkoff sounds') are used to detect the systolic (P_s) and diastolic (P_d) arterial pressures. Direct blood pressure measurement is dangerous because it is an invasive surgical procedure.

Null measurements are made by comparing a calibrated source to an unknown measurand, and then adjusting either one or the other until the difference between them is zero. An **electrical potentiometer** is such an instrument; it is an adjustable calibrated voltage source and a comparison meter (zero-centre galvanometer). The reference voltage from the potentiometer is applied (Fig. 5) to one side of the zero-centre galvanometer (or one input of a difference measuring voltmeter), and the unknown is applied to the other side of the galvanometer (or remaining input of the differential voltmeter). The output of the potentiometer is adjusted until the meter reads zero difference. The setting of the potentiometer under the nulled condition is the same as the unknown measurand voltage.

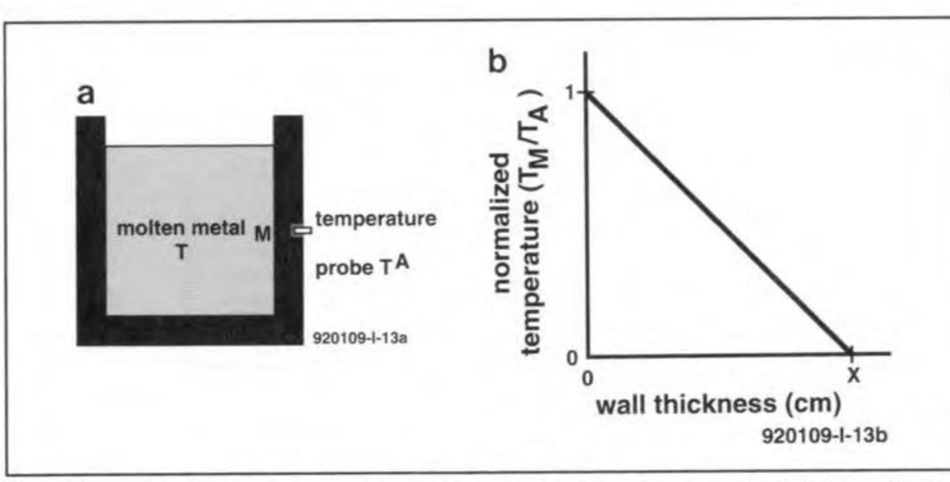


Fig. 3. Example of an indirect measurement: a) temperature probe is placed in the side of a melting furnace; b) relationship between normalized temperature ratio T_m/T_A and wall thickness.

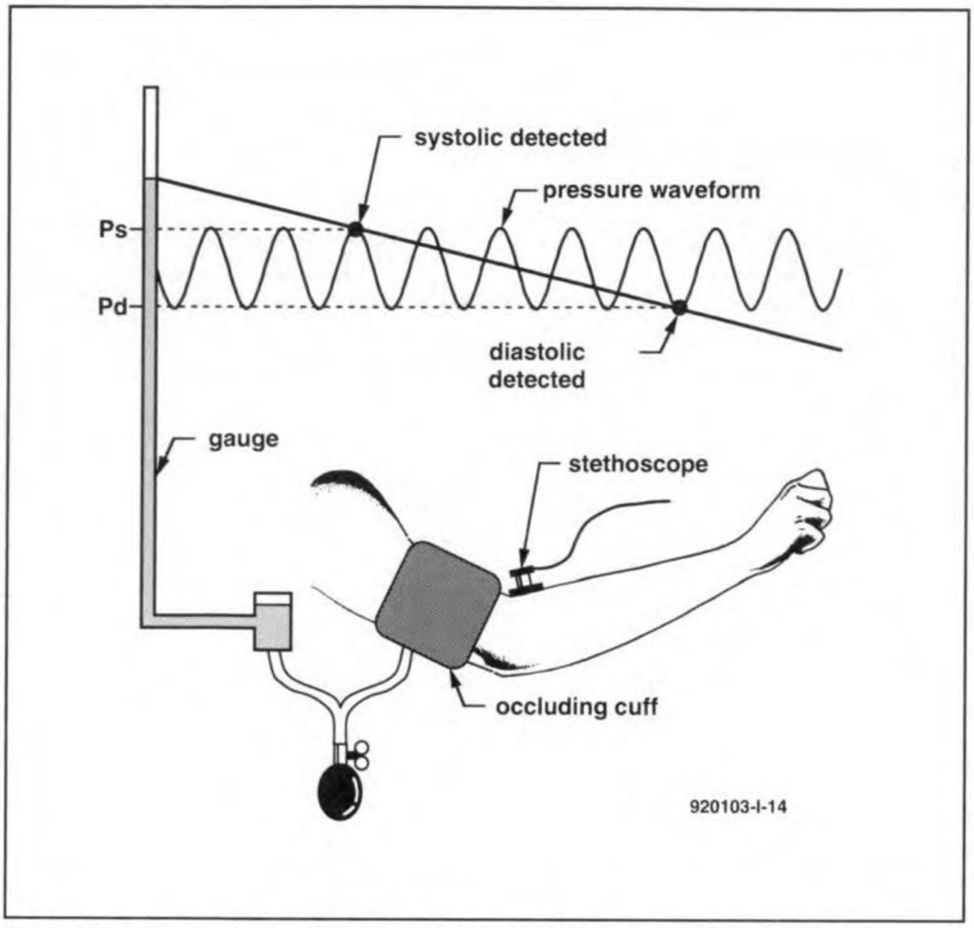


Fig. 4. Example of an indirect measurement: human blood pressure by sphygmomanometry.

Factors in making measurements

The 'goodness' of measurements involves several concepts that must be understood. Some of the more significant of these are: **error**, **validity**, **reliability**, **repeatability**, **accuracy**, **precision**, and **resolution**.

In all measurements there is a certain degree of error present. Recall that the word 'error' in this context refers to normal random variation, and in no way means 'mistakes' or 'blunders'. In short order we will discuss error in greater depth.

If measurements are made repeatedly on the same parameter (which is truly unchanging), or if different instruments or instrument operators are used to make successive measurements, it will be found that the measurements tend to cluster around a central value (X_o in Fig. 6). In most cases, it is assumed that X_o is the true value, but if there is substantial inherent error in the measurement process, then it may deviate from the true value (X_i) by a certain amount (ΔX) — which is the error term. The assumption that the central value of a series of measurements is the true value is only valid when the error term is small. As $\Delta X \rightarrow 0, X_o \rightarrow X_i$.

The **validity** of a measurement is a statement of how well the instrument actually measures what it purports to measure. An electronic pressure sensor may actually be measuring the deflection of a thin metallic diaphragm (of known area, of course), which is in turn measured by the strain applied to a strain gauge cemented to the diaphragm. What determines the validity of a sensor measurement is the extent to which the measurement of the deflection of that diaphragm relates to applied pressure ...and over what range or under what conditions. Many measurement devices exist where the output readings are only meaningful under the right conditions or over a specified range.

The **reliability** of the measurement is a statement of its **consistency** when discerning the values of the measurand on different trials, when the measurand may take on very different values. In the case of the pressure sensor discussed above, a deformation of a diaphragm may change its characteristics sufficient to alter future measurements of the same pressure value. Related to reliability is the idea of **repeatability**, which refers to the ability of the instrument to return the same value when repeatedly exposed to the exact same stimulant. Neither reliability nor repeatability are the same as accuracy, for a measurement may be both 'reliable' and 'repeatable' while being quite wrong.

The **accuracy** of a measurement refers to the freedom from error, or the degree of conformance between the measurand and the standard. **Precision**, on the other hand, refers to the exactness of successive measurements, also sometimes considered the degree of refinement of the measurement. Accuracy and precision are often confused with one another, and these words are often erroneously used interchangeably. One way of stating the situation is to note that a precise measurement has a small standard deviation and variance under repeated trials, while in an accurate measurement the mean value of the normal distribution curve is close to the true value.

Figure 7 shows the concepts of accuracy and precision in various measurement situations. In all of these cases, the data form a normal distribution curve when repeatedly performed over a large number of iterations of the measurement. Compare Figs.7a and 7b. Both of these situations have relatively low accuracy because there is a wide separation between X_o, the measured value, and X_i, the actual value of the measurand. The measurement represented in Fig. 7a has relatively high precision compared with Fig. 7b. The difference is seen in the fact that Fig. 7a has a substantially lower

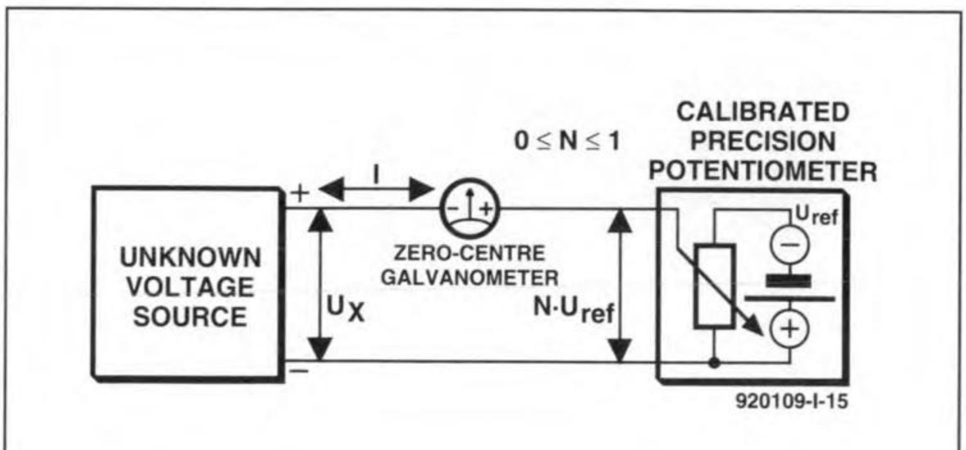


Fig. 5. Electrical voltage measurement using a null method.

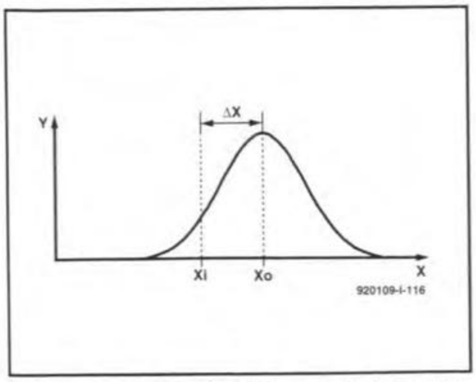


Fig. 6. Error ΔX is the difference between the actual value X_i and the measured value X_o.

variance and standard deviation around the mean value, X_o; this curve shows poor accuracy but good precision. The variance of Fig. 7b is greater than the variance of the previous curve, so it has both low accuracy and poor precision. Somewhat different situations are shown in Figs. 7c and 7d. In both of these distributions, the accuracy is better than the previous cases, because the difference between X_o and X_i is reduced (of course, a perfect, error-free, measurement has a difference of zero). In Fig. 7c, the measurement has good accuracy, and good precision, while in Fig. 7d the measurement has good accuracy and poor precision (its variance is greater than in Fig. 7c).

The standard deviation of the measurement is a good indication of its precision, which also means the inherent error in the measurement. One of my more fondly remembered college professors taught us to reduce error in our laboratory experiments by several methods:

1. Make the measurement a large number of times, and then average the results.
2. Make the measurement using different instruments, if feasible.
3. When using instruments such as rulers or meters, try making the successive measurements on different parts of the scale. For example, on rulers the distance between tick marks is not really constant because of manufacturing error. The same is also true of electrical meter scales. The professor had us measure lengths using different points on the scale as the zero reference point (e.g. on a meter stick use 2, 12, 20, and 30 cm as the zero point), and then average the results. By taking the measurements from different sections of the scale, both individual errors and biases that accumulate will be averaged to a lower overall error.

Resolution refers to the degree which the measurand can be broken into identifiable adjacent parts. An example can be seen on the standard television test pattern broadcast by some stations in the early morning hours between 'broadcast days'. Various features on the test pattern will include parallel vertical or horizontal lines of different densities. One patch may be 100 lines per inch, another 200 lines per inch, and so forth up the scale. The resolution of the video system is the maximum density *at which it is still possible to see adjacent lines with space between them*. For any system, there is a limit above which the lines are blurred into a single entity.

In a digital electronic measuring instrument, the resolution is set by the number of bits used in the data word. Digital instruments use the binary (base-2) numbers system in which the only two permissible digits are 0 and 1. The binary 'word' is a binary number representing a quantity. For example, 0001 represents decimal 1, while 1001 represents decimal 5. An 8-bit data word, the standard for many small computers, can take on values from 00000000_2 to 11111111_2, so can break the range into 2^8 (256) distinct values, or $2^8–1$ (255) different segments. The resolution of that system depends on the value of the measured parameter change that must occur in order to change the least significant bit in the data word. In digital systems, this resolution is often specified as the '1-LSB' resolution.

Measurement errors

No measurement is perfect, and measurement apparatus are never ideal, so there will always be some **error** in all forms of measurement. An error is a deviation between the actual value of a measurand and the indicated value produced by the sensor or instrument used to measure the value. Let me reiterate: error is inherent, and is NOT the fault of the person making the measurement. Error is not the same as mistake! Understanding error can greatly improve our effectiveness in making measurements.

Error can be expressed in either (so-called) 'absolute' terms, or using a relative scale. An absolute error would be expressed in terms of '$X \pm x$ cm,' or some other such unit, while a relative error expression would be '$X \pm 1\%$ cm'. Which to use may be a matter of convention, personal choice or best utility, depending on the situation.

There are four general categories of error: **theoretical error**, **static error**, **dynamic error**, and **instrument insertion error**. Next time we will take a closer look at these categories of error — and figure out how to weigh a cow. ❑

References and notes

1. N.R. Campbell, *Foundations of Science*, Dover (New York, 1957). Cited in John Mandel, *The Statistical Analysis of Experimental Data*, John Wiley & Sons (New York 1964). Dover paperback edition 1984.

2. E.E. Herceg, *Handbook of Measurement and Control*, Schaevitz Engineering (1972, Pennsauken, NJ).

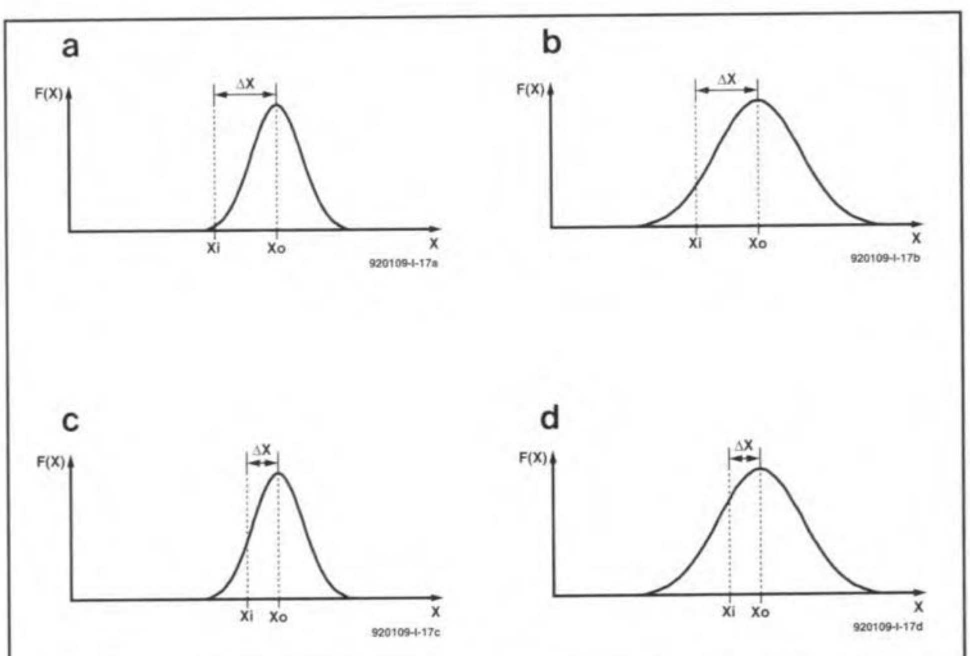


Fig. 7. a) Good precision, poor accuracy; b) poor precision and poor accuracy; c) good precision and good accuracy; d) poor precision, good accuracy.

WORKBENCH POWER SUPPLY UNIT

Design by K. Schönhoff ✓

A stabilized power supply unit with current limiting is described that not only provides but also accepts power. This property is indispensable, for example, for the testing of batteries.

In principle, the power supply unit, PSU, is fairly straightforward. The part that provides power consists of a main circuit whose output voltage is converted into a variable, stabilized potential with the aid of a preset voltage regulator. A resistor in series with the outputfacilitates current measurements and is also needed for the current limiting circuit.

Apart from the main circuit, there is an ancillary circuit that provides power for the measurement and control circuits for the voltage setting and the variable current limiting respectively.

Moreover, the PSU contains a circuit that makes it possible for power applied to the output from an external source to be accepted. This facility enables the charging and discharge currents of batteries, battery chargers and other power supplies to be tested. It could also enable the PSU to serve as a power zener in certain measurements. Owing to its 'bivalent' nature, the unit strongly resembles a battery. This is achieved not with an esoteric design, but with a mirror image of the main circuit.

Circuit description

The upper part of the main circuit in Fig. 2 *provides* power, whereas the lower part *accepts* power from an external source.

The mains transformer—not shown in Fig. 2—is connected to K_3. Its secondary voltage (24 V) is converted by rectifier D_1–D_4 into a direct voltage of about 35 V. Capacitor C_1 is a smoothing element. Additional smoothing and stabilization is provided by double darlington output stage T_4-T_6, which is driven by IC_{3a} and T_2. The collector-emitter circuit of the output stage is in series with the +ve terminal of the output of the unit, K_4. The –ve terminal of K_4 is linked direct to the negative terminal of C_1.

Series regulator T_4-T_6 obtains its base current normally from T_2, which is connected as a constant-current source. This source contains a potential divider, R_8-R_9, whose terminals are connected to the +ve and ⊥ terminals of regulator IC_1, which provides the 5 V ancillary supply. The input of the regulator is obtained from the secondary winding of Tr_1, bridge rectifier B_1 and smoothing capacitor C_2. The value of R_7 determines the level of the current provided by T_2: here 8 mA.

The output stage does not need the same base current all the time, of course. It must be able to adapt its output voltage and peak current to the requirements of the user as well as to the possible variations in the load. Therefor, the constant current provided by T_2 can be varied by IC_{3a}. This IC decreases the base current (by lowering its output voltage) to T_4-T_6 if the output voltage of the PSU tends to become larger than the set value. Part of the current provided by T_2 then no longer flows to T_4-T_6, but to the output of IC_{3a} via D_5.

Since a darlington has a double base-emitter junction, it will commence conducting only when the base voltage is 1.2 V. If the cathode of D_5 is held at earth potential by the output of IC_{3a}, the voltage at the anode of the diode is at most 0.6 V, which is well below the 1.2 V necessary for the output stage to commence conducting. In other words, the output stage is cut off. This has the advantage that there is then not even a quiescent current through the darlington.

In this way IC_{3a} controls the output voltage of the PSU. Before it can do so, however, it must have information as to the level of the output voltage. In other words, and as usual in a regulated supply, before anything can be controlled, measurements will have to be carried out.

Measurements for IC_{3a} are effected as follows. The non-inverting input of IC_{3a} (pin 3) is connected to preset P_1 via R_1 and K_5-K_6 (see also Fig. 2). However, pin 3 is also connected to the –ve terminal of K_4 via R_2.(after a jump lead has been placed between REF and –).

The inverting input of IC_{3a} (pin 2) is linked to the +ve terminal of K_4 via R_3 (this link is connected to the ⊥ of the ancillary supply circuit.

In this way, the inputs of IC_{3a} are supplied with the potential difference across the +ve and –ve terminals of K_4. The IC compares this potential with the set (P_1) reference voltage. Depending on the measured result, the output of IC_{3a} becomes higher or lower, and in this way the IC

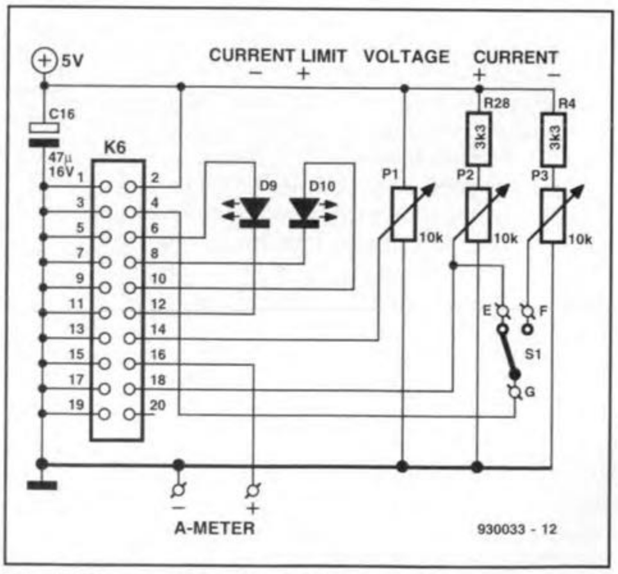


Fig. 1. The control and indicator circuits.

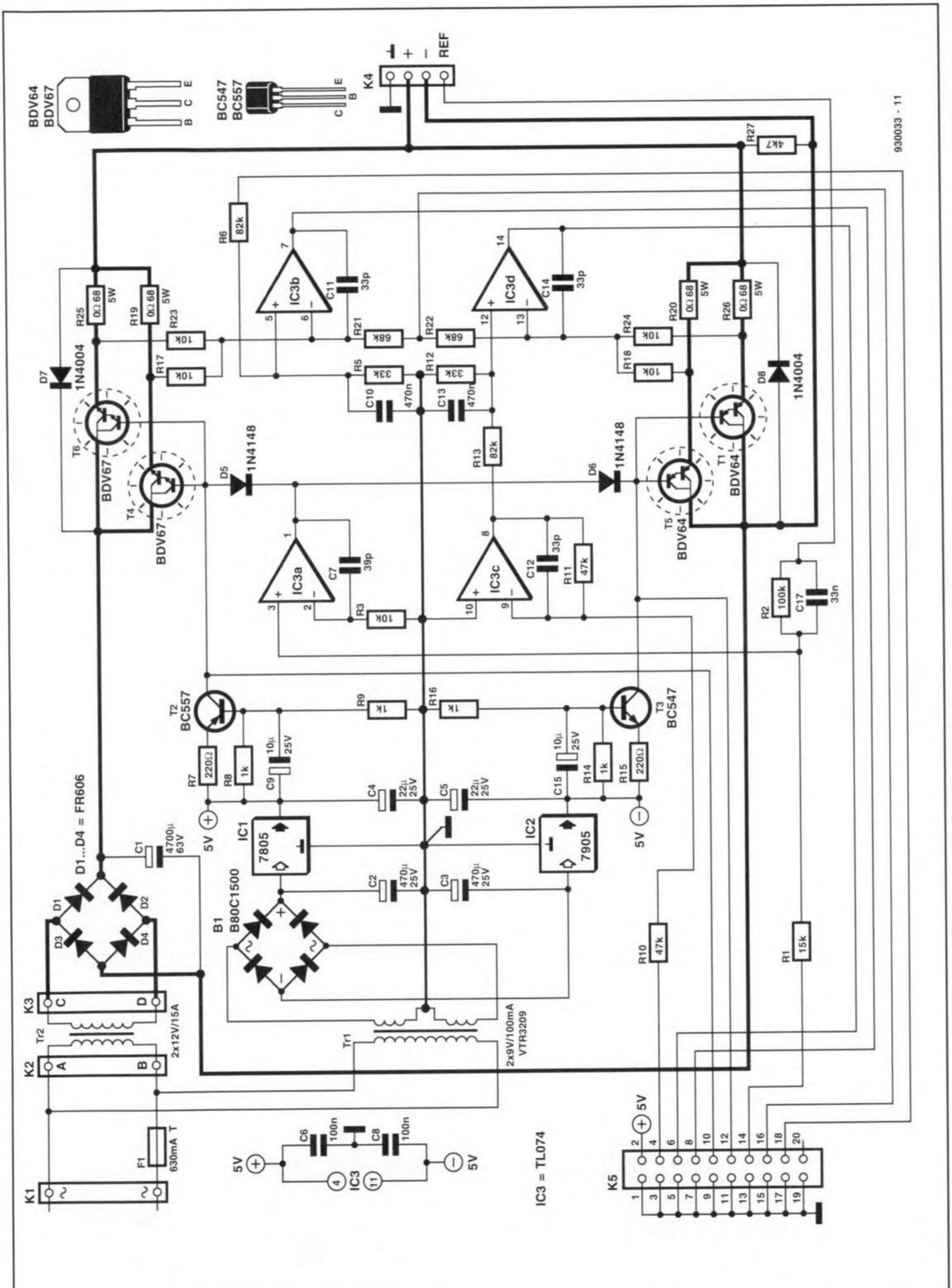


Fig. 2. Circuit diagram of the workbench power supply unit.

