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Model Control

PPP Valve Power Amplifier

KT88's in parallel push-pull

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PPP HiFi Valve Power Amplifier

using KT88 valves

Design by G. Haas (Experience Electronics)

Thanks to their pleasant sound, valve-type power amplifiers continue to enjoy uninterrupted popularity. With such an amplifier, you can eliminate the impression of coldness, sterility and artificiality that many people experience with CDs.

The design for a power amplifier that is presented in this article is based on the PPP principle. PPP stands for 'Parallel Push-Pull'. 'Push-pull' means that the output stage is composed of two active elements acting in phase opposition. One of the valves handles the positive half-cycles, while the other one handles the negative half-cycles. In a Parallel Push-Pull configuration, the valves in the output stage are connected in parallel with respect to the ac signal. The disadvantage is that the power efficiency per valve pair is less than with classical Class AB push-pull operation. Otherwise the PPP principle has only advantages. This output stage configuration was invented in the early 1950s, and it was the configuration of choice in studios. Reduced distortion, good sound and a wide frequency

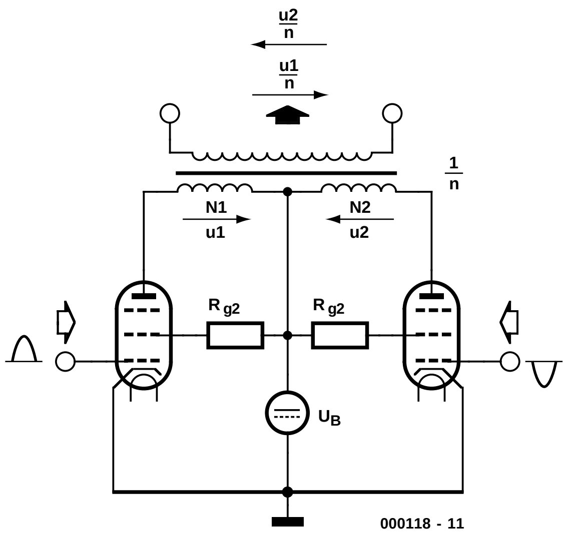
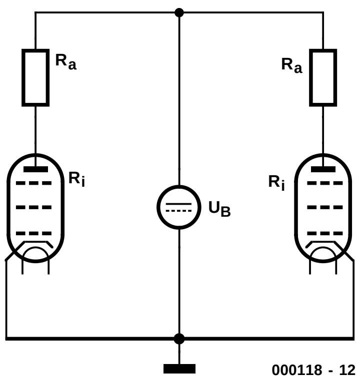
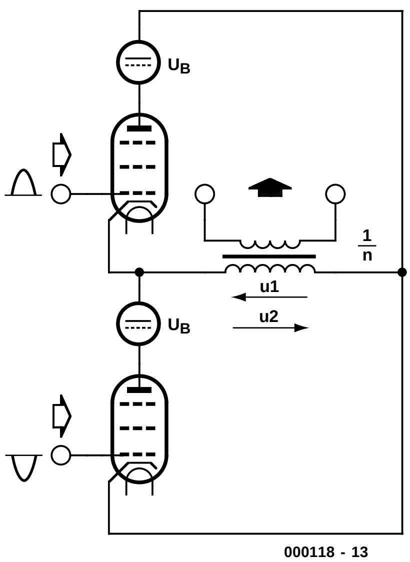


Figure 1. Basic circuit of a conventional Class AB push-pull output stage.



$$R_{l1} = 2 \cdot R_i + 2 \cdot R_a = 2 \cdot (R_i + R_a)$$

Figure 2. The ac equivalent circuit of Figure 1.



$$R_{l2} = R_i/2 + R_a/2 = 1/2 \cdot (R_i + R_a)$$
$$R_{l2} = 1/4 \cdot R_{l1}(!)$$

Figure 3. In a PPP output stage, the signal current flows through the entire primary winding of the transformer.

response were more important in this application area than high power efficiency. With the triumphal march of semiconductor technology, PPP was unable to retain any significant territory, due to its relatively low power efficiency and relatively high price. However, with the aid of modern resources it is possible to construct excellent HiFi PPP power amplifiers at an acceptable price.

AB versus PPP

In order to enable you to better understand the PPP technique, we first have to delve into a bit of theory. The schematic diagram of **Figure 1** shows the basic circuit of a classical Class AB push-pull circuit. Each of the control grids of the power valves is driven by a half-wave signal from a paraphase circuit. This is shown symbolically in the figure; in practice the two valves would naturally be driven by positive half-wave signals, since otherwise the circuit would not work. Here the left-hand valve is responsible for the positive half-cycle and the right-hand valve for the negative half-cycle. The output transformer combines the two half-cycles to form the complete sine wave. The figure shows the polarities of the voltages that appear in opposite phases across two halves of the primary transformer winding (N1 and N2). They generate a voltage on the secondary side of the transformer that is reduced by the transformer ratio *n*. From the figure, you can also see the response of the power stage to hum components in the supply voltage. If a hum voltage is superimposed on the supply voltage U_B, it will be fed into the two windings N1 and N2 in equal measure. If the circuit on the primary side of the transformer is completely symmetric, which means that the windings, output valves and quiescent currents are identical, the hum voltages cancel each other out due to the antiphase feed into the transformer. Unfortunately, perfect symmetry cannot be achieved in practice, so U_B must be adequately filtered.