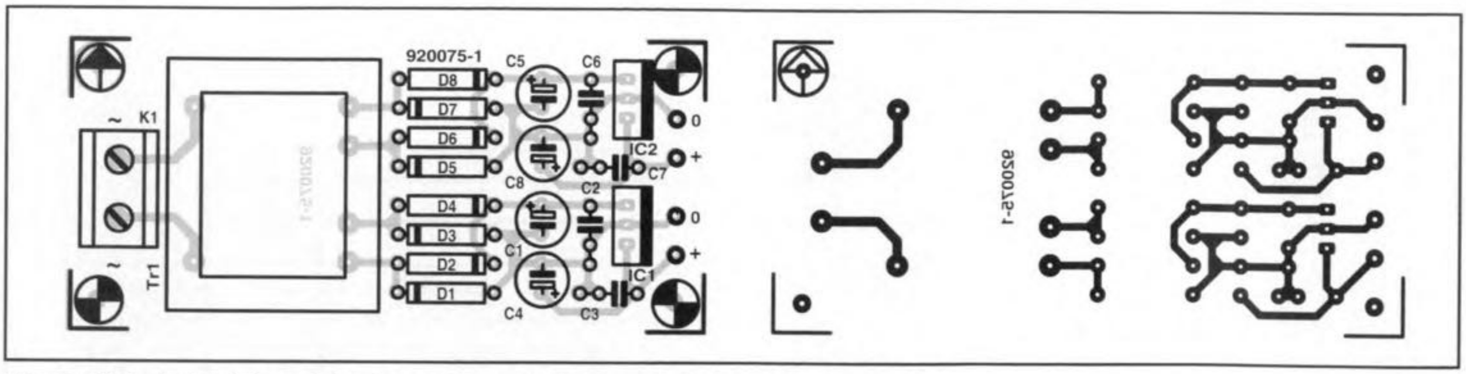


Fig. 3. Printed-circuit board for the auxiliary supply for the digital meter modules.

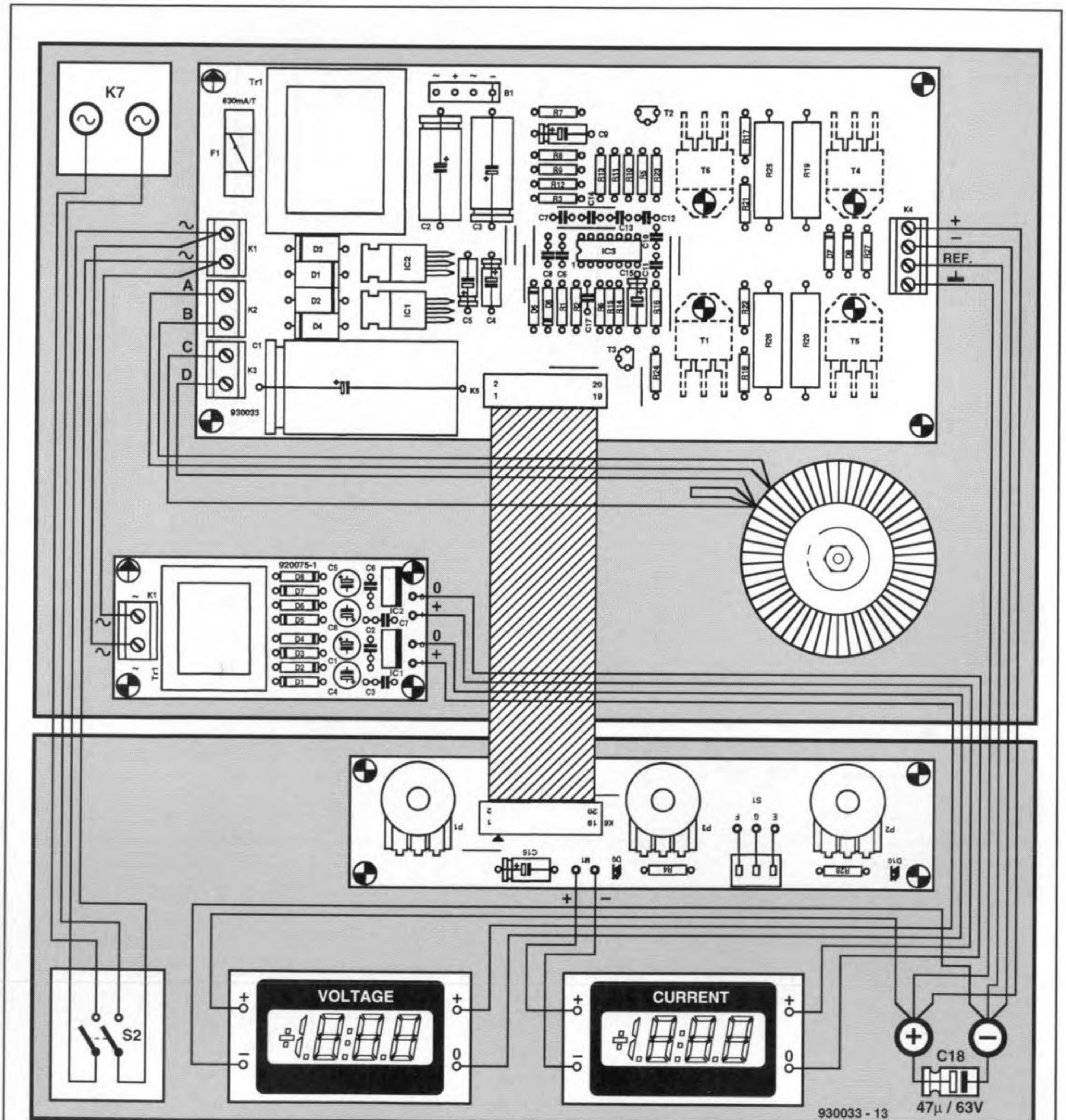


Fig. 4. Wiring diagram of the complete workbench power supply unit.

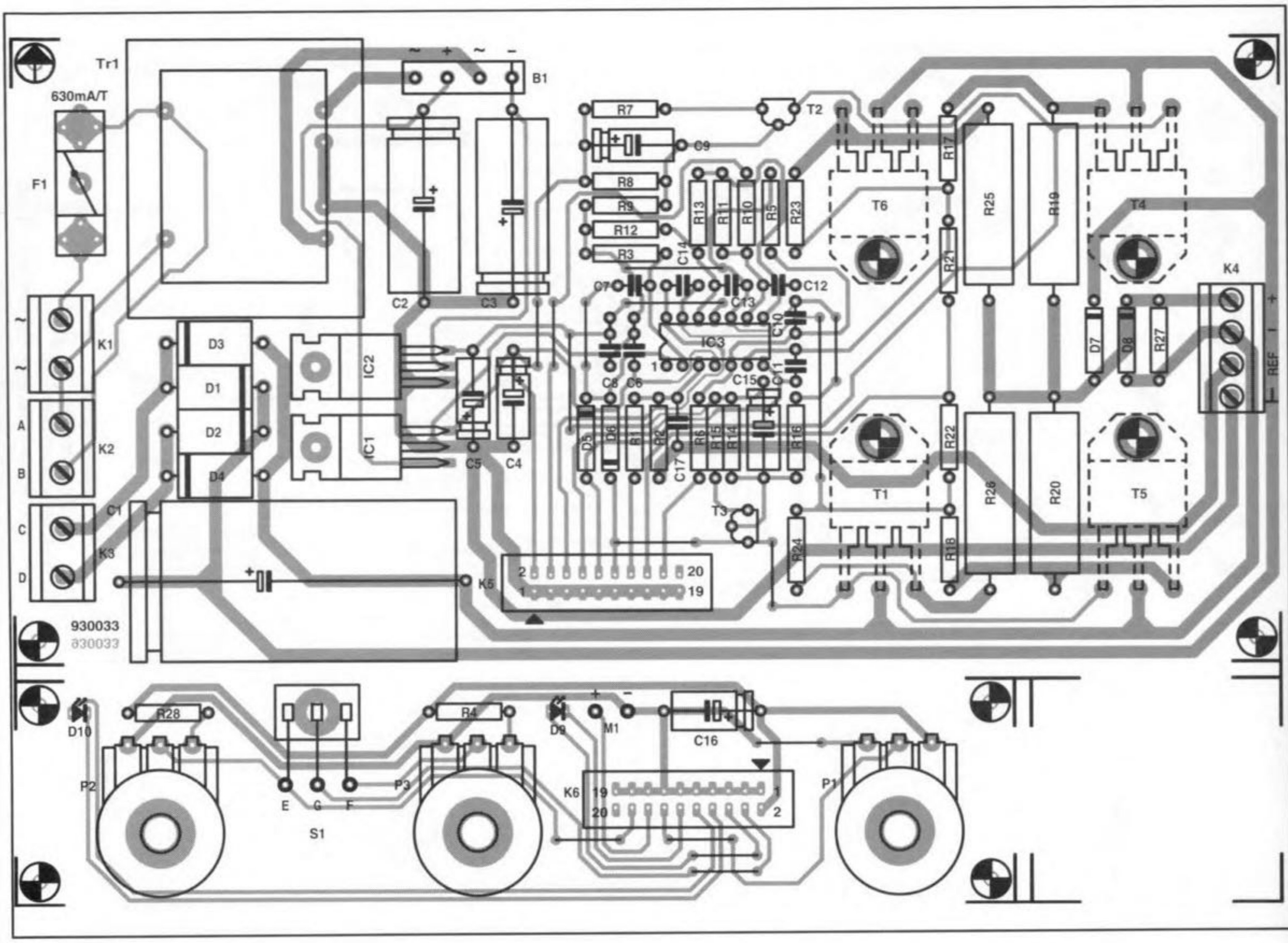


Fig. 5a. Printed-circuit board for the power supply unit – component layout.

PARTS LIST

Resistors:
R1 = 15 kΩ
R2 = 100 kΩ
R3, R17, R18, R23, R24 = 10 kΩ
R4, R28 = 3.3 kΩ
R5, R12 = 33 kΩ
R6, R13 = 82 kΩ
R7, R15 = 220 Ω
R8, R9, R14, R16 = 1 kΩ
R10, R11 = 47 kΩ
R19, R20, R25, R26 = 0.68 Ω, 5 W
R21, R22 = 68 kΩ
R27 = 4.7 kΩ
P1–P3 = 10 kΩ (multiturn)

Capacitors:
C1 = 4700 μF, 63 V
C2, C3 = 470 μF, 25 V
C4, C5 = 22 μF, 25 V
C6, C8 = 100 nF
C7 = 39 pF
C11, C12, C14 = 33 pF
C9, C15 = 10 μF, 25 V
C10, C13 = 470 nF
C16 = 47 μF, 16 V
C17 = 33 nF
C18 = 47 μF, 63 V

Semiconductors:
B1 = B80C1500
D1–D4 = FR606
D5, D6 = 1N4148
D7, D8 = 1N4004
D9, D10 = LED, 3 mm, red
T1, T5 = BDV64 (or TIP147)
T2 = BC557
T3 = BC547
T4, T6 = BDV67 (or TIP142)
IC1 = 7805
IC2 = 7905
IC3 = TL074

Miscellaneous:
K1–K3 = 2-way terminal block, 7.5 mm pitch
K4 = 4-way terminal block, 5 mm pitch
K5, K6 = 20-way box header
K7 = mains entry
S1 = single-pole change-over switch
S2 = double-pole mains switch
F1 = fuse holder with 630 mA slow-blow fuse for PCB mounting
Tr1 = mains transformer, 2×9 V, 100 mA secondary (Monacor VTR3209)
Tr2 = toroidal mains transformer, 2×15 V, 5 A secondary (Amplimo 41012)
Heat sink SK47/100 SA (Dau UK Ltd; 70–75 Barnham Road; Barnham PO22 0ES; Telephone 0243 553 031)
2 off flatcable socket, 20-way
Short length of 20-way flatcable
2 off DVM LCD module (Conrad 136026)
PCB Type 930033
Front panel foil Type 930033-F
Enclosure (Telet LC1050)

MODULE SUPPLY

Capacitors:
C1, C4, C5, C8 = 47 μF, 25 V, radial
C2, C3, C6, C7 = 100 nF, ceramic

Semiconductors:
D1–D8 = 1N4004
IC1, IC2 = 7809

Miscellaneous:
K1 = 2-way terminal block, 7.5 mm pitch
Tr1 = mains transformer, 2×9 V, 1.5 VA secondary (Monacor VTR1209)
PCB Type 920075

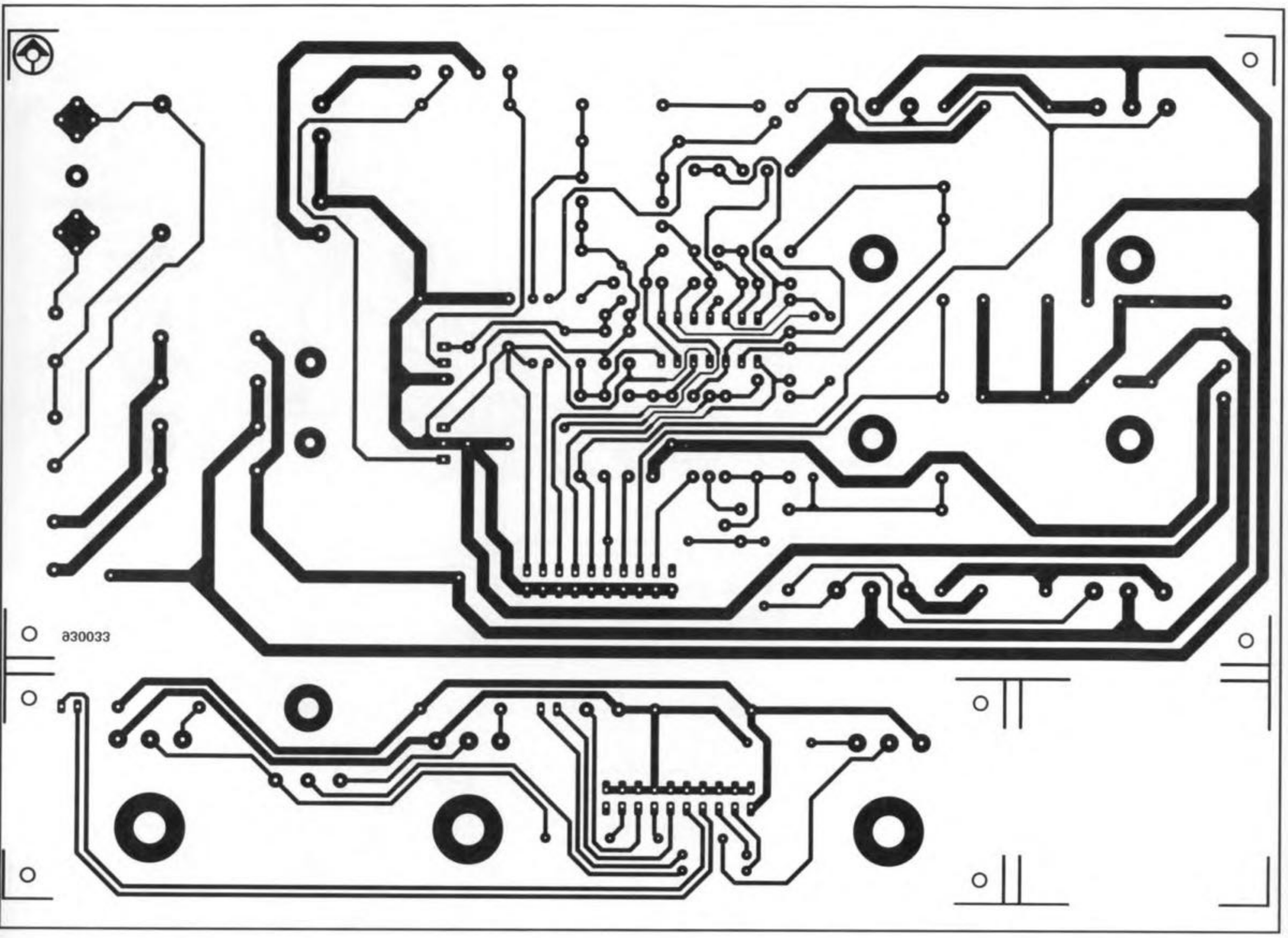


Fig. 5b. Printed-circuit board for the power supply unit – copper track layout.

controls the output voltage at K_4. With values of R_1 and R_2 as shown, the output voltage can be varied to about 33 V.

The current limiting operates roughly in the same manner, but here IC_{3b} carries out the controlling and measuring. Also, the inputs of this IC are not connected across the +ve and –ve terminals of K_4, but across a current-measuring resistance in series with the output. This resistance consists of emitter resistors R_{19} and R_{25}. Since the entire output current flows through these resistors, it is evaluated simply by measuring the voltage across the resistors and applying Ohm's law.

The measuring circuit for the current limiting is based on IC_{3b}. The inverting input of this opamp (pin 6) is linked to one side of R_{19} via R_{17} and to one side of R_{25} via R_{23}. The other input (pin 5) is connected to the current-limiting preset, P_2 (see Fig. 2). As soon as the potential across the emitter resistors rises above the level set with P_2 (corresponding to the peak current), IC_{3b} reduces, or completely cuts off, the base current to the output stage. That current is then diverted to the output of IC_{3b} via D_{10} (Fig. 2). Thus,

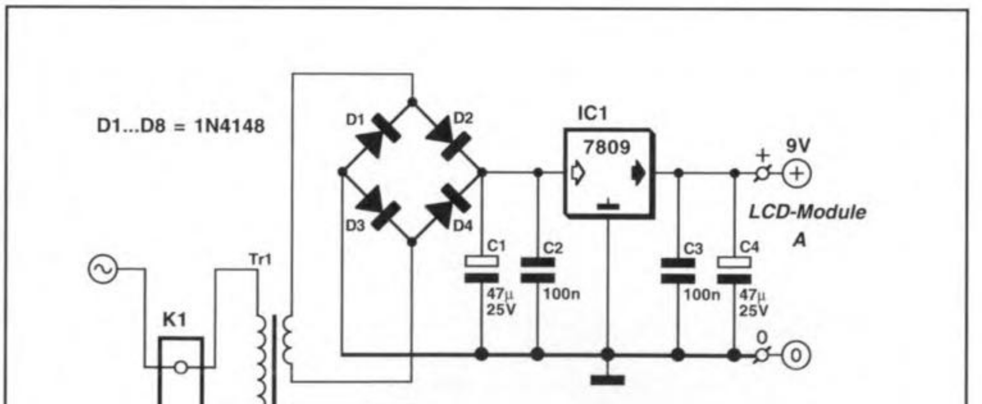


Fig. 6. Circuit diagram of the auxiliary supply for the digital meter modules.

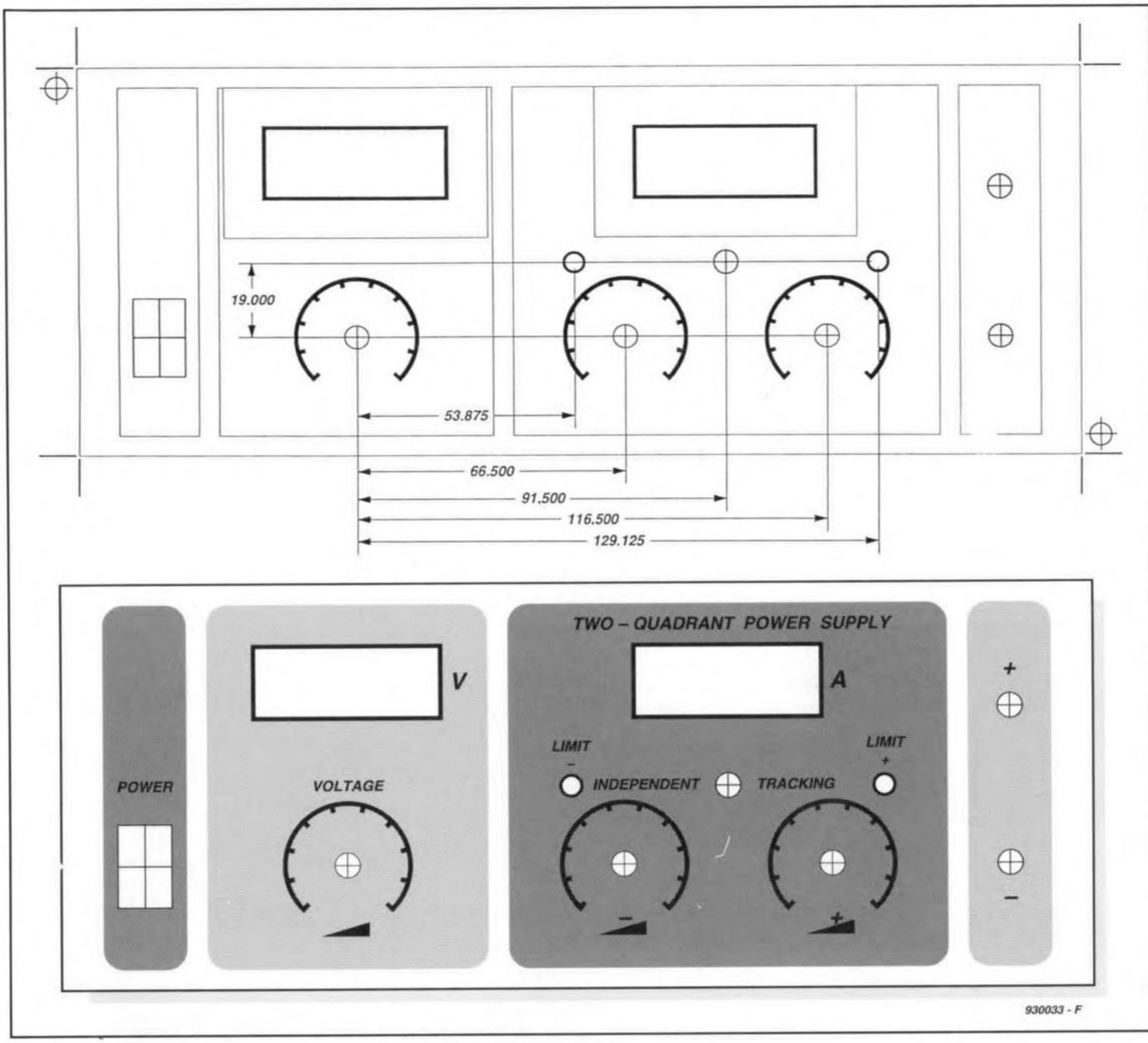


Fig. 7. Front panel design for the workbench power supply unit. Dimensions are shown in millimetres.

the lighting of this LED indicates that current limiting is in operation. With component values as shown, the maximum current that can be set with P_2 is 2.9 A.

The lower half of the diagram functions in virtually the same way as the upper half. Double darlington D_1-D_5 operates as a variable connection between the +ve and –ve terminals of K_4. The base current of this output stage is also controlled by IC_{3a}.

Constant maximum base current for the output stage is provided by T_3; opamp IC_{3d} cuts off the stage if the level of the current provided by the 'load' exceeds the set level. In this case, also, the base current of the output stage is diverted to a diode, D_9. The lighting of this LED thus indicates that the upper current level has been reached or exceeded. Since the current is an incoming, not an outgoing, one, inverter IC_{3c} is needed to provide IC_{3d} with the correct reference level.

From the foregoing description, it is clear that the diagram in Fig. 2 does not show the entire circuit: the remainder is given in Fig. 1. This part of the circuit is constructed on an ancillary printed-circuit board that is connected to the mother board by a short length of flatcable linking K_5 and K_6.

Figure 1 also contains switch S_1, which enables choosing between controlling the limits of both the outgoing and the incoming currents with one preset and controlling each of them separately

The complete wiring diagram is given in Fig. 3. This also shows the main transformer, connected to K_3, and a supply board, which is required if digital meter modules are used. The supplies to these mdules must be kept isolated from each other and from the remainder of the circuit. Diagrams of the circuit and associated PCB of the modules are given in Fig. 3 and Fig.5 respectively.

Figure 3 shows that the supply is a straightforward design, using a mains transformer with two separate secondary windings, rectifiers, smoothing capacitors and regulator ICs, that provides two isolated 9 V supply rails.

Construction

The mother board and ancillary board are shown in one piece in Fig. 6. The two parts must be separated before any work is done on them. The suggested front panel is shown in Fig. 7. The boards and front panel foil are available through our Readers' services—see page 70.

Due care must be taken in the construction of the power supplies and other mains (power line) carrying parts. Pay particuar attention to retaining safe distances between power carrying parts and the enclosure and the low-power circuits. Use heavy-duty, insulated circuit wire for the power carrying lines.

Mount power transistors, T_1, T_2, T_5, and T_6, as follows. Replace the rear panel of the enclosure by a heat sink as specified in the parts list. Drill four fixing holes for the transistors into the heat sink, using the board as template. Fit the transistors on to the heat sink with the aid of insulating washers and heat conducting paste. Bend the transistor terminals such that they fit neatly into the relevant holes in the board when this is laid on to the heat sink assembly. It is assumed here that the board has been completely populated. If all is well, fit the board on to the heat sink assembly using 10 mm (3/8 in) spacers. Then, with a slim soldering iron, solder the transistor terminals to the copper side of the board. Finally, fix the complete assembly to the enclosure.

When populating the ancillary board, note that the LEDs must be fitted to the copper side of the board.

The interwiring is shown clearly in Fig. 3.

If digital meter modules are used (built on the board in Fig. 5), note that a voltmeter is used to measure current, so that an additional resistor is required. If you lay the module in front of you on the table with its copper side up and the inputs at the right, there are some solder pads at the bottom left where a through connection must be made to enable a range to be selected. Make that connection at the second pad from the left (2 V range). Solder a 576 kΩ, ±1%, resistor between that connection and the –ve input. Then, remove the 1 kΩ resistor at the extreme right of the voltage divider. The module will then read input and output currents in A.

Voltage measurements can be taken with an unmodified module set to the 200 V range.

With reference to Fig. 3, note particularly that the through connections at K_4, as well as C_{18}, must be placed at the terminals behind the front panel, that is, C_{18} is fitted in parallel with the terminals.

Next, test the unit with open enclosure by checking whether all direct supply voltages are present.

Check that on K_4 the +ve terminal is connected to ground and the –ve terminal to REF.

Turn the voltage control and check that the voltmeter module shows a voltage.

Then, connect a suitable load to the output and check that the ammeter module reacts correctly.

Check whether the current limiting is actuated when the aappropriate preset is turned down.

If all these checks and tests are satisfactory, carry them out again with higher voltages and currents. Use a suitable rated load resistor.

If everything is still all right, check that the current-sink section operates correctly. For safety's sake, connect a resistor in series with one of the supply terminals: this will function as an emergency current limiter if something is disastrously wrong.

Connect a variable voltage source to K_4 and increase its output gradually.

Check that the current limiter circuit is actuated when the relevant preset is turned.

Then, remove the series resistor and check the correct action of the current limiting preset again.

If all is correct, the enclosure can be closed, whereupon the workbench power supply unit is ready for use. ■

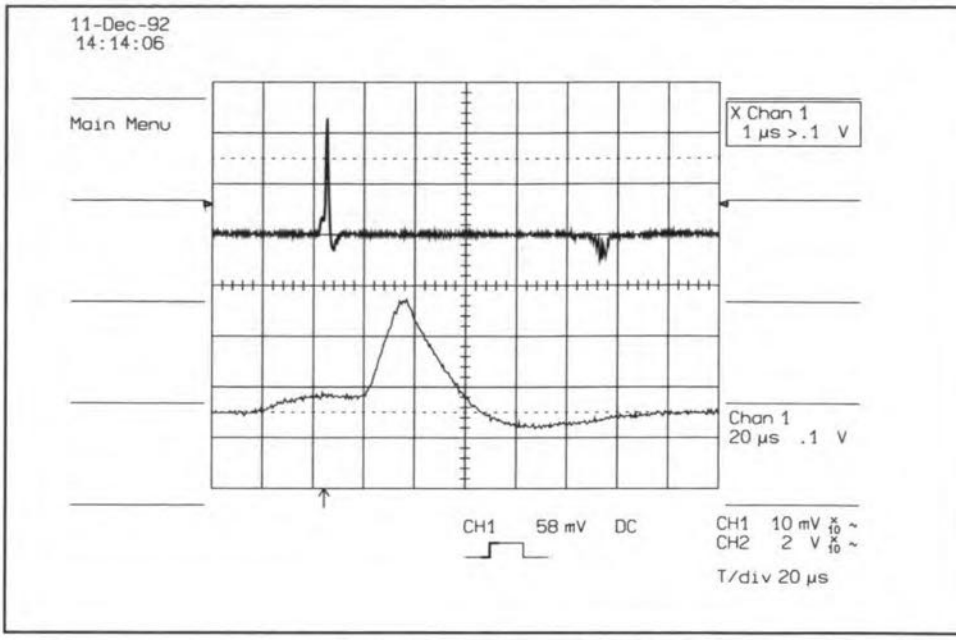
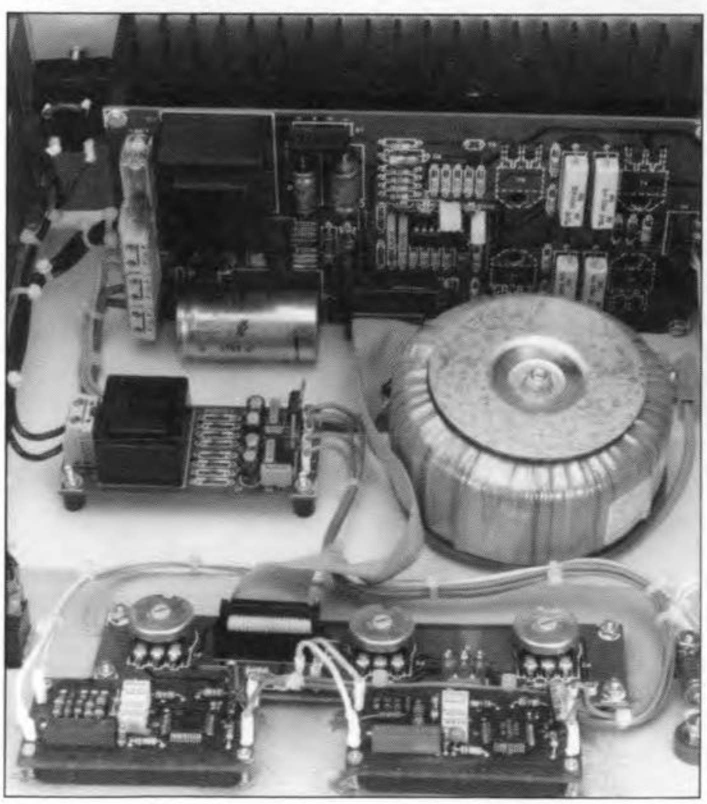
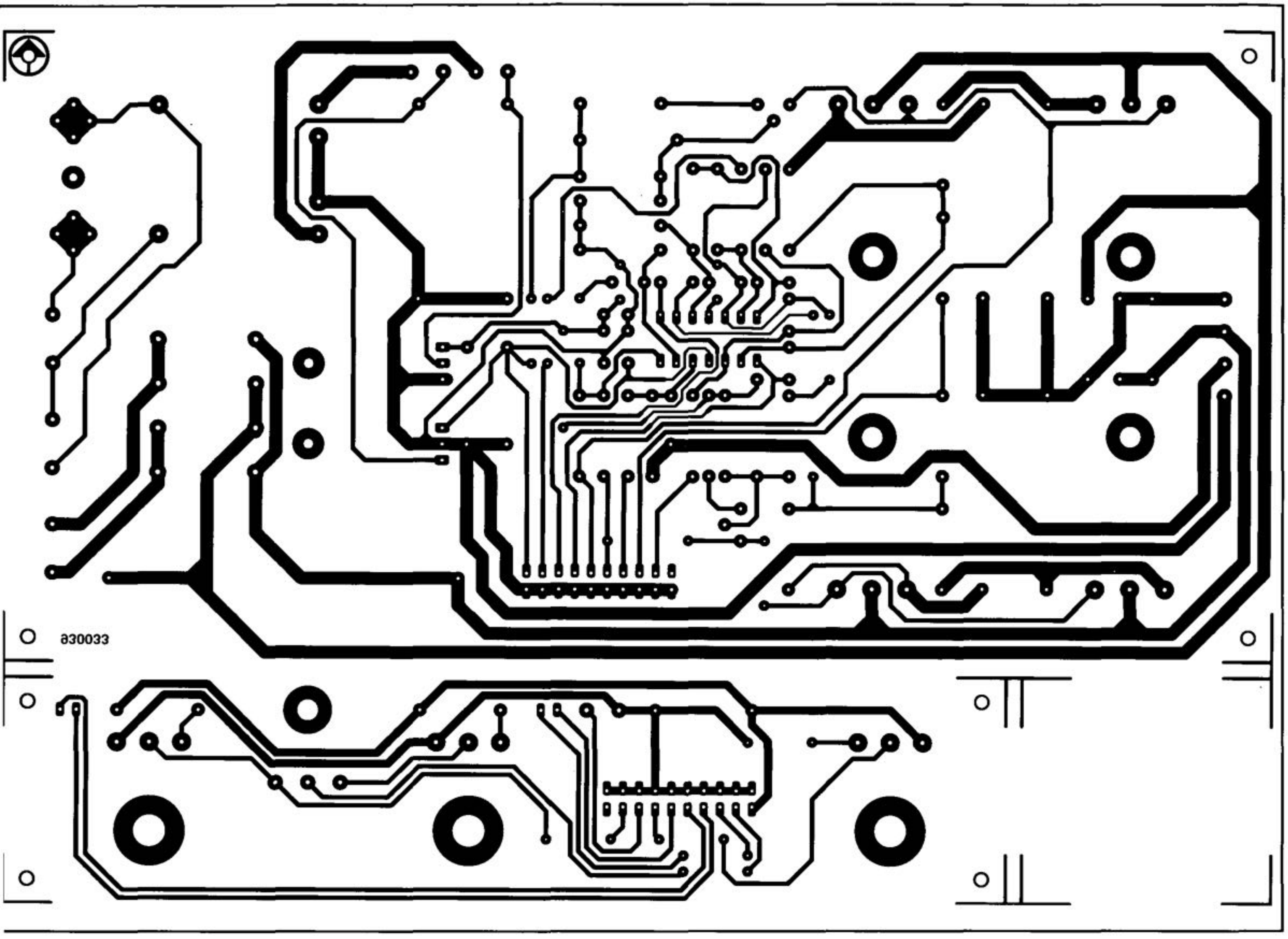


Fig. 8. The measured dynamic performance of the power supply unit. The output current was a square-wave at a level of 2 A. The peak in the upper curve shows the reaction to the switching off of the current, while the dip shows what happens at switch-on. The lower curve shows the peak magnified.

Brief Technical Data

Output voltage continuously variable between 0 V and 30 V
Peak output current 3 A
Peak sink current 3 A
Separately presettable current limiting for output and sink currents
Current limiting indication by LED
Switch for independent/tracking selection.




a30033

CORRECTIONS AND UPDATES

Digital Audio/visual system (Multi-purpose Z80 card)

May and June 1992
An extensive description of a modification to the memory backup circuit on the Multi-purpose Z80 card is available free of charge through our Technical Queries service.

FM stereo signal generator

May 1993
Capacitors C17 and C19 should have a value of 33nF, not 3nF3 as indicated in the circuit diagram and the parts list of the multiplex generator.

Workbench PSU

May 1993
The polarity of capacitor C15 is incorrectly indicated on the PCB component overlay (Fig. 5a), and should be reversed. The circuit diagram (Fig. 2) is correct.

Transformer TR2 is incorrectly specified in the circuit diagram (Fig. 2) and in the parts list. The correct rating of the secondary is 2 × 12V/5A. Also note that the secondary windings are connected in series to give 24 V.

Audio DAC

September 1992
The polarity of capacitors C25 and C58 is incorrectly indicated on the component overlay of the D-A board (order code 920062-2), and should be reversed.

U2400B NiCd battery charger

February 1993
The value of resistors R17 through R27 should be 2.7kΩ, not 12.7kΩ as stated in the parts list.

VHF/UHF receiver

May 1993
In Fig. 4, the connections to ground of the AF amplifier outputs, pins 5 and 8, should be removed. The amplifier outputs are connected to the loudspeaker only. The relevant printed circuit board is all right.

FIGURING IT OUT

PART 5 – INDUCTOR MATHS

By Owen Bishop

This series is intended to help you with the quantitative aspects of electronic design: predicting currents, voltage, waveforms, and other aspects of the behaviour of circuits. Our aim is to provide more than just a collection of rule-of-thumb formulas. We will explain the underlying electronic theory and, whenever appropriate, render some insights into the mathematics involved.

Inductors are in many ways the converse of capacitors. An obvious difference is that a capacitor has an insulating dielectric between its plates, so that no conduction is possible. In actuality, it is an open circuit. By contrast, the coil of an inductors is often (though not necessarily) of extremely low or negligible resistance. It is virtually a closed circuit. There is a special term in electronics for this sort of correspondence between two features of properties of a circuit or part of a circuit. The members of such a pair are called **duals** of each other. With respect to the ability to conduct a current, an open circuit is the dual of a closed circuit. To this extent, an inductor is the dual of a capacitor. The duality of inductors and capacitors extends to far more than the physical ability to conduct a current. This month we contrast the two types of component and the mathematics associated with them.

Although inductors as such are not found in electronic circuits as often as they used to be (being replaced by opamps in active filters, for example), most components have a certain amount of inductance. The self-inductance of the leads of a capacitor or resistor becomes of major importance in high-frequency circuits, which is why lead-less surface-mount resistors and capacitors are so often used in such applications.

A recap on caps

We dealt with several aspects of capacitor maths in Part 2, but will briefly quote the main points again so that the duality of the inductor may be better understood. The basic equation for capacitors is that by which the farad is defined:

$$C = Q/U, \qquad \text{[Eq. 19]}$$

where C is the capacitance in farad, Q is the charge in coulomb, and U is the pd between the plates in volts. (This equation was also given unnumbered in Part 2).

From Eq. 19, we derive an equation of the form

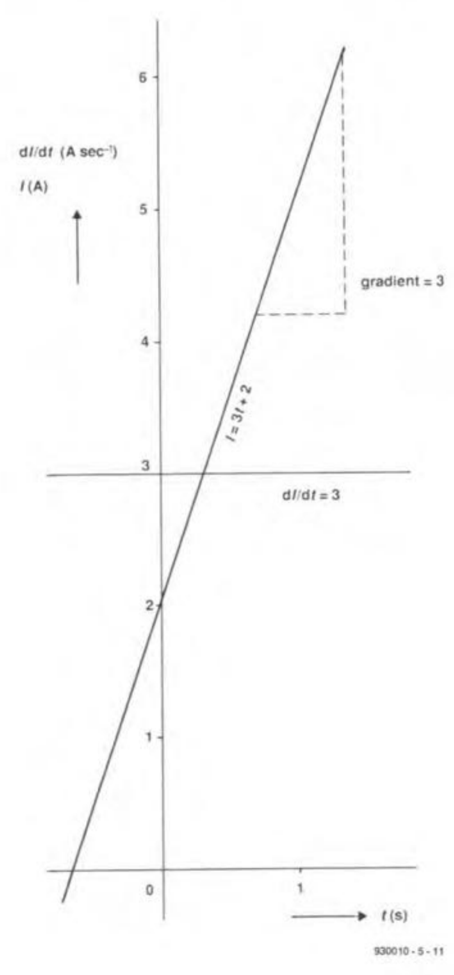
$$Q = \int I\,dt + c \qquad \text{[Eq. 20]}$$

where c is the initial charge, if any. This equation was used in several examples in Part 2 to calculate charge on a capacitor in various charging conditions. Having found Q, we can easily calculate U from Eq. 19:

$$U = \frac{1}{C}\int I\,dt + \frac{c}{C}. \qquad \text{[Eq. 21]}$$

An important property of a capacitor is that, if a periodic current is supplied to it (typically a current that varies sinusoidally), the charge on the capacitor and the voltage across it also vary sinusoidally. The charge or voltage curve is a negative cosine curve, or we can think of it as a sine curve that lags behind the current curve by $\pi/2$ rad (90°).

38



Inductor equations

The basic equation which defines the henry, the unit of inductance, is

$$L = \frac{N\Phi}{I}. \qquad \text{[Eq. 22]}$$

Referring to a single coil, its self-inductance in henry is the flux linkage (the magnetic flux ϕ in weber multiplied by the number of turns N, since the flux threads through each turn), when a unit current flows through it ($I = 1$ A). A more useful definition may be derived from Eq. 22, using Neumann's equation. This states that:

$$U = N\frac{d\Phi}{dt}. \qquad \text{[Eq. 23]}$$

What this equation says is that the e.m.f. induced in a coil is proportional to the **rate of change** of flux linkage. The faster the flux changes, the greater the induced e.m.f. For example, if the flux through the coil is increasing at the rate of 2 mWb s^{-1}, and the coil has 25 turns, then $U = 25\times2 = 50$ V. The direction of the e.m.f. is such that it opposes the change in flux according to Lenz's Law. This fact is sometimes indicated by preceding the right side of the equation with a negative sign but, provided that we state that U is the **back e.m.f.**, the sign may be omitted.

We shall take up this discussion again later, but next we need to examine the equation in more detail.

Differentials

The expression $d\Phi/dt$ in Eq. 23 represents what we have termed the **rate of change** of flux. The expression is known as the **differential** of Φ with respect to t, and a whole branch of mathematics is devoted to dealing with expressions of this kind. The expression has the form of one quantity ($d\Phi$) divided by another quantity (dt). Differentials are used to relate various pairs of quantities: dy/dx, for example, relates quantities x and y. If a graph is plotted with values of x along the horizontal axis and values of y along the vertical axis, dy/dx represents the steepness of the slope, or **gradient**, of the graph. If U is the gate-source voltage of a FET, with I being the corresponding drain current, then dI/dU specifies the amount by which I changes for given changes in U. In other words, dI/dU is the **transconductance**. In the context of inductors, $d\Phi/dt$ is the rate of change of flux with time.

Calculating a differential is the inverse operation to integration, which we described in Part 2. There is not enough space here to go into the theory of differentiation, but we can set out a table of frequently used expressions and their differentials.

Standard differentials

Function of I	Differential dI/dt
at^n	ant^{n-1}
e^{at}	ae^{at}
$\sin(at+b)$	$a\cos(at+b)$
$\cos(at+b)$	$-a\sin(at+b)$

The table uses I and t as the variables in place of the customary x and y, so that its contents are more directly applicable to electronics. For example, given that $I = 3t+2$, the way I varies with time is shown in the graph of Fig. 38. To find the rate of change of I at any given time, we calculate dI/dt, using the table. The first expression in the function is $3t$ or, to put it in the form of the first line of the table, $3t^1$. If $n = 1$, the differential is $3t^{1-1} = 3t^0$. Since $t^0 = 1$, the differential is 3. The second expression is 2, which can be written as $2t^0$. This makes $n = 0$, and the value of the differential is zero. The differential of a constant is always zero. Putting these two together, we have

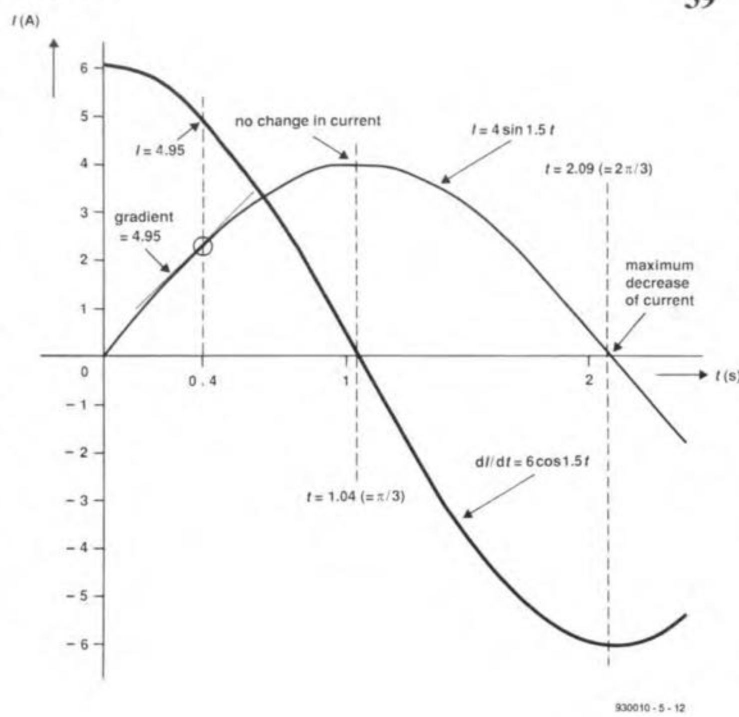
$$dI/dt = 3.$$

In this example the differential does not contain t, so the differential is independent of time. In other words, the slope or gradient of the curve in Fig. 38 is 3 at all instants of time.

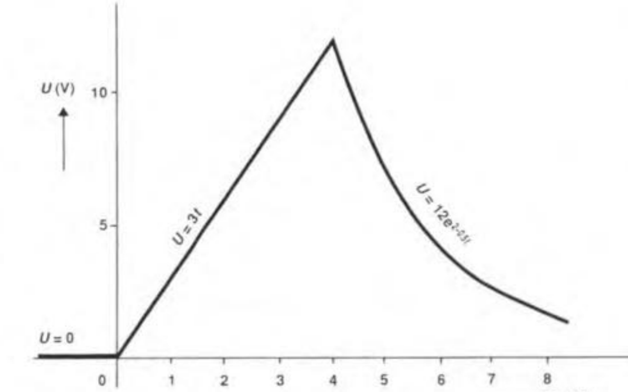
Figure 39 shows the curve for another function, $I = 4\sin 1.5t$. Referring to the table, $a = 1.5$ and $b = 0$ in this function. According to the table, $dI/dt = 4\times1.5\cos 1.5t = 6\cos 1.5t$. This tells us the rate of change of I at any instant. For example, at $t = 0.4$, $dI/dt = 6\cos 0.6 = 4.95$. This can be seen in Fig. 39, where the curve for $6\cos 0.6t$ has the value 4.95 when $t = 0.4$. It can also be seen that the slope of the curve $4\sin 1.5t$ when $t = 0.4$, as shown by the tangent to the curve at that point, is 4.95 (taking account of the different scales on the two axes). When $t = \pi/3$, a point at which the current is at its maximum and is neither increasing nor decreasing, then $dI/dt = 6\cos(1.5\times\pi/3) = 0$, corresponding to zero change of current. Conversely, when $t = 2\pi/3$ and the current is changing at maximum rate from positive to negative, $dI/dt = 6\cos\pi = -6$, its maximum negative value, showing maximum decrease of current.

The practical advantage of differentials is that they make it possible to calculate rates of change for functions much more complicated than $I = 4\sin 1.5t$, and without the need to plot the curves. We shall be looking further into this in future issues.

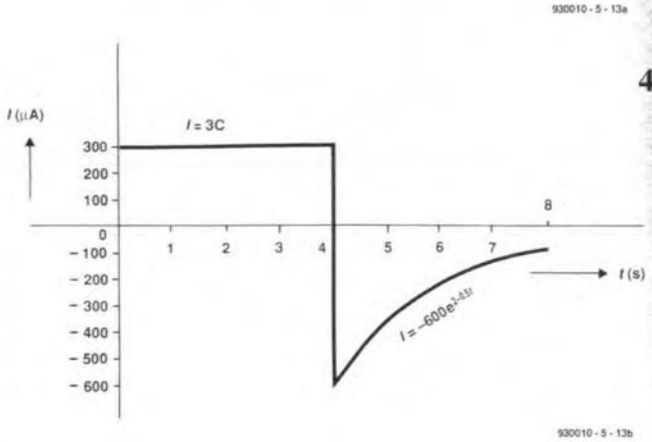
39



40a



40b



Back to induction

Rearranging the terms of Eq. 22 gives

$$\Phi = \frac{L}{N}\cdot I, \qquad \text{[Eq. 24]}$$

where L and N are constant, but Φ and I may both vary with time. Differentiating both sides of this equation with respect to time:

$$\frac{d\Phi}{dt} = \frac{L}{N}\cdot\frac{dI}{dt}. \qquad \text{[Eq. 25]}$$

Writng this equality into Eq. 23, we obtain

$$U = N\cdot\frac{L}{N}\cdot\frac{dI}{dt} = L\frac{dI}{dt}. \qquad \text{[Eq. 26]}$$

This is an alternative way of defining the henry. In words, in a coil of self-inductance 1 H, a change of current of 1 A s^{-1} produces a back e.m.f. of 1 V.

Given a function which determines how a current varies with time, Eq. 26 allows us to calculate the back e.m.f. at any instant. Here are some examples.

A $I = 3$ (a constant current of 3 A); $dI/dt = 0$. There is no change of current so the back e.m.f. is zero.

B $I = 2t$ (a ramp current); $dI/dt = 2$ A s^{-1}. Given a coil with $L = 0.25$ H, the back e.m.f. is constant at $0.25\times2 = 0.5$ V.

C $I = 4\sin 1.5t$ (the sinusoidal current given in an earlier example; $dI/dt = 6\cos 1.5t$. Given a coil with $L = 0.1$ H, the back e.m.f. at any instant is $0.6\cos 1.5t$. The back e.m.f. follows the differential curve of Fig. 39, but with amplitude 0.6 V instead of 6 A.

D We leave until a later issue the interesting case of what happens when a voltage is applied to a coil instantly by closing a switch.

Duals again

Equations 21 and 26 are the voltage equations of a capacitor and an inductor respectively:

$$U = \frac{1}{C}\int I\,\mathrm{d}t + \frac{\mathrm{c}}{C}. \qquad \text{[Eq. 21]}$$

$$U = L\frac{\mathrm{d}I}{\mathrm{d}t}. \qquad \text{[Eq. 26]}$$

They tell us the voltage produced across the device by a current varying in time according to a given function.

Let us see what happens if we **differentiate** both sides of Eq. 21 with respect to time:

$$\frac{\mathrm{d}U}{\mathrm{d}t} = \frac{1}{C}\cdot I$$

Since differentiation and integration are inverse operations, differentiating $\int I\,\mathrm{d}t$ undoes the integration, leaving simply I. The differential of the constant c/C is zero. Rearranging terms gives:

$$I = C\cdot\frac{\mathrm{d}U}{\mathrm{d}t}. \qquad \text{[Eq. 27]}$$

This tells us the current flowing into or out of a capacitor when the applied voltage varies according to a given function. It is just a mathematical way of stating the obvious: that the current flowing into (or out of) a capacitor equals the capacitance multiplied by the rate of increase (or decrease) of pd.

The interesting point is that this **current** equation for the capacitor has the same form as the **voltage** equation (26) of the inductor. Capacitance and inductance are duals. Likewise, pds and currents (and also rates of change of pd and current) are duals when applied to capacitors and inductors.

To complete the quartet of equations, we **integrate** both sides of Eq. 26 with respect to time:

$$\int U\,\mathrm{d}t + c = LI.$$

The integral of $\mathrm{d}I/\mathrm{d}t$ is I. Rearranging terms gives

$$I = \frac{1}{L}\int U\,\mathrm{d}t + \frac{\mathrm{c}}{L}. \qquad \text{[Eq. 28]}$$

This has the same form as Eq. 21, again illustrating the dual nature of capacitance and inductance, and of pd and current.

We now have a set of four equations for solving problems associated with capacitors and inductors. Examples using Eq. 21 and Eq. 26 have already been given. Eq. 27 can be used to solve problems such as the following.

A voltage pulse applied to a 100 µF capacitor has the shape shown in Fig. 40a, and is described by this piecewise function:

U=0	$t \le 0$
U=3t	$0 < t \le 4$
U=12e$^{2-0.5t}$	$4 < t$.

Calculate the current flowing to or from the capacitor and plot its graph.

Solution: for the first stage, using Eq. 27:

$$I = C\cdot\frac{\mathrm{d}U}{\mathrm{d}t} = 0.$$

For the second stage, when U is increrasing at a constant rate:

$$I = C\cdot\frac{\mathrm{d}U}{\mathrm{d}t} = 3C = 300\,\mu\mathrm{A}.$$

This is a constant charging current.

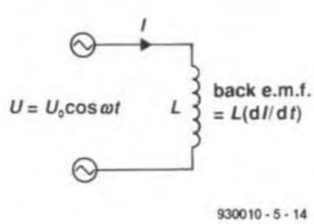
For the third stage, when U is decreasing exponentially:

$$\begin{aligned} I &= C\cdot\frac{\mathrm{d}U}{\mathrm{d}t} \\ &= 12C(-0.5\mathrm{e}^{2-0.5t}) \\ &= -600\mathrm{e}^{2-0.5t}\,\mu\mathrm{A}. \end{aligned}$$

Current flows away from the capacitor at an exponentially decreasing rate. The flow of current is plotted in Fig. 40b.

Phase relationships

41



In Fig. 41, an alternating pd is applied to an inductor. At any instant, the instantaneous applied pd is:

$$U = U_0\cos\omega t, \qquad \text{[Eq. 29]}$$

where U_0 is the peak value of the pd. Treating the circuit as a loop, by KVL in a clockwise direction:

$$U_0\cos\omega t - L\cdot\frac{\mathrm{d}I}{\mathrm{d}t} = 0,$$

$$\therefore\ \frac{\mathrm{d}I}{\mathrm{d}t} = \frac{U_0}{L}\cdot\cos\omega t.$$

Integrating both sides of this equation:

$$I = \left(\frac{U_0}{\omega L}\right)\sin\omega t.$$

But, given that the current is sinusoidal, the expression in brackets must be equal to the peak current, I_0:

$$I = I_0\sin\omega t. \qquad \text{[Eq. 30]}$$

A comparison of Eq. 29 and 30 shows that I and U have the same frequency and that I lags behind U by $\pi/2$. Their curves are plotted in Fig. 42.

The relationship for the inductor contrasts with that of the capacitor as described in Part 2. In the inductor, the current lags the pd by $\pi/2$, but in the capacitor it **leads** the pd, another manifestation of their duality.

Test yourself

1. The current in A at time t s through an inductor, L = 0.1 H, is I = 3–0.5t. Calculate the back e.m.f. and state its direction.

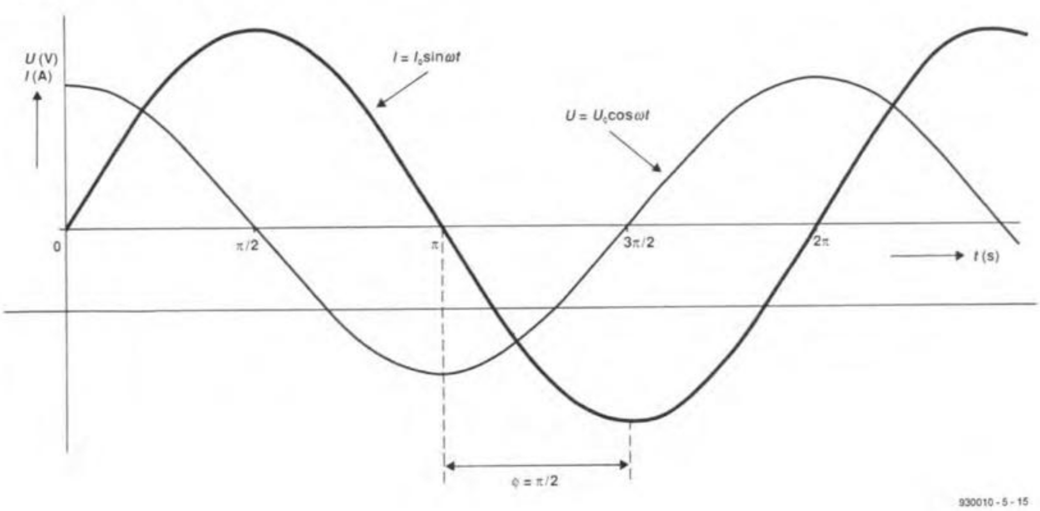
2. The current in A at time t s through an inductor, L = 0.2 H, is I = 1–e^{-2t}. What is the back e.m.f. after 2 s?

3. The voltage in µV at time t across an inductor, L = 100 µH, is defined by
   $U = 0 \quad t \le 0$
   $U = 2t \quad 0 < t$.
   Calculate the current through the inductor after 3 s.

Answers to
Test yourself (Part 4)

a. –0.032 A (–1/31)
b. 0.259 A (22/85)
c. 0.250 A (1/4)
d. –0.166 A (–112/676)

42



Author's note

The last paragraph of Part 2 is misleading, since it implies that the phase lag of a low-pass RC filter is **always** 90°. This assumes that the reactance of the capacitor is infinite, which is the practical situation at high frequencies. Although output lags input at any frequency, as implied in that paragraph, the phase angle is less than 90° at lower frequencies. I am sorry if I have mislead the reader. The action of filters will be discussed in more detail in next month's issue.

A progressive and holistic design for an economical soundwall

By S. Courtney

Look at a pair of dynamic high-quality headphones and you will see that the 'phone' is actually almost identical to a very small loudspeaker cushioned and placed close to the ear.

Thus, we can infer that the frequency response of a loudspeaker depends on the amplitude of the signal and also that an entire wall of 'headphone loudspeakers' operating at normal volume should make a pleasing sound.

There is a tension between the requirement for good high-frequency (treble) response, which requires a small moving cone-mass so that the accelerations required do not absorb too much energy, and that for good low-frequency (bass) response for which a large-diameter cone is needed. This tension is, of course, the reason for the traditional cross-over network to feed separate tweeters, howlers, woofers, and subwoofers.

Uniting the requirements in one device

Evidently, at very low sound levels a suitably mounted small loudspeaker will provide reasonably good sound reproduction. These levels are far too low for the human ear, however, and thus we must use a number of small loudspeakers to produce adequate sound levels. In this way, the two requirements outlined earlier can be met: the soundwall becomes possible. Buy a lage number of small, inexpensive loudspeakers, mount them close together on a baffle and, for what turns out to be a very modest outlay, a quality speaker system can be constructed. This array, square with columns parallel connected and whole rows in series, turns out in operation to have the properties of a very efficient loudspeaker system that does not require any auxiliary units, except a subwoofer. Not only is the efficiency quite remarkable, but only low power is required from the amplifier.

The reason for the very good efficiency is that the cone movement in each loudspeaker is very small: displacement is smaller than one per cent of the limits of movement. Therefore, all the work is done in the region where very nearly all the energy is used in moving the voice coil and cone, rather than working against mechanical impedance.

A baffle must be used, of course, because the speaker response is much affected by the surround: pressure waves tend to 'nip around' an unbaffled speaker to the back. Thus, in a region of the audio spectrum where pressure changes are greatest, the degradation of performance is most marked. This is, of course, also true in the proposed design, but only inasfar the very low frequencies are concerned.

Removing bandwidth restrictions

At very low sound levels, small drive units are excellent. non-specialized tweeters. Nevertheless, a purist designing a project like this may wish to shop around for the best type and not consider the 60 mm diameter units (cost 90 pence or 50 ¢) used for the prototype for this article.

Since the drive units are mounted close together on the baffle, the low-frequency bandwidth limitation is overcome to a large degree. This is because the individual cone movements are very small and not constrained by the movements of the suspension. As a large number of drive units is required, ceramic models are preferable. In this context, job lots of inferior drive units need not be ignored, because the extent to which the very low levels of operation broaden the frequency response is not predictable and depends very much on the individual units. Here then is a nice field for experimentation: the reader is invited to try with about 280 drive units for a soundwall at whatever bargain price can be had. Wholesale purchase would seem the best way, but sometimes retailers wish to sell off old stock!

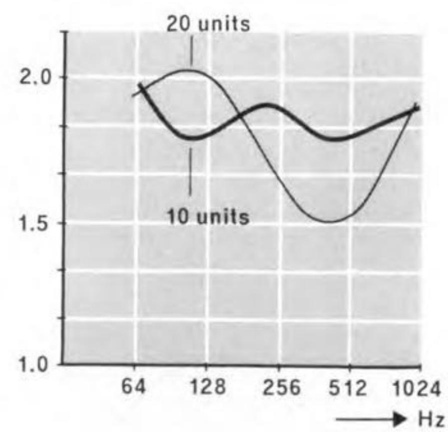
What advantages?

1. **Efficiency**. The cone movements are very small so that the cone meets little non-linear mechanical impedance. Consequently, hardly any energy is lost.
2. **Large area**. The proposed arrangement is ideal for a soundwall, which is much less conspicuous and obvious than large speaker cabinets that many women find objectionable in their living area. Four inches of foam behind the units is an adequate amount of 'depth' with this design.
3. **Linear cone movemen**t. With a large in-phase array, the sound has many of the properties of a linear-phase speaker system, which is often said to embody the most rational approach to hi-fi principles.
4. **Small depth**. With the addition of a subwoofer (disguised as a cupboard), a one-brick thicker wall can be a soundwall, thus achieving the goal(with a suitably decorated grill on the flat unit) of a space-saving and non-intrusive sound source.
5. **Compatibility with room acoustics**. This system produces a sound that blends with room acoustics and is not shadowed by furniture in the room.
7. **Compatibility with room decor**. Hi-fi cabinets, normally bought by men, tend to have an assertive design because of the male psychology. Left to a woman, the design would not be intrusive. The soundwall harmonizes with the home, providing domestic peace and a solution that blends in with the decor.

Construction

Assess the area that has to be covered and buy a baffle suitable for that area. Cut the holes for the speakers as appropriate. Note that an hexagonal pattern is better than rectangular rows and columns that tend to set up resonances.

Buy as many speakers as initially possible to fill the baffle. The curve below shows how the bass response improves when the number of drive units is doubled from ten to twenty.



Over the range 64–1024 Hz, the maximum-minimum for a curve, which we shall call the variation, is roughly halved when the number of drive units is doubled. In fact, the variation of frequency response with this small number of drive units drops from 5.6 dB to 2.6 dB when the number of units is doubled.

Once the drive units are mounted on the baffle, construct a 2–4 in (5–10 cm) deep rectangular faced cabinet behind the baffle. Fill the space in the cabinet with sheets of foam, laid one on to the other and tacked into place. The larger depth is preferable, but 2 in (5 cm) should be acceptable with a very large number of drive units.

Wire the N^2 drive units as N series 'blocks', each of N drive units in parallel. The impedance will then be electrically the same as that of a single speaker.

Complete the structure to personal taste, after which it should look superficially like a decorative wall panel. If the finish matches the room decor and colour, a virtually invisible, unobtrusive sound system results.

Listening

Like all linear-phase systems, the soundwall seems low on bass. The overall response can be improved by the addition of a subwoofer (mounted in a cupboard with a gap at the bottom of the door). Since the high frequency sound beams straight out of the small speakers, the best response is obtained by facing the soundwall; to either side there is a slight loss of top.

The high-frequency response given this cavil is nicely uncoloured and natural; the purist can mount a single tweeter disguised as some kind of ornament above each wall to correct this effect.

EXTERNAL FERRITE AERIAL UNITS FOR SHORT, MEDIUM AND LONG WAVE RADIOS

By Richard Q. Marris, G2BZQ

Over the years, the writer has often needed handy, external aerials for short, medium and long wave radio receivers, such as communication, vintage and modern domestic models. This involves a wide variety of aerial input impedances with, perhaps, coaxial, twin-feed or simple end-fed aerial/earth sockets. Furthermore, these aerial units should be capable of being taken to the location of the radio, rather than the more usual method of taking the radio to the aerial. The immediate target was a frequency range of 150 kHz (2000 m) up to, and including, the 15.10–15.45 MHz (19 m) band.

In the event, it was possible to produce two simple ferrite-rod-based circuits covering [1] 150 kHz (2000 m) continuously up to 2143 kHz (140 m), and [2] 3333 kHz (90 m) to 16 MHz (16.7 m). Furthermore, simple impedance matching arrangements enable them to be used with a wide range of receiver input impedances, whether coaxial, twin-feed or end-fed.

The short-wave unit

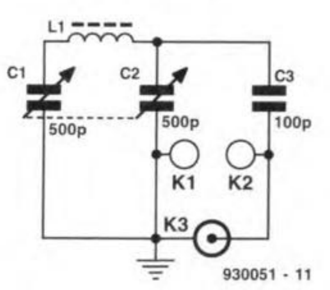


Figure 1.

Figure 1 shows the short-wave circuit that is based on a 6¼ in (158 mm) long by ½ in (12.5 mm) dia. ferrite rod made of high-grade nickel-zinc Type 61 material. The rod is cut from a standard 7½ in (190 mm) long by ½ in (12.5 mm) dia. rod.

Twelve turns of SWG16 (AWG14) wire are wound on to the rod to form coil L. The coil is tuned by a 2-gang, 500+500 pF, variable capacitor in a balanced circuit. Various methods of connection to receivers were tried, and the simple C coupling was found to be the best to cover ther wide frequency range and impedance requirements.

The unit can be coupled to receivers with a coaxial input via socket K_3 and to receivers with a twin-feed or single input via sockets K_2 and K_1.

In operation, L is resonated by C_1-C_2 to the receiver frequency for maximum signal. Since the unit is narrow-band and directional, it eliminates much of the inter-station and man-made interference. Maximum signal is received when the long side of the aerial rod is towards the received station, and the minimum signal is at the ends of the rod. The polar diagram is a figure of 8.

The frequency range on the prototype is 90 m (3333 kHz) to 16.7 m (18 MHz). No preamplifier has been required with the high-gain receivers used, but, if necessary, a small one may be placed between the aerial unit and the receiver, or even built into the aerial unit.

Long and medium wave unit

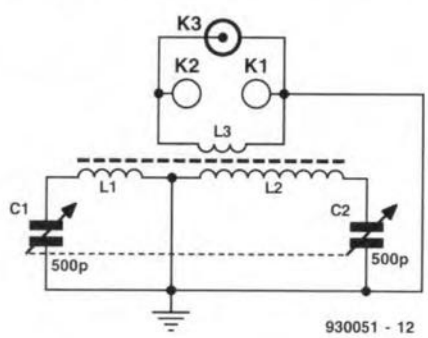


Figure 2.

The circuit, shown in Fig. 2, is based on a standard commercial MW/LW ferrite aerial rod (L_1/L_2) that can be salvaged from an old domestic radio or purchased from a number of suppliers. As will be shown later, in the construction details, the MW and LW coils, L_1 and L_2 respectively, are moved outwards towards the ends of the rod, whereupon a coupling winding, L_3, is wound near L_2. Here again, a 2-gang, 500+500 pF, variable capacitor is used to tune the circuit: C_2 tunes L_1 and C_2 tunes L_2 to provide continuous coverage from 150 kHz (2000 m) to about 2143 kHz (140 m). This range includes that interesting frequency spectrum between the medium and long wave band and has a tuning overlap near the low-frequency end of the medium wave band.

Again, socket K_3 is for use with receivers that have a coaxial input, whereas K_1 and K_2 are for use with twin-feed or end-fed inputs with an earth connection.

Since the unit is narrow-band and directional—its polar diagram is a figure of 8—it appreciable reduces, or even removes, the mass of interference met during the hours of darkness.

Construction

General. A straighforward, basic form of construction layout has been adopted for both the short wave and the long/medium wave units—see Fig. 5.

An all-plastic box (children's sandwich box with hinged lid), about 6½×4½×2 in (165×115×50 mm) (L×W×D) box was used for the prototypes. The dimensions and the type of box are not very important, however. The 2-gang variable capacitor is bolted into the box, after any preset padding capacitors have been removed. The capacitor used had a built-in slow-motion drive. If a different type of variable capacitor is used, there is plenty of room for a separate slow-motion drive to be added.

Three sockets are fitted at the rear of the box. A complete set of inter-connecting feedlines for use between the aerial units and receivers consists of:

1. A 3-feet long RG58 coaxial cable with suitable coaxial plugs at either end.
2. A 3-feet length of twin feedline with 4 mm banana plugs at one end and matching plugs at the other end. Twin loudspeaker extension cable is a good compromise. This feedline can also be used with receivers that have an input for end-fed aerial wires and earth.

Short wave unit. The general layout is shown in Fig. 6. The unit uses the basic layout of Fig. 5 plus a fabricated ferrite cored coil L (see Fig. 3) and capacitor C_3.

The construction of coil L is shown in Fig. 3. Cut a standard 7½ in long by ½ in dia. (190×12.5 mm) ferrite rod (Amidon Type R61-050-750) to length with a sharp hacksaw blade (Fig. 3A). Cut a length of ⅝ in (16 mm) dia. plastic pipe to the same length (Fig. 3B). Insert the ferrite rod into the tube, with turns of masking tape at either to ensure core rigidity (Fig. 3C). Note that it is **essential** to use the specified type of ferrite rod.

Wind 12 turns of SWG16 (AWG14) tinned copper wire on to a 9/16 in (14.5 mm) dia. former, slip the winding on to the assembly of Fig. 3C and evenly space it out to 2⅜ in (60 mm) as shown in Fig. 3D.

Mount the inductor in the plastic box on plastic stand-off wall clips as shown in Fig. 6 and then wire up the unit as illustrated in Fig. 1. An end-of-coil tap as in Fig. 6 enables the winding to be reduced by a turn or thereabout, if neces-

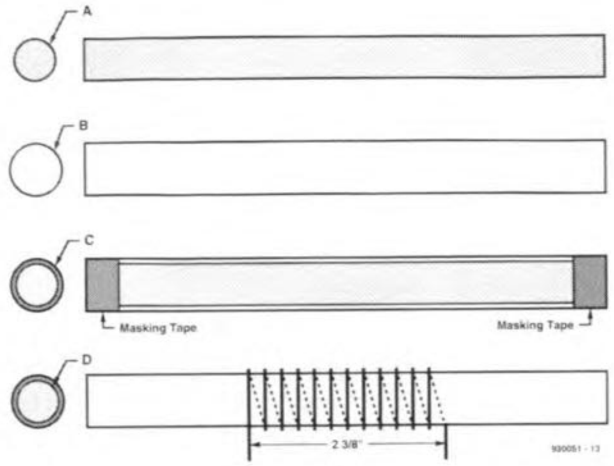


Figure 3.

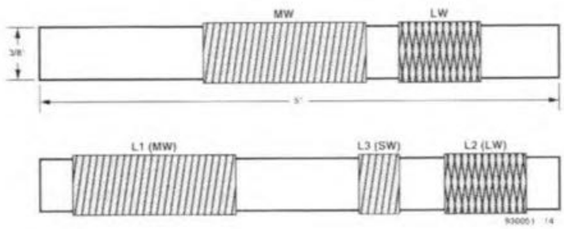


Figure 4.

sary, to make the unit include the 19 m band. All wiring should be in SWG16 (AWG14) wire for rigidity.

MW/LW unit. Again, the basic layout is as in Fig. 5. The final layout, with the coils and wiring added, is shown in Fig. 7.
The MW/LW aerial inductor is made from a standard 5 in (127 mm) long, 3/8 in (9.5 mm) dia. ferrite rod antenna coil salvaged from an old valve radio. Push the MW (L_1) and LW (L_2) windings outwards to the ends of the rod, leaving just enough room for the plastic cable clips used for mounting the asembly—see Fig. 7. Wind two turns of masking tape on the rod, close to L_1, and on to this close-wind L_3, which consists of 30 turns of SWG32 (AWG29) enalled copper wire—see Fig. 4. Note that a new commercial MW/LW ferrite rod aerial can, of course, be used; the rod length is not critical, but 5 in (127 mm) appears to be the standard length.

Mount the coil assembly into the box as shown in Fig. 7, securing it with stand-off plastic clips at both ends. The wiring is as shown in Fig. 2 and 7.

Testing and operation

Connect each unit in turn to a suitable receiver and resonate it to the receiver frequency by rotating C_1-C_2 for maximum signal. Turn the unit through 90° to check for directional properties.

A simple way to check the frequency range of each unit is the use of a calibrated receiver and an electronic calculator. The calculator produces 'noise' that is picked up by the aerial coil, which will peak when it is tuned to the receiver frequency.

In operation, the units should be kept well away from walls, large metal items and electrical wiring.

Conclusion

These units have been found particularly useful by the author; when not in use they can be stacked on a bookshelf. They can be taken to a receiver, rather than the more usual act of having to take the receiver to a fixed aerial. A preamplifier has not been found necessary with high RF gain receivers, but could be inserted between the unit and the receiver.

The performance of both the short wave

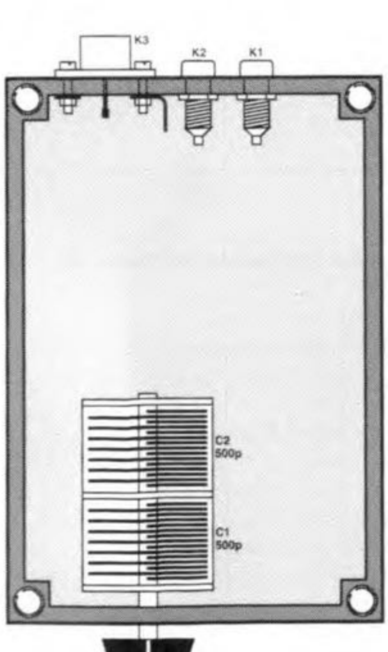


Figure 5.

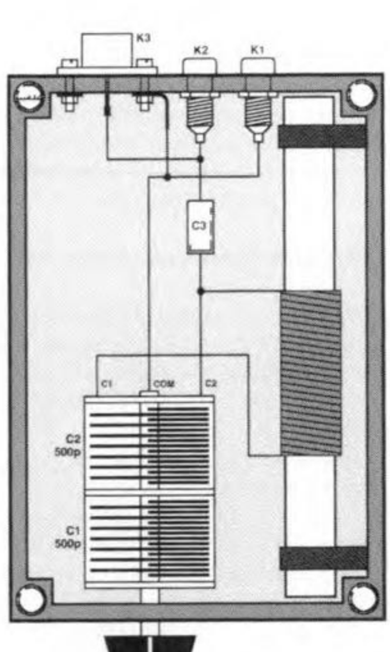


Figure 6.

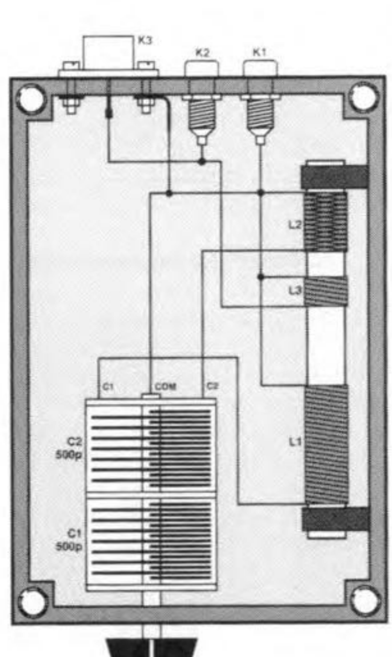


Figure 7.

and the MW/LW units has been found to be very satisfactory with a considerable reduction in general noise and interference.

At the time of writing, a version for use on Band 2 (VHF 87.5–108 MHz) is being worked on. Hopefully, details will be passed on in the future.

Parts list

1. For basic layout (Fig. 5)

Plastic box about 6½×4½×2 in (L×W×D) (165×150×50 mm).

C_1-C_2 = 2-gang, 500+500 pF variable capacitor with built-in slow motion drive (but see text for external drive) and knob.

K_1, K_2 = 4 mm banana sockets (black and red respectively): Tandy/Radio Shack Type 274-725 or similar. Also, suitable black/red cable end plugs to fit: Tandy/Radio Shack Type 274-721 or similar.

2. For short wave assembly (Fig. 6)

Basic layout as above.

Ferrite rod 6¼ in (158 mm) long by ½ in (12.5 mm) dia. cut from a Type R61-050-750 rod (7½ in [190 mm] long by ½ in [12.5 mm] dia. from Amidon Associates Inc. P.O. Box 956, Torrance, CA 90508, USA.

6¼ in (158 mm) long by ⅝ in (16 mm) outside dia. plastic pipe from hardware or DIY store.

2 clips, stand-off wall mounting type to fit the plastic pipe from hardware or DIY store.

SWG16 (AW14) tinned copper wire.

C_3 = 100 pF silver mica or ceramic disc capacitor.

3. For MW/LW assembly (Fig. 7).

Basic layout as in Fig. 5.

L_1 and L_2 = commercial LW/MW ferrite rod antenna assy (⅜ in or 9.5 mm dia. rod). May be salvaged from old receiver or purchased new from various suppliers; for instance, in UK, use Type LB12N from Maplin Electronics, P.O. Box 3, Rayleigh SS6 2BR.

2 plastic clips, stand-of wall mounting type, to fit the aerial rod. Available from most DIY stores.

SWG30 (AWG28) enamelled copper wire.

PHILIPS PREAMPLIFIER

By T. Giesberts

Computer-controlled hi-fi equipment is no longer a novelty. Such apparatus has given rise to special analogue ICs becoming available that are provided with a computer interface. Since hi-fi equipment is getting smaller, these ICs are normally made in surface mount technology—SMT.

This article is concerned primarily with two of such ICs from Philips: a stereo control amplifier and a five-band stereo equalizer. The computer interface is an I²C type. The performance of the ICs can be evaluated with the aid of a special program (for PCs only). This program enables the control of all functions with a mouse or the cursor keys. Even an I²C interface is not required, for that is simulated via the printer port. The program uses an 8-column text display.

Control amplifier Type TEA6330T

The block schematic of this IC is shown in Fig. 1. The audio input is taken direct to the volume/balance control. The volume for the left-hand and right-hand channel can be set independently between –66 dB and +20 dB. In this way, the volume control also acts as balance control.

Both the bass and the treble control adjust the two channels in tandem. The bass control has a range of –12 dB to +15 dB, while the treble control range is ±12 dB. The software can adjust the tone controls in accordance with the position of the volume control to give physiological volume control.

The tone control circuits may also be used as input for an equalizer, but the treble control is then out of action and the bass control is restricted to 0–15 dB.

The output circuit consists of two double stages. Particularly for application in car radios, the balance between the stages can be controlled independently of the volume with the aid of two faders. The stages may also be muted independently or jointly. Joint muting may also be effected via an additional input of the IC without using the I²C interface.

The design of the bass control is shown in Fig. 2, where sections *a*, *b*, and *c*, indicate bass amplification, neutral, and bass attenuation respectively. Figure 2*d* shows how the *RC* network in *a*, *b*, and *c* is transformed into a variable filter. The switches are not part of the IC: the resistors are switched by multi-input opamps. According to the manufacturer, this solution gives better results than other solutions as regards

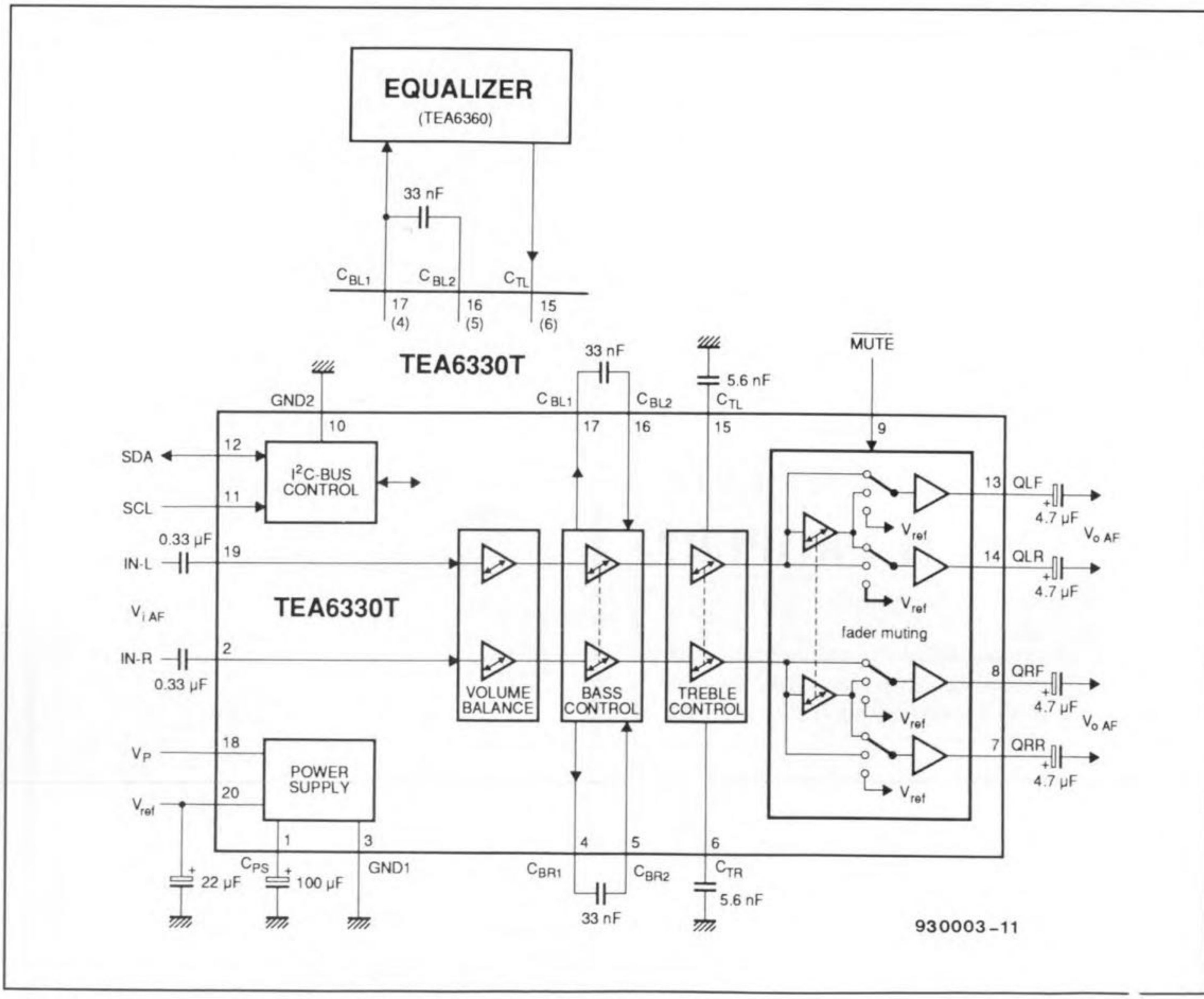


Fig. 1. Block schematic of the Philips preamplifier.

noise,, distortion and dynamic range.

The circuit diagram of the treble control is given in Fig. 3. The design is fundementally identical to that of the bass control circuit, but the position of C_1 is different.

Equalizer Type TEA6360

The equalizer (see Fig. 7) consists of not more than a couple of filter sections and an I²C interface. The frequency-determining components, in the form of a T-filter,

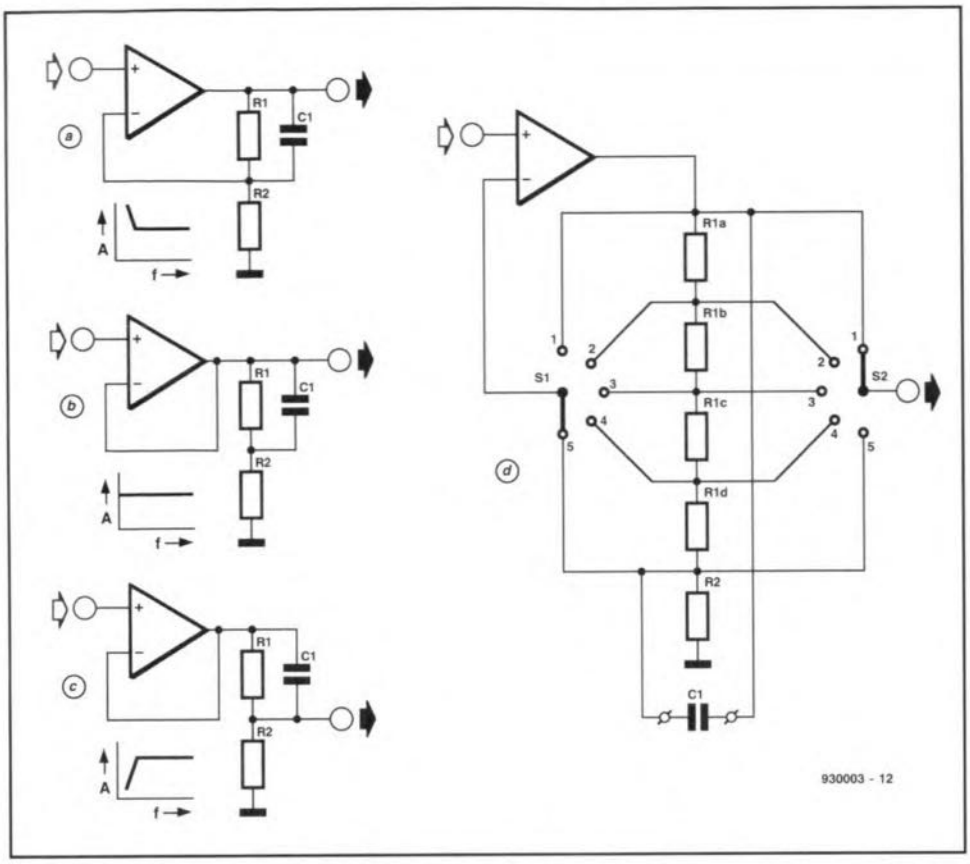


Fig. 2. Basic design of the bass control circuit.

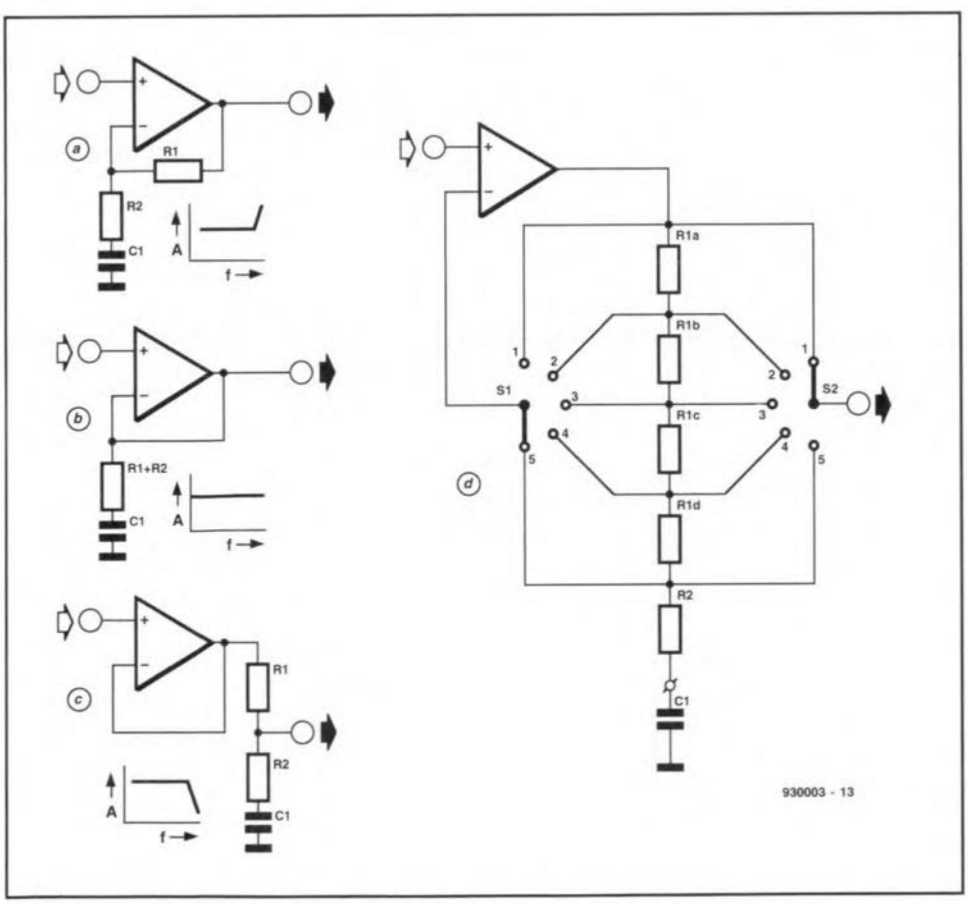


Fig. 3. The basic design of the treble control is identical to that of the bass control.

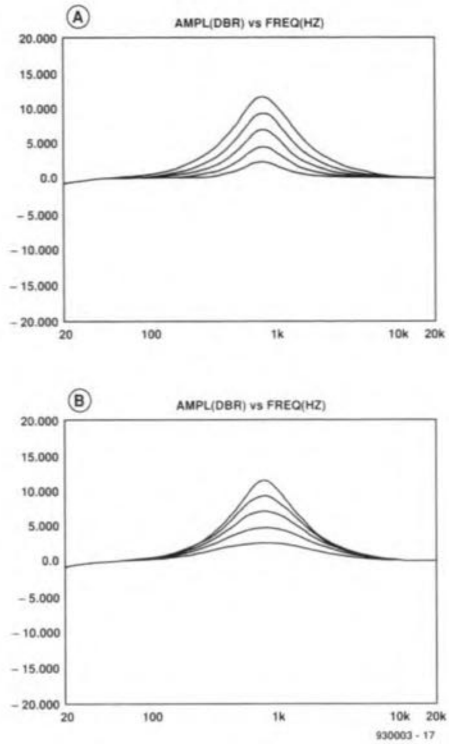


Fig. 4. The filter can operate with constant Q (a) or variable Q (b).

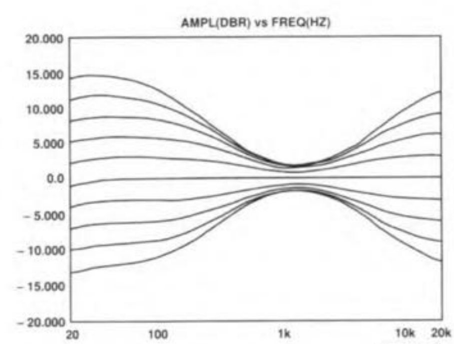


Fig. 5. If the equalizer is not used, the treble control circuit can amplify in asccorance with these characteristics.

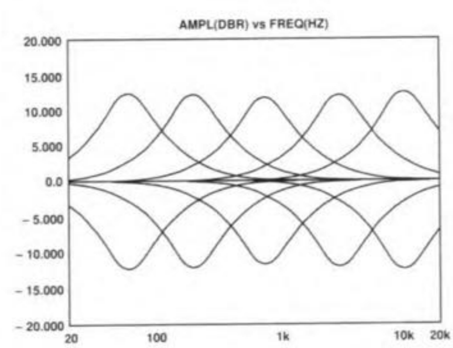


Fig. 6. The effect of each discrete filter section, i.e., one section at maximum (or minimum) and the others at zero.

are external to the IC. This arrangement makes it possible, at least in principle, to arrange the amplification and attenuation, the resonance frequency, and the *Q*-factor of each filter section, as required. In practice, this is not so straightforward, particularly when it is borne in mind that the IC is optimized internally for an amplification or attenuation of 12 dB. Moreover, in a five-band equalizer, the resonance frequencies and *Q*-factors asre never far away from those of the standard application. Although it is not shown, there is a facility for switching off the equalizer via the software. The inputs and outputs are then interconnected.

The design of each filter section—given in Fig. 8—is very similar to that of the tone control in the TEA6330. It is based on a T-section, R_2-C_1-C_2, which is shunted by resistor R_1 to enable the amplification or attenuation to be made variable with S_1 or S_2 respectively. With this design, the amplification or attenuation may be varied with constant *Q* or variable *Q*. Variable *Q* is the more usual in variable filters. The more the amplification or attenuation de-

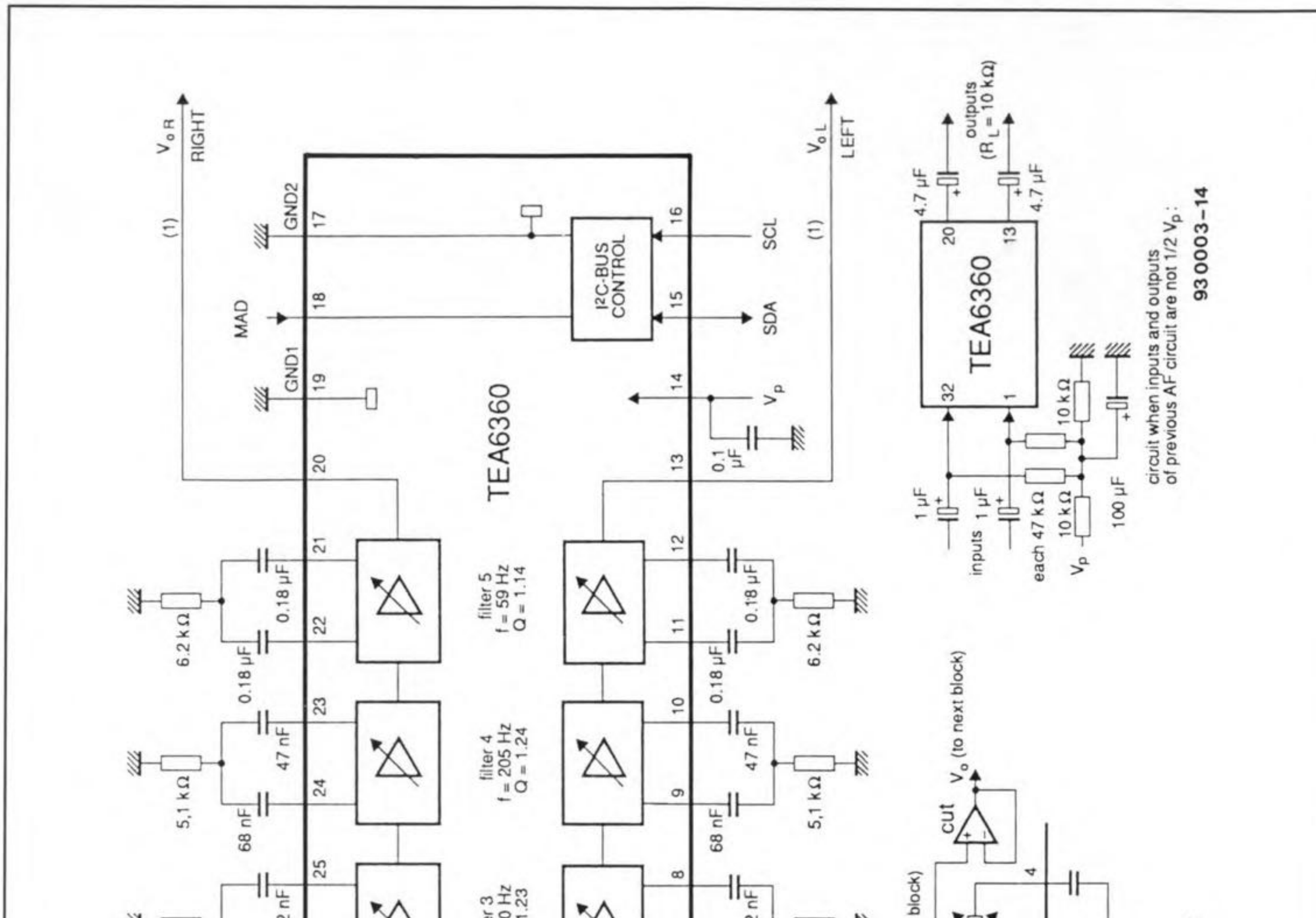


Fig. 7. Block schematic of equalizer Type TEA6360.

creases, the smaller Q gets and the flatter the filter response becomes. The difference between constant Q and variable Q operation can be seen in Fig. 4.

The two ways of filtering are obtained by operating switches S_1 and S_2 differently. In variable Q operation, the method of switching is the more logical. To obtain amplification, S_2 is set to position 1 and S_1 is adjusted to get the desired gain. To obtain attenuation, S_1 is set to position 1 and S_2 for the required attenuation.

In constant Q operation, the relevant switch is set for maximum attenuation or amplification, as the case may be, and the other for the required level of attenuation or amplification. This way of working has its drawback when a range of signals is being amplified. Since at the onset

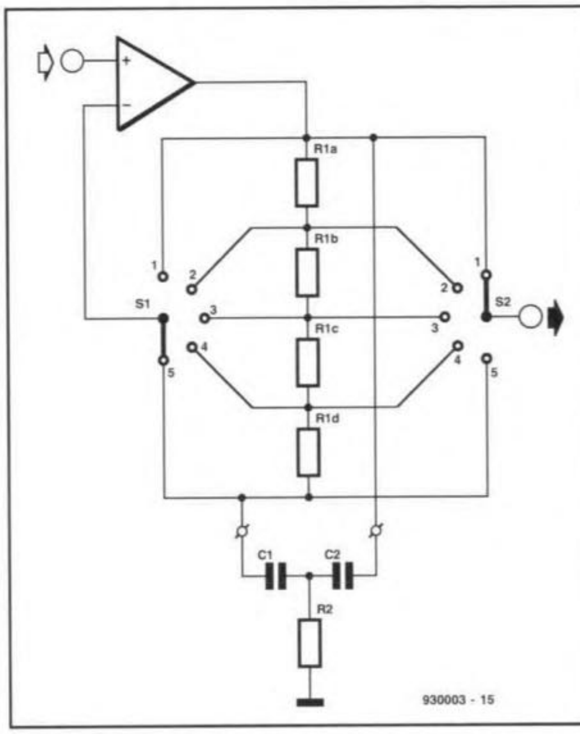


Fig. 8. The filter sections of the equalizer are based on a T-filter.

PARTS LIST

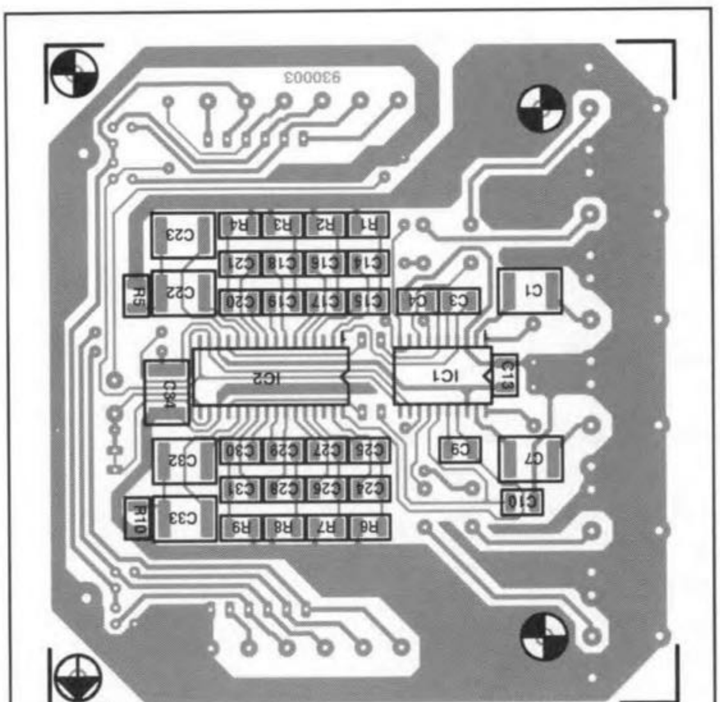
All resistors and capacitors, except electrolytic types, are surface-mount components.

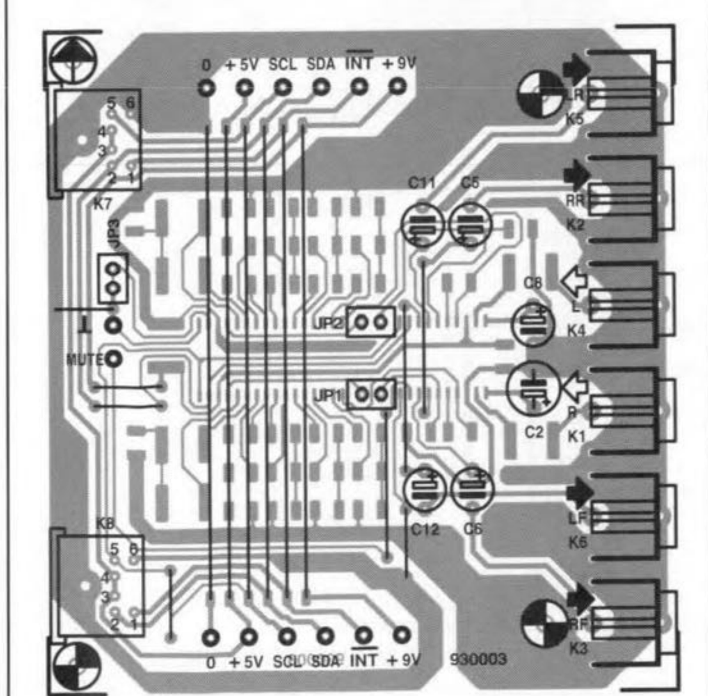
Resistors:
R1, R3, R4, R6, R8, R9 = 5.1 kΩ
R2, R5, R7, R10 = 6.2 kΩ

Capacitors:
C1, C7 = 330 nF
C2 = 100 μF, 10 V, radial
C3, C9 = 33 nF
C4, C10 = 5.6 nF (see text)C5, C6, C11, C12 = 4.7 μF, 63 V, radial
C8 = 22 μF, 25 V, radial
C13, C34 = 100 nF
C14, C24 = 4.7 nF
C15, C25 = 3.3 nF
C16, C17, C26, C27 = 1 nF
C18, C28 = 18 nF
C19, C29 = 12 nF
C20, C30 = 68 nF
C21, C31 = 47 nF
C22, C23, C32, C33 = 180 nF

Semiconductors:
IC1 = TEA6330T (Philips)
IC2 = TEA6360 (Philips)

Miscellaneous:
JP1, JP2, JP3 = 2-way header with jumper
K1–K6 = audio socket for PCB
K7, K8 = 6-way mini-DIN-socket for PCB
PCB Type 930003
Diskette Type 1861





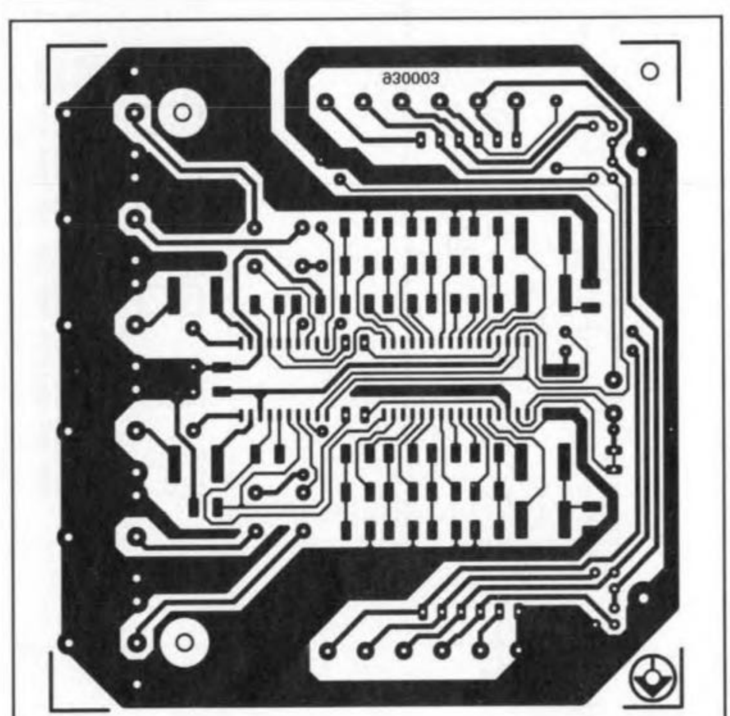


Fig. 9. The printed-circuit board for the amplifier has a double-sided component layout.

there is maximum amplification, strong signals may cause overdrive. To prevent that happening, a mix of the two methods may be used: the filter will then have a quasi-constant Q. In this way, maximum amplification is used only if it is absolutely necessary. This matter will be reverted to in the description of programming the IC.

Circuit description

Apart from the two ICs discussed earlier, the preamplifier circuit in Fig. 10 consists of the necessary input and output sockets and some peripheral capacitors and resistors. The mini-DIN-connectors are for use with the I²C interface. This interface is, as regards wiring, identical to the one published last year [1].

Which audio outputs (left or right) belong together is indicated by the second letter: F(ront) or R(ear).

If the mute input of IC1 is not used, it may be left open: the external mute is then off. The mute function is enabled by making the input low (≤1.5 V). The mute function is disabled not only when the input is left open, but also when the input signal is higher than 3 V (but ≤the supply

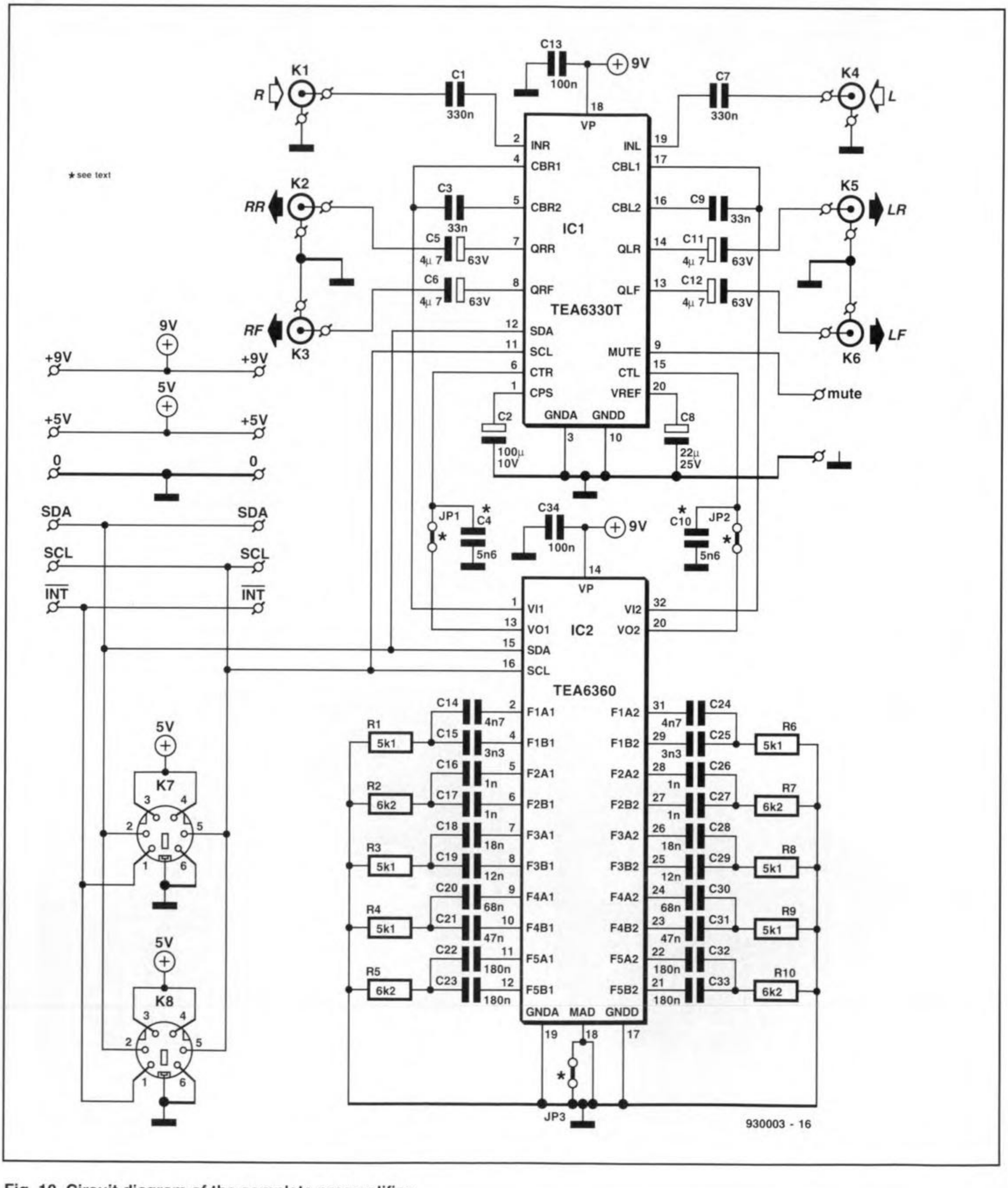


Fig. 10. Circuit diagram of the complete preamplifier.

voltage). In other words, the external mute input can be driven not only with open-collector outputs or discrete transistors, but also with standard TTL outputs.

If the equalizer is definitely not required, bridges JP_1 and JP_2 may be left open and capacitors C_4 and C_{10} used. The equalizer is then disabled and the treble control is enabled.

The sub-address for the equalizer on the I²C bus, 10000100_{BIN} or 10000110_{BIN}, may be set with JP_3 (see Table 1). On the PCB this jumper is already in place: break the relevant copper track for a high level; if JP_3 is then placed, the level is low again.

Construction

The doubled-sided board (Fig. 9) uses both sides for the components. The ICs, resistors and capacitors (except electrolytic types) must be fitted on the 'copper track' side, while the electrolytic capacitors, connectors and wire bridges (jumpers) are to be fitted on the component side.

Note that C_4 and C_{10} must be fitted only if the equalizer is not used (jumper bridges JP_1 and JP_2 open). The IC (TEA6360) and associated components may then, of course, also be omitted.

The ICs are best mounted by first soldering the two pins at diagonally opposing corners.

The control amplifier and equalizer cannot be used without a microprocessor. This controller converts the (analogue) signals from the operating controls into data for the amplifier and equalizer. It would be ideal, of course, to use a controller with an I²C interface (such as the 80C552 [2]), but this not really essential, because the I²C signals are easily generated with relevant software [3]. Philips uses the Centronics interface for experimental purposes, on the basis that this interface has both open-collector outputs and inputs with a pull-up resistor. Figure 11a shows how the Centronics connector of a PC must be wired for this purpose. Figire 11b gives a solution for PCs that (against all protocols) have no open-collector output.

The evaluation program is based on the assumption that the interface is connected as standard to printer port LPT1. This may be changed to, for instance, LPT2 by adding the instruction SET IICPORT=2 to the DOS prompt before starting the program. The speed of communication on the I²C interface must not exceed 100 kbit s^{-1}. To prevent this speed being exceeded in fast PCs, slowing down is achieved with a DOS environment-variable, for instance, SET IICSPEED=4. Values from 0 (no slowing down) to 9 (maximum slowing down) may be entered; 4 is the default value. Some retardations measured on an 8 MHz IBM PC/AT-03 are: IICSPEED=0: 0.43 ms $byte^{-1}$; =1: 0.68 ms $byte^{-1}$; =2: 0.98 ms $byte^{-1}$; =3: 1.28 ms $byte^{-1}$; =4: 1.58 ms $byte^{-1}$. Values larger than 4 will be required only with 386 machines

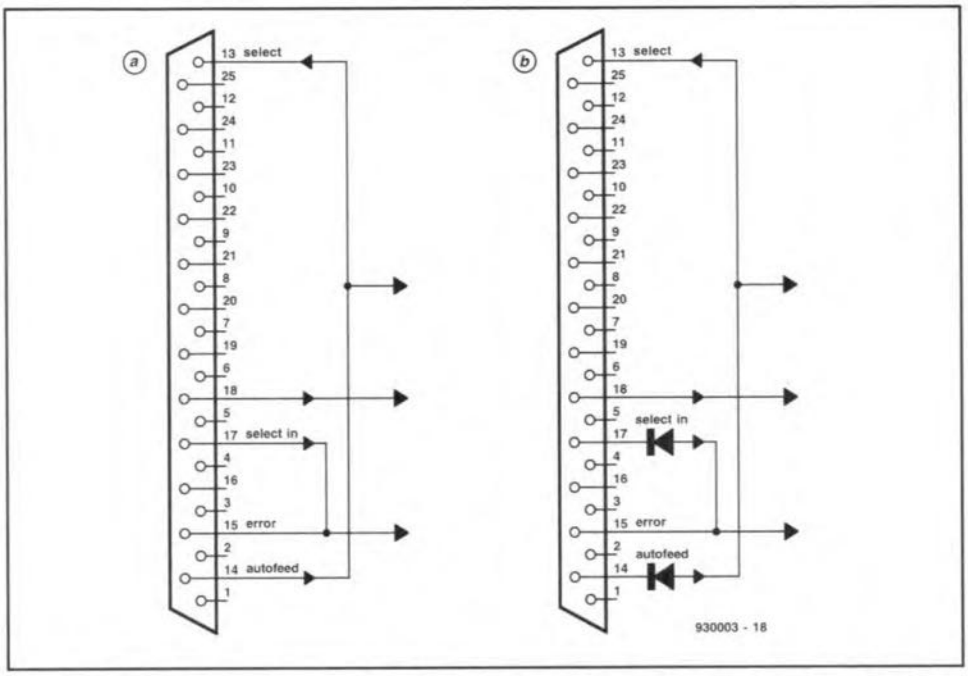


Fig. 11. The I²C interface may be controlled via the Centronics port. A standard Centronics interface with open collector outputs is shown in a, while a printer port without open collector outputs is shown in b.

S slave address A sub-address A data P

S = start bit
A = acknowledge bit
P = stop bit
data: see Tables 2–4
sub-address: see Table 2
slave address: TEA6330: 1000000x
where x = read/write> bit (1/0)
TEA6360: 10000100
when pin 18 is low
10000110
when pin 18 is high/open

When more than 1 data byte is sent, the sub-address is increased automatically by 1 after each byte.

Table 1. I²C bus format.

Gain (dB)	Data VL5 VR5	VL4 VR4	VL3 VR3	VL2 VR2	VL1 VR1	VL0 VR0
+20	1	1	1	1	1	1
+18	1	1	1	1	1	0
...						
+2	1	1	0	1	1	0
0	1	1	0	1	0	1
−2	1	1	0	1	0	0
...						
−64	0	1	0	1	0	1
−66	0	1	0	1	0	0
mute	0	1	0	0	1	1
...						
mute	0	0	0	0	0	0

Table 2. Volume setting left and right.

Function	Sub-address byte								Data byte D7	D6	D5	D4	D3	D2	D1	D0
volume left	0	0	0	0	0	0	0	0	0	0	VL5	VL4	VL3	VL2	VL1	VL0
volume right	0	0	0	0	0	0	0	1	0	0	VR5	VR4	VR3	VR2	VR1	VR0
bass	0	0	0	0	0	0	1	0	0	0	0	0	BA3	BA2	BA1	BA0
treble	0	0	0	0	0	0	1	1	0	0	0	0	TR3	TR2	TR1	TR0
fader	0	0	0	0	0	1	0	0	0	0	MFN	FCH	FA3	FA2	FA1	FA0
audio switch	0	0	0	0	0	1	0	1	GMU	EQN	0	0	0	0	0	0

VLx is volume/balance left-hand channel
VRx is volume/balance right-hand channel
BAx is bass control both channels
TRx is treble control both channels
FAx is fader setting front/rear

FCH is select fader channel, 1 = front; 0 = rear
MFN is mute-selected fader channel (mute front or rear)
GMU is general mute
EQN is external equalizer, 0 = connected; 1 = not available

Table 3. Control functions TEA6330.

Gain (dB)				Data			
Bass		Treble		BA3	BA2	BA	BA0
EQN=1	EQN=0	EQN=1	EQN=0	TR3	TR2	TR1	TR0
+15	+15	+12	0	1	1	1	1
+15	+15	+12	0	1	1	1	0
+15	+15	+12	0	1	1	0	1
+15	+15	+12	0	1	1	0	0
+12	+12	+12	0	1	0	1	1
+9	+9	+9	0	1	0	1	0
+6	+6	+6	0	1	0	0	1
+3	+3	+3	0	1	0	0	0
0	0	0	0	0	1	1	1
−3	0	−3	0	0	1	1	0
−6	0	−6	0	0	1	0	1
−12	0	−12	0	0	0	1	1
−12	0	−12	0	0	0	1	0
−12	0	−12	0	0	0	0	0

Table 4. Tone control settings.

	Gain (dB)				Data			
	MFN=1		MFN=0					
FCH=1	F	R	F	R				
FCH=0	R	F	R	F	FA3	FA2	FA1	FA0
	0	0	0	0	1	1	1	1
	−2	0	−84	0	1	1	1	0
	−4	0	−84	0	1	1	0	1
	−6	0	−84	0	1	1	0	0
	−8	0	−84	0	1	0	1	1
	−10	0	−84	0	1	0	1	0
	−12	0	−84	0	1	0	0	1
	−14	0	−84	0	1	0	0	0
	−16	0	−84	0	0	1	1	1
	−18	0	−84	0	0	1	1	0
	−20	0	−84	0	0	1	0	1
	−22	0	−84	0	0	1	0	0
	−24	0	−84	0	0	0	1	1
	−26	0	−84	0	0	0	1	0
	−28	0	−84	0	0	0	0	1
	−30	0	−84	0	0	0	0	0

Table 5. Fader settings.

Function	Sub-address byte								Data byte							
									D7	D6	D5	D4	D3	D2	D1	D0
filter 1/defeat	0	0	0	0	0	0	0	0	DEF	1B2	1B1	1B0	0	1C2	1C1	1C0
filter 2	0	0	0	0	0	0	0	1	0	2B2	2B1	2B0	0	2C2	2C1	2C0
filter 3	0	0	0	0	0	0	1	0	0	3B2	3B1	3B0	0	3C2	3C1	3C0
filter 4	0	0	0	0	0	0	1	1	0	4B2	4B1	4B0	0	4C2	4C1	4C0
filter 5	0	0	0	0	0	1	0	0	0	5B2	5B1	5B0	0	5C2	5C1	5C0

nB0–nB2 is boost control for filter *n*
nC0–nC2 is cut control for filter *n*
DEF is defeat bit; DEF=0, equalizer on; DEF=1, equalizer disabled with filter settings retained

Table 6. Control functions TEA6360

Gain	Data					
(dB)	nB2	nB1	nB0	nC2	nC1	nC0
+12	1	0	1	0	0	0
+9.6	1	0	0	0	0	0
+7.2	0	1	1	0	0	0
+4.8	0	1	0	0	0	0
+2.4	0	0	1	0	0	0
0	0	0	0	0	0	0
−2.4	0	0	0	0	0	1
−4.8	0	0	0	0	1	0
−7.2	0	0	0	0	1	1
−9.6	0	0	0	1	0	0
−12	0	0	0	1	0	1

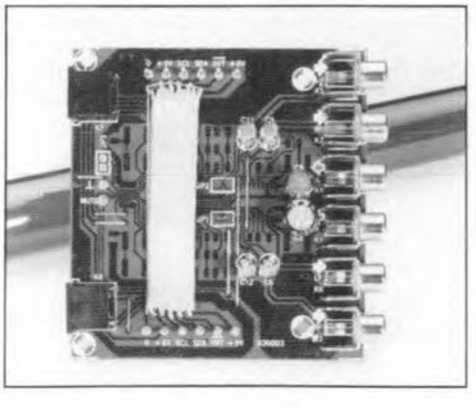
Table 7. Filter with variable *Q*.

Gain	Data					
(dB)	nB2	nB1	nB0	nC2	nC1	nC0
+12	1	0	1	0	0	0
+9.6	1	0	1	0	0	1
+7.2	1	1	1	0	1	0
+4.8	1	1	1	0	1	1
+2.4	1	0	1	1	0	0
0	0	0	0	0	0	0
−2.4	1	0	0	1	0	1
−4.8	0	1	1	1	0	1
−7.2	0	1	0	1	0	1
−9.6	0	0	1	1	0	0
−12	0	0	0	1	0	1

Table 8. Filter with constant *Q*.

Gain	Data					
(dB)	nB2	nB1	nB0	nC2	nC1	nC0
+12	1	0	1	0	0	0
+9.6	1	0	1	0	0	1
+7.2	1	0	1	0	1	0
+4.8	1	0	0	0	1	0
+2.4	0	1	1	0	1	0
0	0	0	0	0	0	0
−2.4	0	1	0	0	1	1
−4.8	0	1	0	1	0	0
−7.2	0	1	0	1	0	1
−9.6	0	0	1	1	0	1
−12	0	0	0	1	0	1

Table 9. Filter with quasi-constant *Q*.



with a 25 MHz clock.

The data that must be sent via the I²C bus have been summarized in Tables 1–9. Table 1 shows in what sequence the bytes must be sent. The slave address indicates for which IC the data is intended. If the IC is connected to the bus, it sends an acknowledgment to the microcontroller. The equalizer IC may be set to one of the two available addresses with jumper JP_3. The IC address is followed by the sub-address that indicates for which function of the IC the data is intended. These functions are given in Tables 3 and 6, which also show the composition of the data for the relevant functions.

Functions such as balance and physiological volume control must be set up with the aid of software. Bass control may be independent for the left-hand and right-hand channels (balance). Physiological volume control is obtained by increasing the bass tones when the volume is decreased.

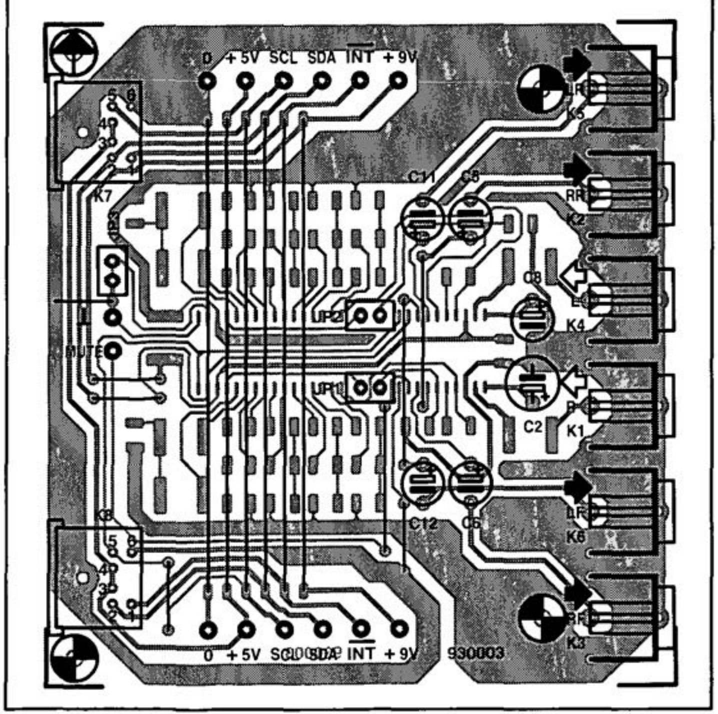
Tables 2, 4 and 5 show how the bits have to be completed for the control amplifier, while Tables 7–9 clarify how to control the equalizer with a different *Q* of the filter. Note that at the settings for quasi-constant *Q* the amplification is lower than at constant *Q*, so that overdriving of the equalizer happens fairly slowly. ■

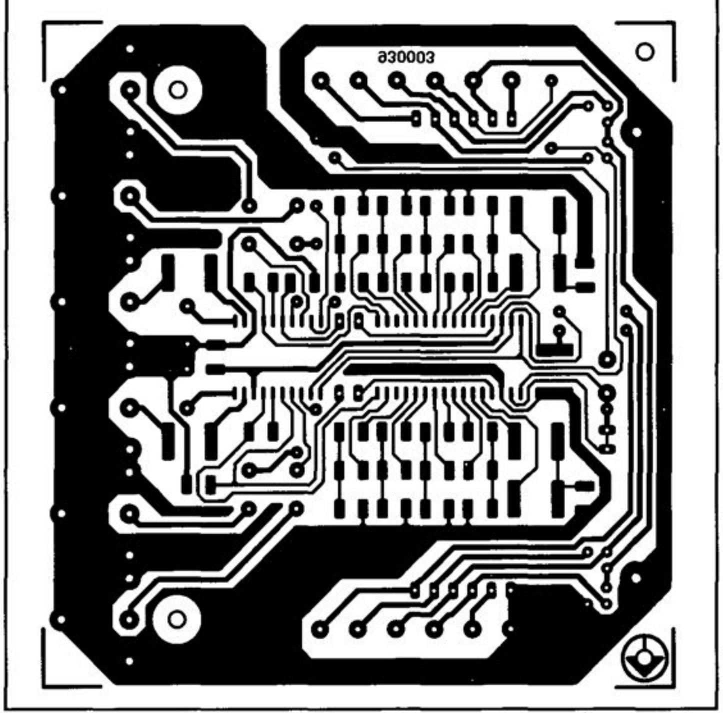
References:

1. 'I²C interface for PCs', *Elektor Electronics*, February 1992.

2. '80C552 Microprocessor System', *Elektor Electronics*, December 1992.

3. Various projects with an I²C interface published in *Elektor Electronics* during the past 12 months.


0
+5V
SCL
SDA
INT
+9V
K5
K7
C11
C5
RR
K2
C8
UP2
K4
MUTE
UP1
C2
K1
C12
C6
LR
K6
K8
RR
K3
930003


930003

VHF/UHF RECEIVER

The extremely wide tuning range and the AM/FM compatibility of the receiver described here allows you to listen to virtually any RF signal in the ether between 47 MHz and 890 MHz. Based on a ready made TV tuner unit, the receiver is simple to construct, and has a minimum of adjustments. Build it, and listen in on various types of communication: aircraft, ships, radio amateurs, mobile and portable radio networks, cellular radio, and carphones. In addition, the receiver covers TV sound channels in the VHF and UHF band, and in the S band and Hyperband used on cable networks.

Design by J. Barendrecht and B. Romijn

THE emphasis during the design phase of this project was on repeatability and ease of adjustment. To begin with, any problems that could possibly be expected to arise from the construction of the VHF/UHF front end of the receiver are eradicated by using a commercially available TV tuner module. Further-more, the selectivity of the IF (intermediate frequency) stage is ensured by a ready-made filter, which obviates inductor construction and adjustment. The result of these design goals is an exciting project aimed at the more advanced radio enthusiasts among you.

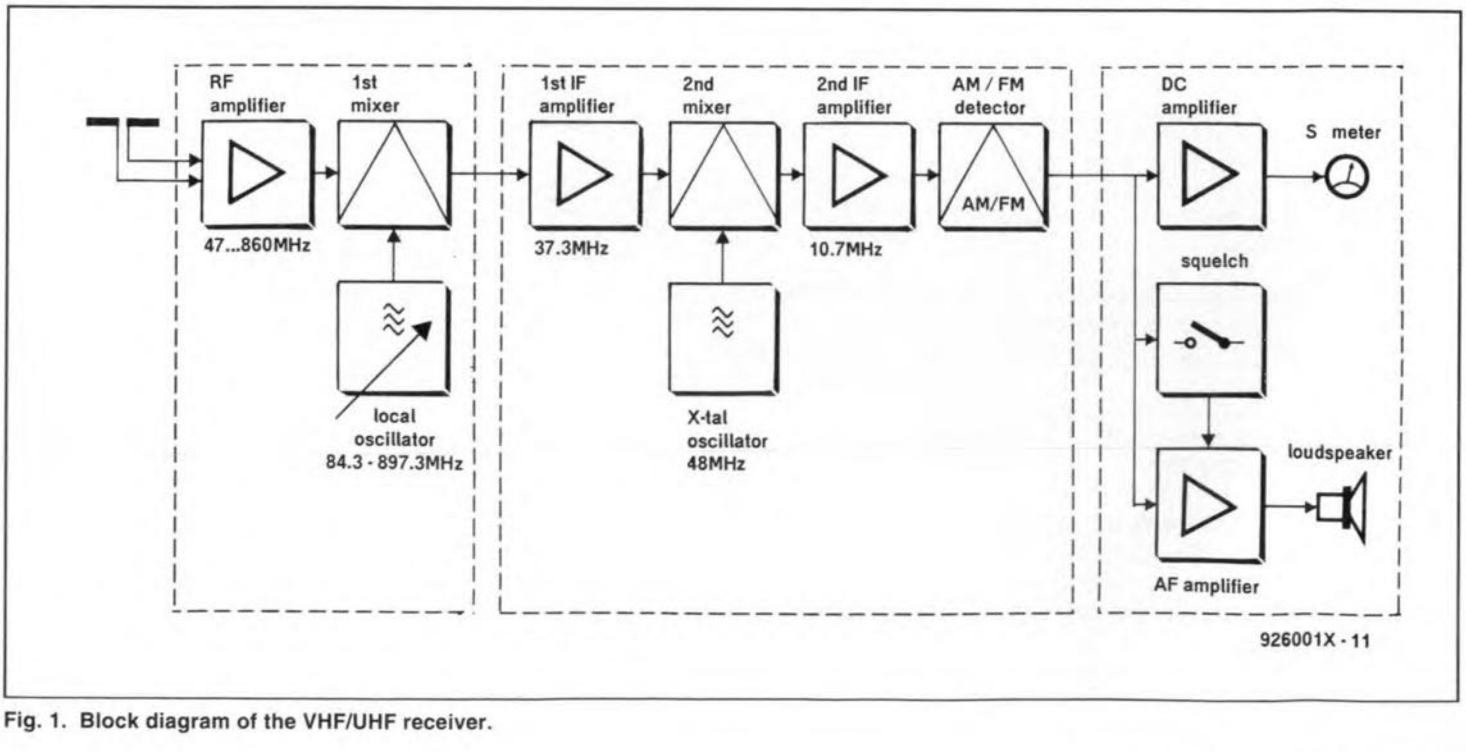


Fig. 1. Block diagram of the VHF/UHF receiver.

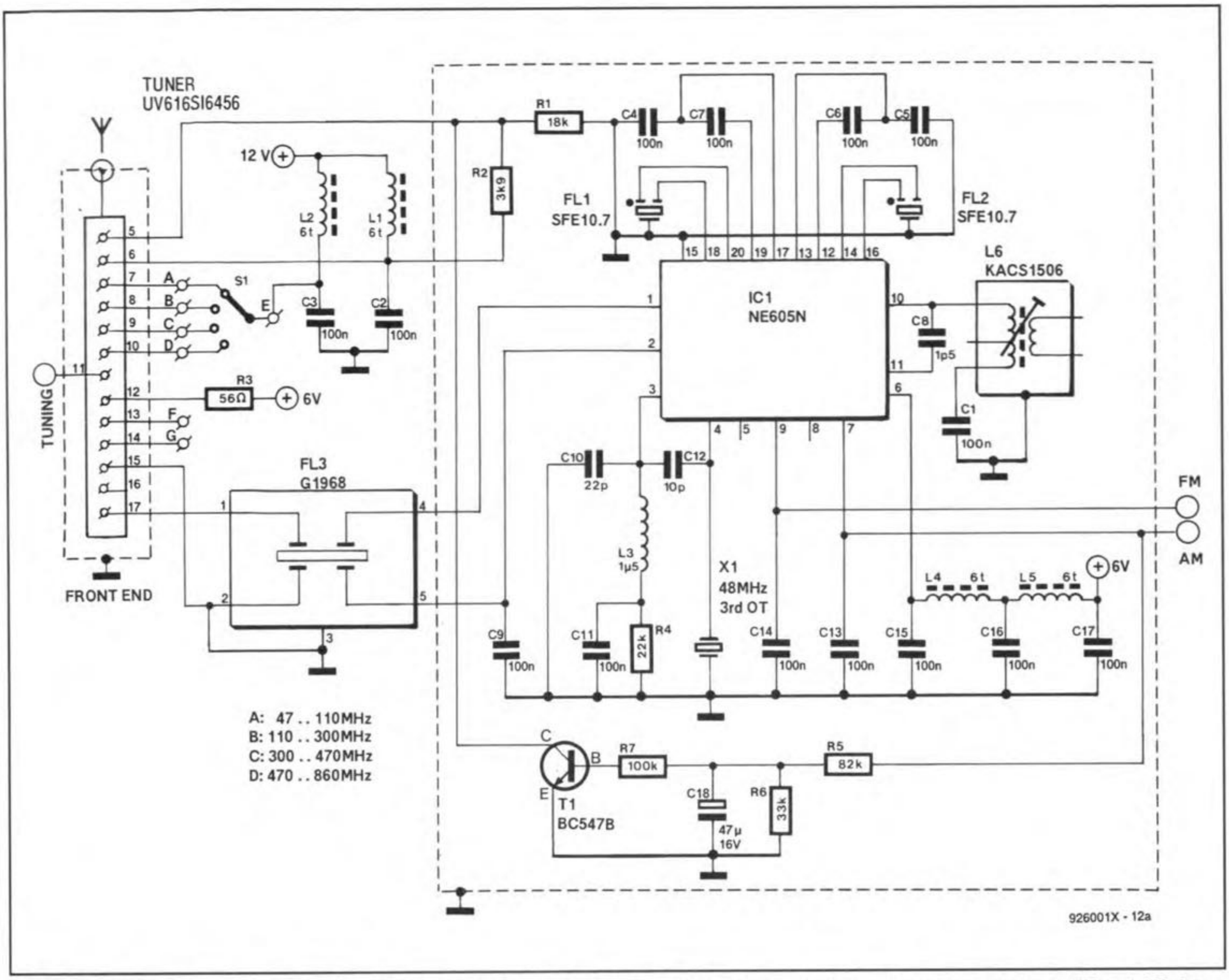


Fig. 2. Circuit diagram of the RF and IF sections of the receiver. Note that a single IC Type NE605 handles IF amplification, FM and AM detection, as well as AGC and S meter driving.

Those of you keen on knowing 'where they are' in the extremely wide frequency spectrum covered by the receiver will be pleased to learn that a 4-digit digital frequency read-out will be described in a future issue of *Elektor Electronics*.

Block diagram

The block diagram in Fig. 1 shows a superheterodyne (dual frequency conversion) receiver, whose most remarkable feature is its wide input frequency range covering more than 800 MHz.

Everything contained in the left-hand block is contained in a ready-made TV tune module from Philips Compo-nents. The home construction part actually starts with the IF amplifier and detector contained in the centre block. The selectivity of the first IF amplifier is ensured by a single ceramic filter with a bandwidth of about 6 MHz centred around 37.5 MHz. Such a filter is contained in most modern TV receivers, and has the same function as the three or five IF inductors fitted in yesteryears' models. Since the present receiver is essentially a narrow-band type, the 7-MHz bandwidth of the first IF stage has to be reduced to about 100 kHz. This achieved by mixing the 37.5-MHz signal down to 10.7 MHz, at which frequency the demodulation (AM or FM) takes place. The second mixer uses an inexpensive 48 MHz crystal for this purpose. The AM (amplitude demodulation) option is aimed at those of you interested in listening to aircraft communication (108-120 MHz).

Finally, the right-hand block contains the S-meter driver, the mute (squelch) circuit, and the AF output amplifier. The mute function may be used to 'silence' the receiver noise when there is no input signal. If a signal is picked up, the mute automatically 'opens', and the demodulated signal will become audible. The mute function is particularly useful when the receiver is on stand-by and tuned to a voice communication channel.

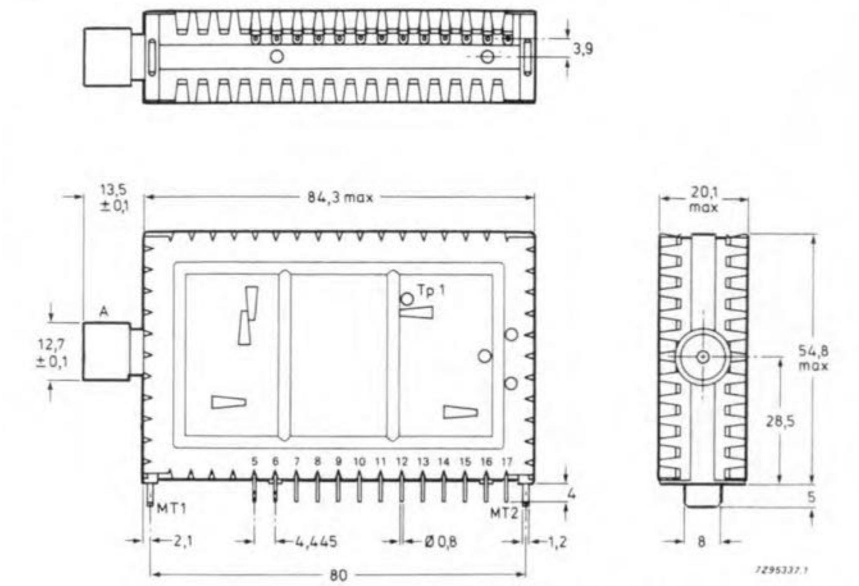
Circuit description

Front end

Figure 2 shows the RF and IF sections of the receiver. The antenna signal (47-860 MHz) is fed to the RF input of the TV tuner via a 75-Ω coax input socket. The down converted signal leaves the tuner via pin 17, and is fed to ceramic filter FL3. The pin connections of the Philips tuner module are given in Fig. 3. Filter L1-C2 prevents RF noise entering the tuner via the supply voltage. The tuning voltage (0.7-28 V) is connected to pin 11 of the tuner. Since it is technically almost impossible to have a single local oscillator cover the entire frequency range of 84-897 MHz, the tuning range is divided into four bands, which are selected by applying +12 V to the appropriate tuner pins (7-10). This is achieved with the aid of rotary switch S1. The AGC (automatic gain control) input pin of the receiver, pin 5, is driven with a control voltage supplied by the NE605 oscillator/mixer/demodulator, IC1.

MECHANICAL DATA

Dimensions in mm



Unless otherwise stated the tolerance is ± 0,05 mm.

Pin/connector identity

A	= aerial input (IEC female 75 Ω)
5	= AGC voltage, + 9,2 to + 0,85 V
6	= supply voltage, tuning part, + 12 V
7	= supply voltage, low VHF, + 12 V
8	= supply voltage, high VHF, + 12 V
9	= supply voltage, hyperband, + 12 V
10	= supply voltage, UHF, + 12 V
11	= tuning voltage, + 0,7 to + 28 V
12	= supply voltage, frequency divider, + 5 V (for UV616S versions only)
13, 14	= balanced output voltage of frequency divider (1 kΩ) (for UV616S versions only)
15	= prescaler switch mode (UV616S/6456) with pin 15 floating, division ratio = 1:64; with pin 15 grounded, division ratio = 1:256
16	= IF symmetrical output (approximately 75 Ω)
17	= IF symmetrical output (approximately 75 Ω)

926001X - 15

Fig. 3. Electrical connections of the UV616S/6456 four-band TV tuner from Philips Components.

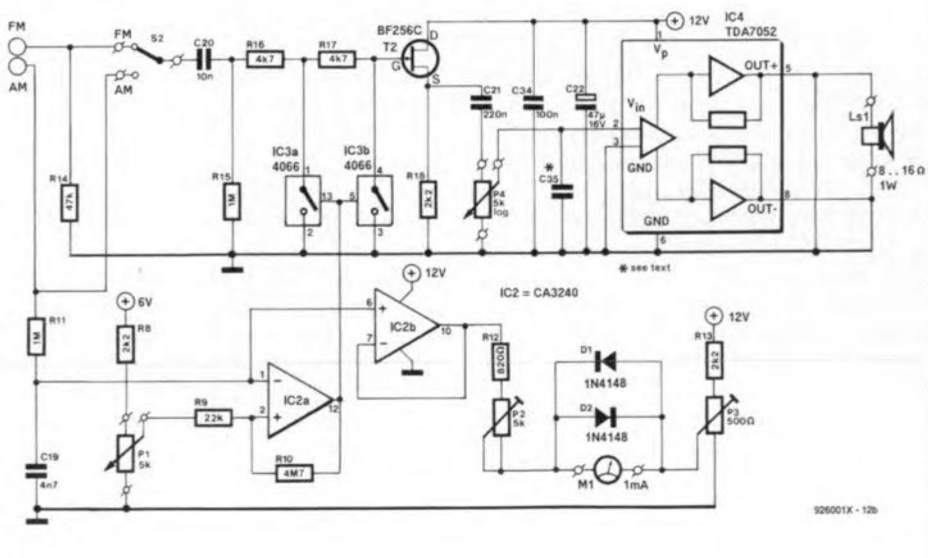


Fig. 4. Circuit diagram of the AF output amplifier, squelch and S meter circuits.

The tuner module used has an internal prescaler IC, which divides the local oscillator frequency by 256 (or 64). The prescaler outputs are available on tuner pins 13 and 14 (connections marked F and G in the circuit diagram), and may be used to drive a PLL (phase locked loop) or a counter. Since pin 15 of the tuner is tied to ground, the prescaler divides by 256, which results in an output frequency range of 0.33 MHz to 3.5 MHz.

IF amplifier/demodulator

The centre block in Fig. 1 is realized by IC1 and its surrounding components. The tuner output signal arrives at pin 1 of the NE605 via ceramic filter FL3. An amplifier contained in the NE605 is turned into a 48-MHz oscillator with the aid of a quartz crystal and a handful of passive components (L3, C10, C11 and C12, and R4). The quartz crystal used here is a third overtone type. Parallel tuned circuit L3-C10 short-circuits the fundamental frequency, 16 MHz, and so forces the crystal to resonate at its third overtone.

The 48-MHz signal and the first IF signal (37.3 MHz) are mixed in IC1, and the mixer output signal is applied to the input of the second IF amplifier. Like the first, this amplifier is contained in the NE605, and only its external components, capacitors C4-C7 and ceramic filters FL1 and FL2, are visible. The 10.7-MHz ceramic filters give the receiver an effective bandwidth of about 100 kHz.

The FM detector (demodulator) in the NE605 is driven by the output of the second IF amplifier, and has only three external components: C1, C8 and an IF transformer, L6, which is used as the quadrature inductor. By the way, L6 is the only inductor in the receiver that needs to be adjusted! The output signal produced by the FM demodulator is available on pin 9 of IC1. Capacitor C14 suppresses any residual 10.7-MHz components that may upset the AF output amplifier.

AM detection is achieved with a trick. Actually, the NE605 has no AM demodulator proper. However, it does have an output to connect an S-meter with a logarithmic scale. Here, the S-meter output is used as an AM take off. The fact that the demodulated signal is heavily compressed by the lin-to-log converter is useful rather than annoying: with AM, the compression produces a kind of fast automatic volume control, and also a quite effective limiter that keeps noise peaks away from the loudspeaker or headphones. Like the FM output, the AM output has a capacitor to suppress residual 10.7-MHz components.

Automatic gain control (AGC)

Although the S-meter output of the NE605 is 'misused' to provide AM demodulation, we have not forfeited its proper function: driving a moving-coil meter that indicates the relative signal strength of the received signal. The output voltage at pin 7 of IC1 is used for yet another purpose, namely driving the AGC circuit in the TV tuner. This is achieved with the aid of transistor T1, which functions as an inverting direct voltage amplifier — the higher the signal strength, the higher the voltage at pin 7 of IC1, and the lower the control voltage at the AGC input of the TV tuner. To prevent a too fast response of the AGC (for instance, as a result of AM signals), a low-pass filter, R5-R6-C18, removes all audio components, and feeds a 'slow' direct current into the base of T1, via base resistor R7.

Finally, the NE605 is operated from a well-filtered 6-V supply voltage applied to pin 6 of the device. Pin 15 is the negative supply (ground) connection.

AF amplifier and squelch

The audio amplifier in the receiver has few surprises, and few components, too — see Fig. 4! FET T2 serves to buffer the AM and FM output lines. This is necessary because load variation could upset the S-meter circuit and the squelch shown in the lower part of the circuit diagram. A single IC Type TDA7052 is used to provide the current gain required to make the demodulated AM and FM signals audible in a loudspeaker (16 Ω), or a pair of headphones. The supply voltage of the AF amplifier is decoupled with the aid of C22 and C34.

The squelch is formed by opamp IC2a and two electronic switches, IC3a and IC3b. Below a certain signal strength (adjustable with potentiometer P1), the switches close, and short-circuit the noise voltage to ground. Two switches are used to prevent any likelihood of signals 'leaking through' to the gate of the FET. The squelch opamp has some hysteresis and a low-pass filter, R11-C19, to prevent oscillation.

S meter

The control voltage at junction R11-C15 is not only sent to the squelch, but also to the S meter driver. Note that the S meter driver does not require the control voltage to be inverted as with the squelch. Hence, the control voltage is applied to the non-inverting input of IC2b. The function of this opamp is similar to that of FET T2: prevent the control voltage line from being unduly loaded (in this case, by the S meter driver). Without this buffering function of the voltage follower (amplification ×1), the demodulated AM signal would affect the AGC and the squelch. The output current supplied by IC2b causes the S meter, M1, to deflect. The amount of deflection is adjusted with the aid of preset P2. The other preset, P3, serves to null the meter when no signal is received. The meter coil is shunted by diodes D1 and D2 to prevent damage caused by noise peaks or voltages greater than about 0.6 V.

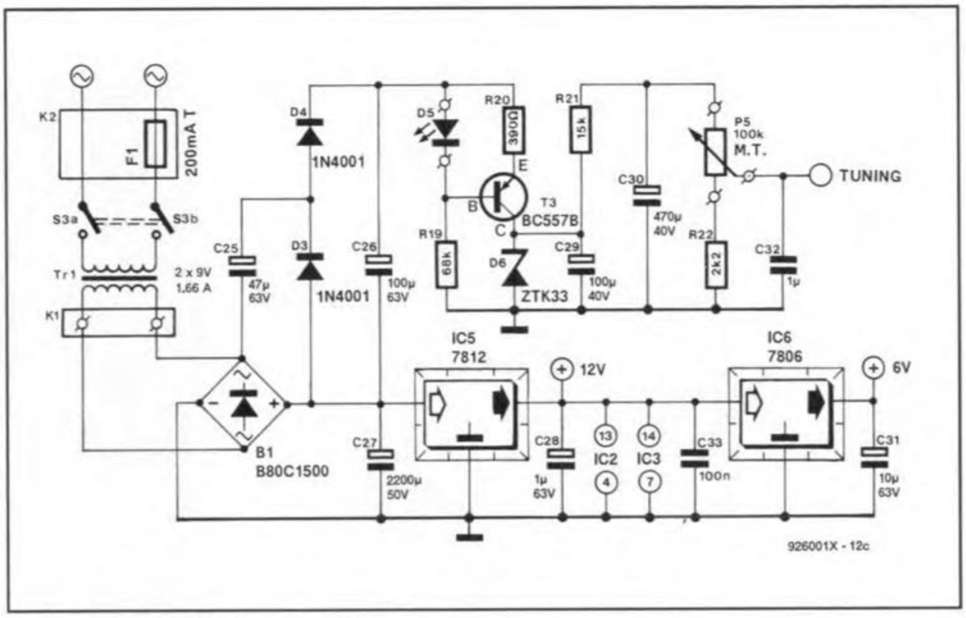


Fig. 5. The receiver's power supply.

Power supply

The receiver's power supply is a straightforward design — see Fig. 5. The circuit supplies regulated output voltages of 6 V, 12 V, and the tuning voltage, 0.7-28 V. The tuning voltage supply starts with a voltage doubler, C25-D3 and C26-D4, which produces an unregulated voltage of about 50 V across C26 and C27. This is converted into a stable 33-V reference voltage by the circuit around transistor T3. Actually, diode D6 does the job: this is a temperature-compensated zener diode specially designed for tuning voltage supplies. An equivalent of the indicated diode Type ZTK33 is the TAA550. The zener diode is fed with a constant current of about 2.5 mA, which is supplied by current source D5-R19-R20-T3. This is dimensioned such that a constant voltage of 1 V is maintained across resistor R20. The stable 33-V reference voltage is turned into a variable tuning voltage with the aid of potentiometer P5. The full tuning voltage range is limited somewhat by resistors R21 and R22 connected in series with the fixed terminals of the tuning potentiometer.

Construction

The receiver is built on a high-quality double-sided printed circuit board

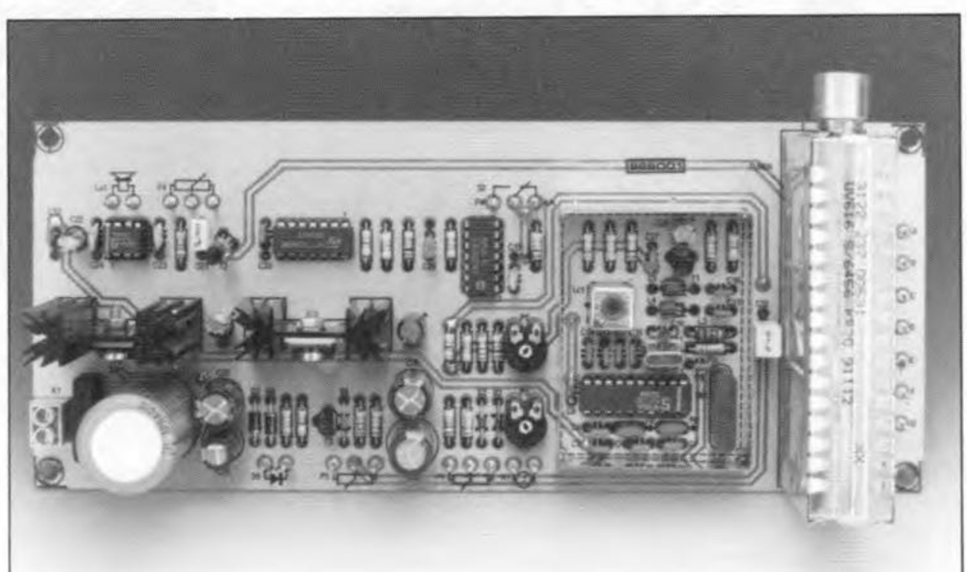


Fig. 6. Populated printed circuit board ready for testing.

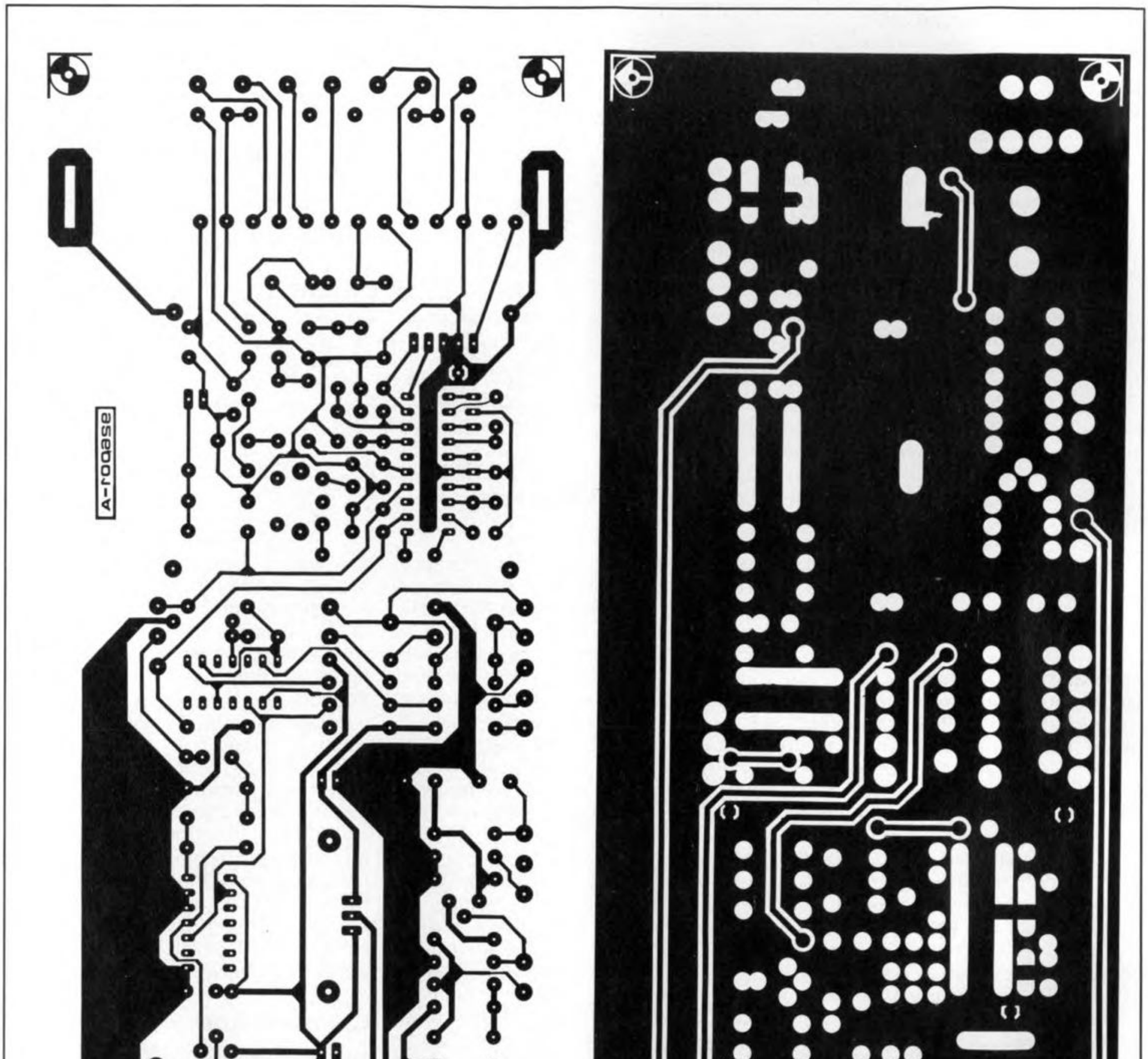


Fig. 7a. Track layouts (mirror images) of the double-sided printed circuit board designed for the receiver.

supplied through the Readers Services. If you want to make the PCB yourself, you may use the artwork given in Fig. 7a.

Construction is relatively easy, and there are few points to note in this regard. The most important of these is that the PCB is not through plated, which means that there are 18 locations where a component wire has to be soldered at both sides of the board (for example, one wire of C34). IC sockets are best not used in view of the high frequencies on the board. Do not yet mount IC3 (4066) and the TV tuner.

Chokes L1, L2, L4 and L5 are home made. They consist of 6 turns of 0.2-mm dia. enamelled copper wire through a small (3-mm long) ferrite bead. Pay attention to the orientation of the ceramic filters: pin 1 is marked with a dot (FL1 and FL2) or a '1' (FL3) on the filter body (Fig. 8). Regulators

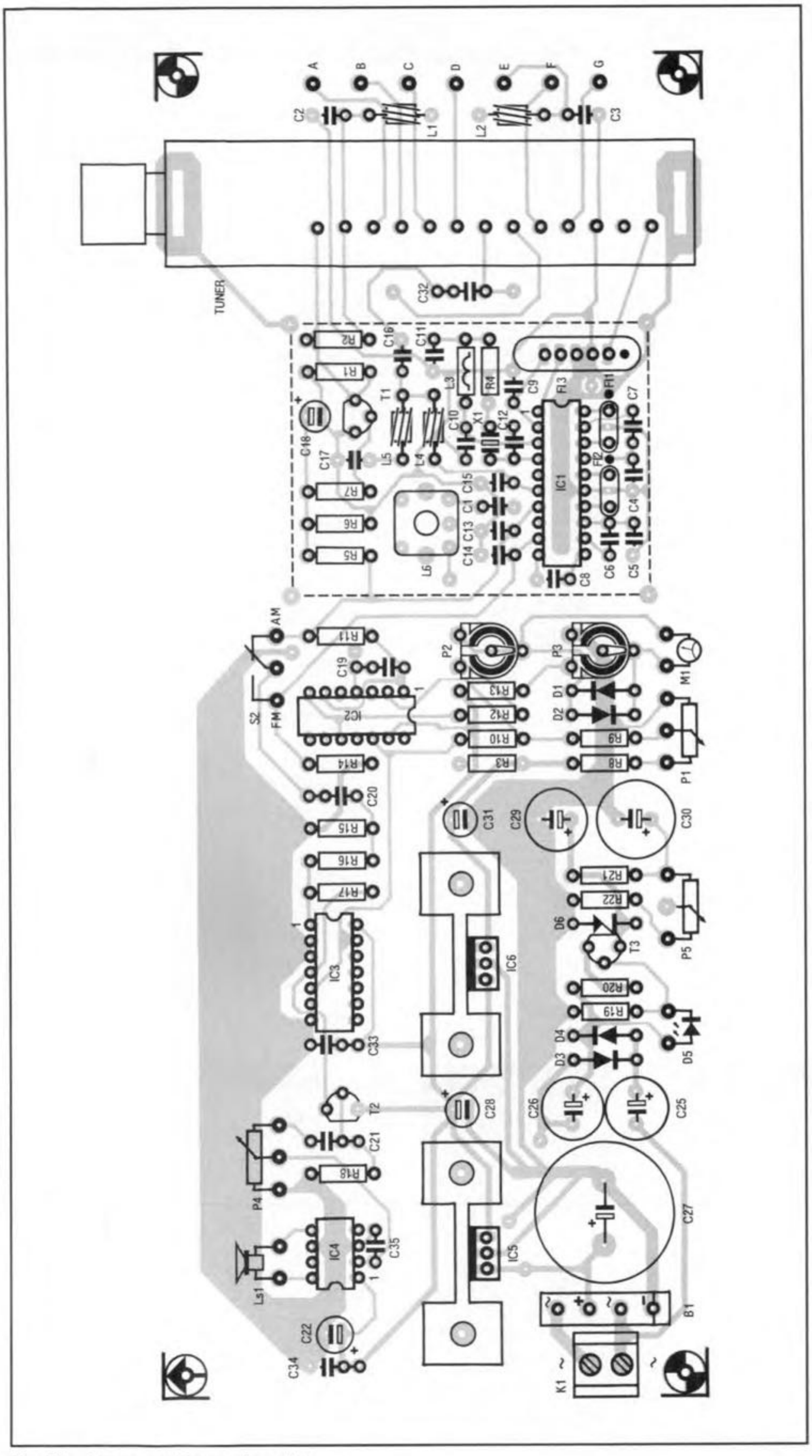


Fig. 7b. Component mounting plan.

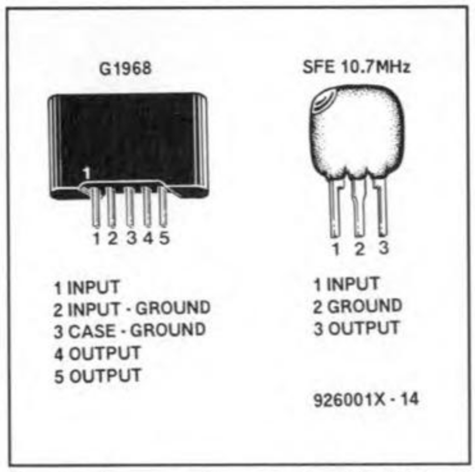


Fig. 8. Pin connections of the 10.7-MHz filters used.

IC5 and IC6 are mounted on heatsinks using ceramic or mica isolation elements. The heat-sink for IC6 need not be as large as that for IC5.

Before proceeding with the initial tests and adjustment of the receiver, inspect your solder work carefully, and check all component positions and values against the PCB overlay and the parts list.

Initial tests and adjustment

First, check that the power supply output voltages are present. The tuning voltage supply takes about 1 minute to stabilize. Check that the tuning voltage can be adjusted between about 0.7 V and 28 V. Connect the loudspeaker, the S meter and the two potentiometers, P1 and P4, to the relevant PCB terminals. Turn P4 to maximum volume and touch the wiper. The loudspeaker should produce hum. The same hum should be audible when you touch the centre PCB terminal of the AM/FM switch, S2.

Proceed with the S meter and the squelch circuits. First, switch off the receiver, and fit IC3 (4066). Set P2 to minimum and P3 about midway. The next test requires a signal to be simulated on the 'AM' line. This can be achieved with the aid of a small direct voltage taken from the top side of P1. Connect a test lead with croc clips to the PCB terminal of P1 closest to C30, and the 'AM' PCB terminal of S2. The S meter should deflect, and the amount of deflection should be adjustable with P2, and, to a much smaller extent, with P3. The hum that is audible when the centre pin for S2 is touched should disappear when P1 is turned to its top position. If this just about fails, it should be possible in any case to achieve the same effect by removing the test lead between the AM line and the top side of P1.

If all of this checks out, you are ready to proceed with the HF/IF section of the circuit. The best point to start there is the crystal. Lacking an oscilloscope, you may use a portable FM radio to check that the 48 MHz oscillator works. Simply tune the radio to 96 MHz, hold its antenna close to the crystal, and check for the first harmonic.

Next, connect the band selection switch, S1, to the PCB, and check that the +12-V band selection voltage appears at the right solder spots for the TV tuner. Also check the presence of the tuning voltage and the tuner supply voltage.

COMPONENTS LIST

Resistors:

1	18kΩ	R1
1	3kΩ9	R2
1	56Ω	R3
2	22kΩ	R4;R9
1	82kΩ	R5
1	33kΩ	R6
1	100kΩ	R7
4	2kΩ2	R8;R13;R18; R22
1	4MΩ7	R10
2	1MΩ	R11;R15
1	820Ω	R12
1	47kΩ	R14
2	4kΩ7	R16;R17
1	68kΩ	R19
1	390Ω	R20
1	15kΩ	R21
1	4kΩ7 lin. potentiometer	P1
1	5kΩ preset H	P2
1	500Ω preset H	P3
1	4kΩ7 log. potentiometer	P4
1	100kΩ multiturn preset	P5

Capacitors:

Ceramic types unless otherwise noted

16	100nF	C1-C7;C9; C11;C13-C17; C33;C34
1	1pF5	C8
1	22pF	C10
1	10pF	C12
1	4nF7	C19
1	10nF	C20
1	220nF MKT	C21
1	1µF MKT	C32

Electrolytic capacitors:

2	47µF 16V radial	C18;C22
1	47µF 63V radial	C25
1	100µF 63V radial	C26
1	2200µF 50V radial (pitch 10mm)	C27
1	1µF 63V radial	C28
1	100µF 40V radial	C29
1	470µF 40V radial	C30
1	10µF 63Vaxial	C31
	not fitted	C23;C24

Inductors:

4	3mm ferrite bead with 6 turns 0.2mm CuL	L1;L2;L4;L5
1	1µH5 choke	L3
1	KACS1506 (Toko)	L6

Semiconductors:

1	B80C1500	B1
2	1N4148	D1;D2
2	1N4001	D3;D4
1	LED, red, 3mm	D5
1	ZTK33 or TAA550	D6
1	BC547B	T1
1	BF256C	T2
1	BC557B	T3
1	NE605N	IC1
1	CA3240	IC2
1	4066	IC3
1	TDA7052	IC4
1	7812	IC5
1	7806	IC6

Miscellaneous:

1	UV616S/6456 TV tuner module (Philips Components)	
2	SFE10.7 ceramic filter (Murata)	FL1;FL2
1	G1968 ceramic filter (Siemens)	FL3
1	Quartz crystal 48MHz	X1
2	Heatsink isolation set (IC5;IC6)	
1	Loudspeaker 8-16Ω/1W	Ls1
1	1mA moving coil meter	M1
1	2×9V @1.66A toroid mains transformer	Tr1
1	200mAT fuse	F1
1	PCB terminal block; pitch 5mm	K1
1	Mains appliance socket with integral fuseholder	K2
1	1-pole 4-way rotary switch	S1
1	SPDT switch	S2
1	DPDT mains switch	S3
2	Heatsink (IC5;IC6)	
1	Metal enclosure, e.g., Telet LC860	
1	Printed circuit board 926001 (see page 70)	

Suggested component suppliers for this project are:

Bonex Ltd. (0753) 549502;
Cirkit Distribution Ltd. (0992) 441306;
Cricklewood Electronics Ltd. (081) 452 0161;
C-I Electronics, fax (+31) 45 241877;
ElectroValue (0784) 433603.

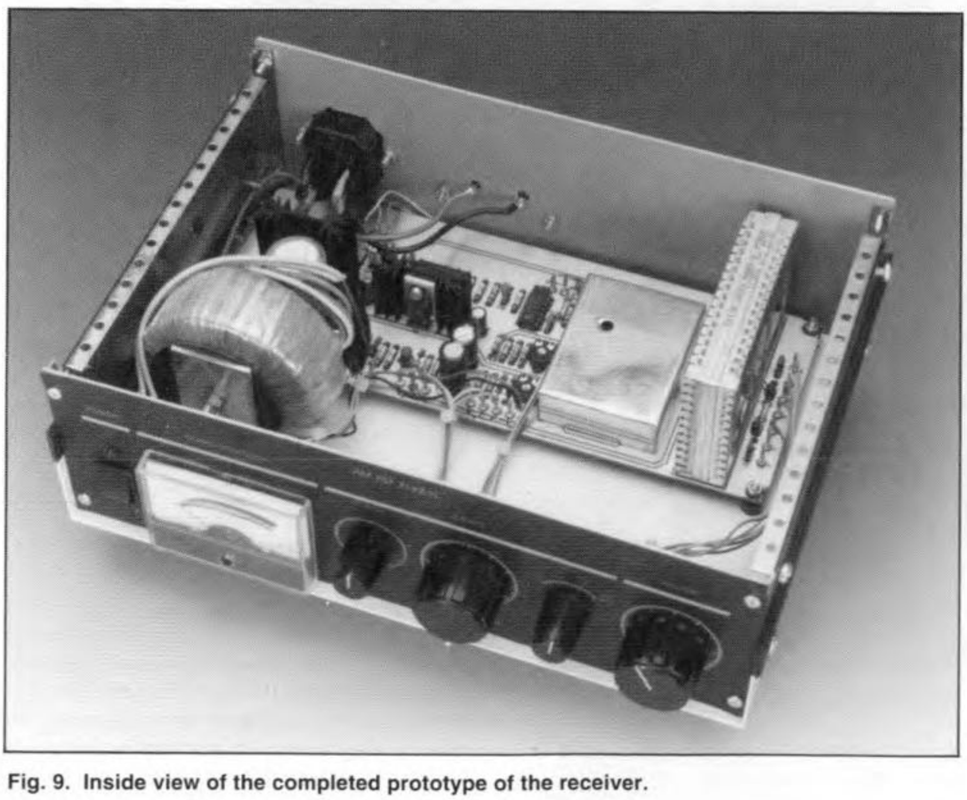


Fig. 9. Inside view of the completed prototype of the receiver.

Finally, switch off and mount the TV tuner module on to the board.

Connect a wire with a length of about 75 cm to the tuner input, and apply power. Wait about one minute to allow the tuning voltage to stabilize. Switch the band selection to 'A' (47-110 MHz), and tune the receiver with P5. It should be possible hear some VHF FM broadcast stations. Disconnect the antenna for a moment, and carefully adjust L6 for highest noise output. Next, adjust P3 until the S meter reads zero. Connect the antenna again, and tune to a strong signal. Adjust P2 for maximum meter deflection (the needle should reach a point just before the mechanical stop). Disconnect the antenna, and if necessary redo the adjustment of P3. Connect the antenna, and disconnect it again. If the AGC works properly, it should respond by regulating the tuner gain from minimum to maximum. This should result in a slowly increasing noise level, and an even slower reduction of the S meter deflection.

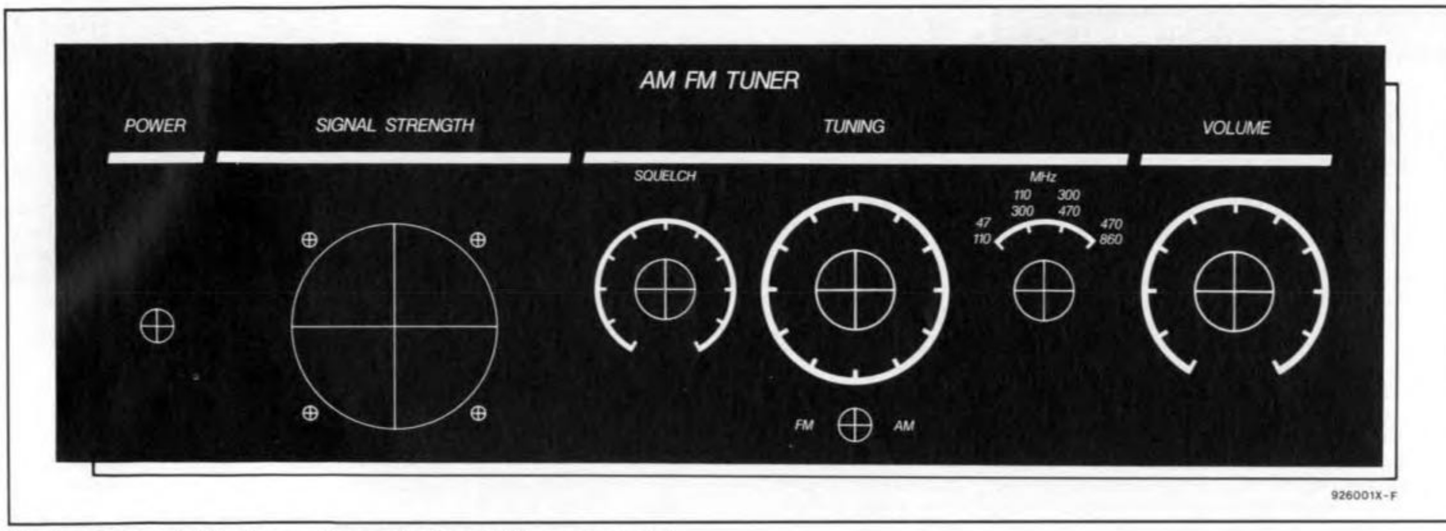


Fig. 10. Suggested front panel layout (not available ready-made).

Finally, switch over to the other bands, in which, with some luck, you should be able to detect signs of life (TV signals, aircraft beacons, car radio repeaters, etc.) too.

Finishing touch

The photographs in this article should give you a number of useful suggestions regarding the way the receiver is built into a metal enclosure. Before you start filing and drilling, however, fit a 20-mm high screening box around the IF/demodulator section on the board. The position of the box is indicated by the dashed lines on the component overlay printed on the board. The box is fitted with a cover to ensure complete shielding from the rest of the circuit. To reduce the radiation produced by the crystal, you may want to connect its case to the PCB ground via a short wire (note that this requires scratching away some of the protective lacquer on the board). In some cases, the sensitivity of the receiver may be improved a little by connecting the ground terminal of a number of components to the PCB ground plane at the component side. The relevant points are (grounded terminal only): C4, C5, C9, C14 and the tabs of L6. At the underside of the PCB, a drop of solder tin may be landed between pin 15 of IC1 and the finger-shaped ground area underneath the IC. After improving the grounding of L6, this inductor may have to be re-adjusted for highest noise output in the absence of an RF input signal.

Front panel control elements P5 (tuning), P4 (volume) and AM/FM mode selection (S2) should be connected to the PCB using shielded audio wire, of which the braid is connected at the PCB side only.

The loudspeaker may be built into the cabinet, or connected as an external unit via a socket on the rear panel. Alternatively, you may want to fit a headphones socket on the front panel. The mains transformer, preferably a toroidal type, should be fitted as securely as possible. Your transformer supplier should be able to advise you on this.

Finally, the choice of the antenna to be connected to the receiver will depend on the frequency range you wish to receive. In many cases, a normal VHF FM dipole and a TV band yagi antenna will already bring in a host of signals. For VHF mobile radio reception we recommend a ground plane like the one shown in Fig.12. The clever thing about this antenna is that it can be 'tuned' from VHF-FM to the communication sections in the UHF band (460 MHz) by pulling out or pushing in the elements of the four telescope antennas it consists of. The base impedance of this ground plane is about 70 Ω, which gives a fair enough match to the receiver input. ■

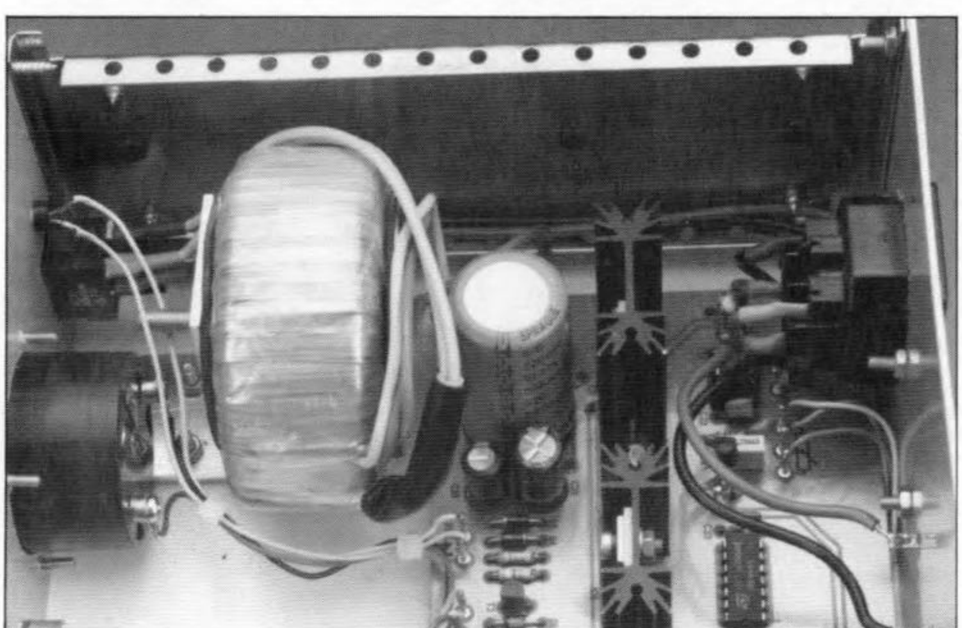


Fig. 11. Close up of the power supply section in the receiver case.

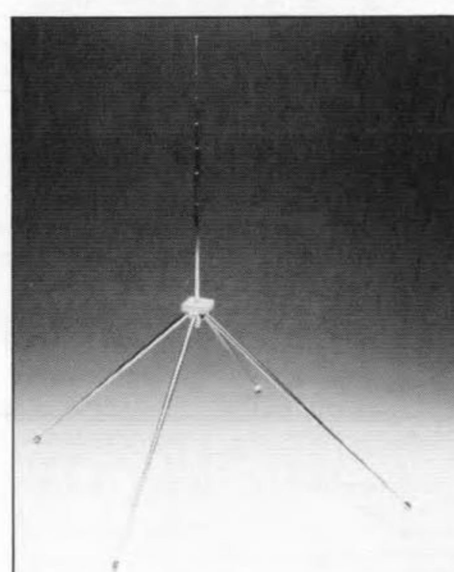
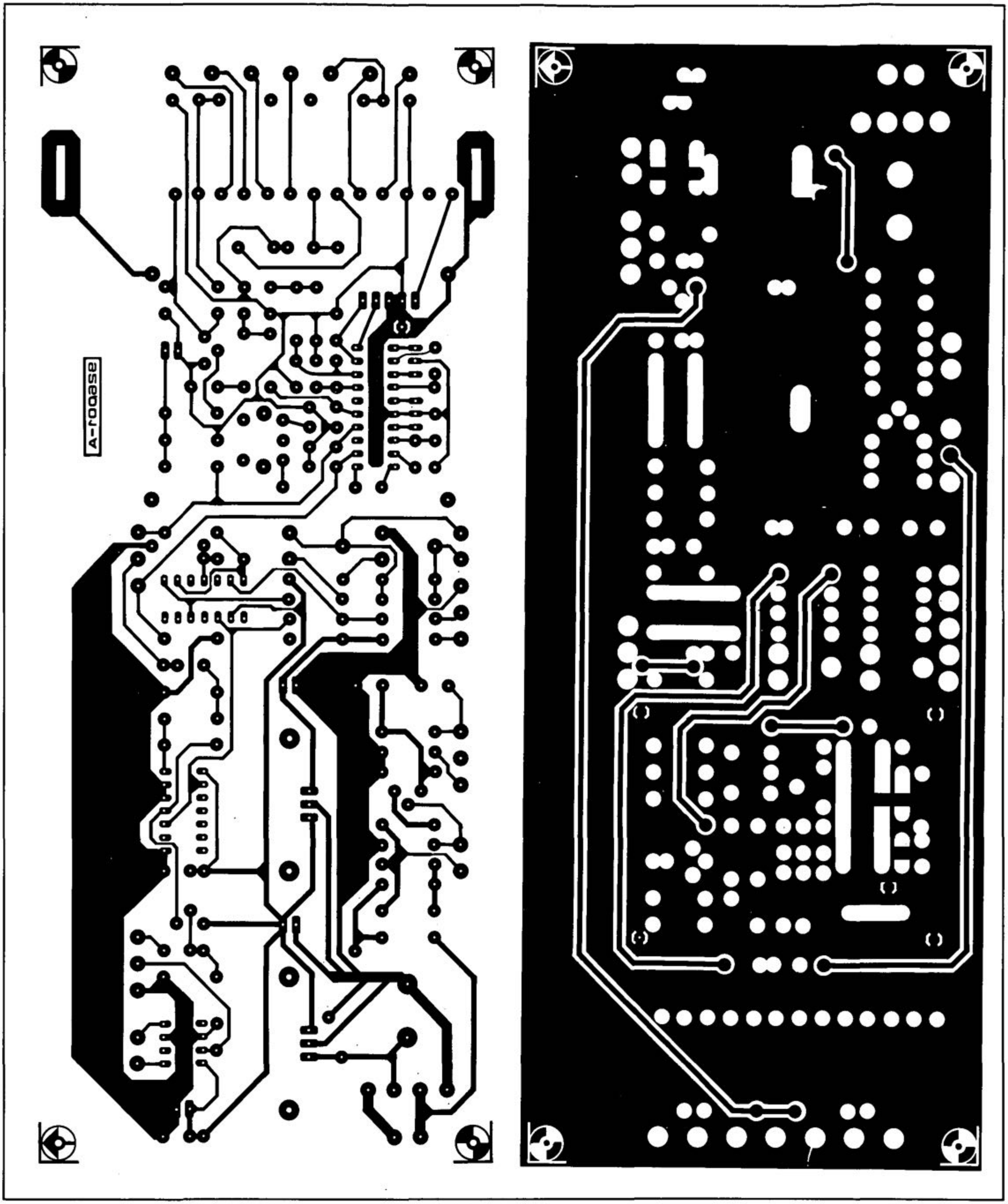


Fig. 12. An adjustable ground plane antenna made from four FM radio telescope antennas and a BNC socket.



CORRECTIONS AND UPDATES

Digital Audio/visual system (Multi-purpose Z80 card)

May and June 1992
An extensive description of a modification to the memory backup circuit on the Multi-purpose Z80 card is available free of charge through our Technical Queries service.

FM stereo signal generator

May 1993
Capacitors C17 and C19 should have a value of 33nF, not 3nF3 as indicated in the circuit diagram and the parts list of the multiplex generator.

Workbench PSU

May 1993
The polarity of capacitor C15 is incorrectly indicated on the PCB component overlay (Fig. 5a), and should be reversed. The circuit diagram (Fig. 2) is correct.

Transformer TR2 is incorrectly specified in the circuit diagram (Fig. 2) and in the parts list. The correct rating of the secondary is 2 × 12V/5A. Also note that the secondary windings are connected in series to give 24 V.

Audio DAC

September 1992
The polarity of capacitors C25 and C58 is incorrectly indicated on the component overlay of the D-A board (order code 920062-2), and should be reversed.

U2400B NiCd battery charger

February 1993
The value of resistors R17 through R27 should be 2.7kΩ, not 12.7kΩ as stated in the parts list.

VHF/UHF receiver

May 1993
In Fig. 4, the connections to ground of the AF amplifier outputs, pins 5 and 8, should be removed. The amplifier outputs are connected to the loudspeaker only. The relevant printed circuit board is all right.