Figure 2 shows the ac equivalent circuit of the basic circuit of Figure 1. The power supply U_B represents a short circuit for ac signals. The ac internal resistance is given by the series connection of the internal resistances of the two valves and the two windings N1 and N2 with their equivalent resistances Ra. Both output valves are operated with a non-zero quiescent current, in order to avoid cut-off distortion when the sine wave signal passes through zero.

The situation with a PPP output stage is different, as can be seen in **Figure 3**. You will notice that there are two supply voltage sources and that the valves are connected to

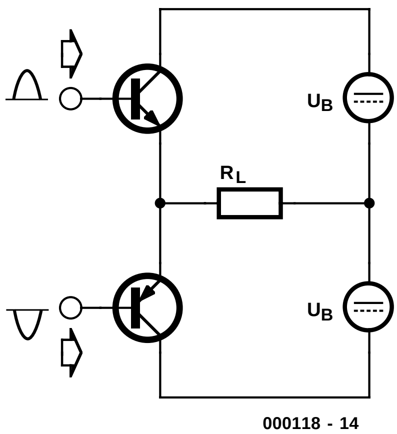


Figure 4. The ac equivalent circuit of Figure 3.

the transformer in a completely different manner. The ac signal current flows through the complete primary winding of the transformer, which is similar to the situation with a mains transformer. Here the valves are connected in parallel for ac signals, which reduces the internal resistance of the PPP circuit by a factor of four. This has various benefits. First of all, the transformer may have a smaller transformer ratio. This reduces the effects of stray inductance and winding capacitance, which yields an improved frequency response. Besides this, a genuine ac current flows through the complete primary winding, instead of one half-cycle per half winding as in the classic Class AB push-pull circuit. This avoids the much-feared 'flyback' effect that occurs with Class AB push-pull output stages, in which a half-wave voltage across one half of the primary winding generates a voltage that is twice as large on the plate of the opposite valve, due to the 2:1 ratio of the autotransformer formed by the primary winding. If the transformer core is magnetised and the stored energy is not fully consumed on the secondary side, the voltage on the primary side rises sharply, and to make matters worse it is again multiplied by the same 2:1 factor. The end result is a voltage breakdown at the weakest point — valve, socket or transformer. This may be a desirable effect in an automobile ignition coil, but in an amplifier it is devastating. In the event of a short circuit, a Class AB push-pull stage will become hot, but it will not be destroyed. Such a valve output stage is thus by nature short-circuit proof, but not open-circuit proof.

A PPP output stage, by contrast, is safe under both extreme operating conditions. Since the transformer is connected as previously described, the output stage does suffer from 'flyback'. It can be compared to a mains transformer, which also suffers no ill effects if it is connected to the mains with no secondary load. The PPP power stage can be seen as a predecessor of the standard transistor push-pull output stage. As we have seen, a whole series of indisputable benefits are associated with this design, to wit low internal resistance and tolerance of short-circuit and open-circuit loads, as well as an inherently good frequency response.

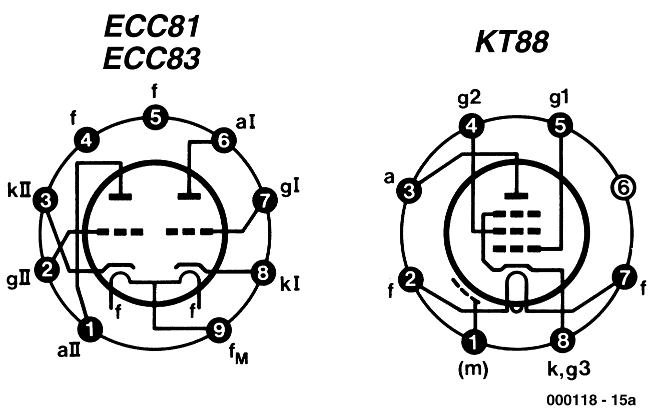
A power amplifier in three building blocks

Figure 5 shows the complete schematic diagram of a monophonic power amplifier. In consists of three blocks, and is consequently divided into three circuit boards. This allows the amplifier to be built such that the valves show to best advantage, as can be seen from the photograph. The transformers and the large electrolytic capacitors, in contrast to many other amplifier designs, are 'embedded' in the equipment.

Block 1 is the amplifier circuit, block 2 is the power supply and block 3 is the switch-on current limiter. At the input to the amplifier, provision is made for an E-1220 audio transformer. If you want to have a floating symmetric input, you can use the transformer together with an XLR socket; otherwise, a Cinch socket is adequate. Thanks to the transformer, it is possible to use floating symmetric cabling, as is common practice in studios. This avoids earth loops and quality losses due to the cabling. Furthermore, the transformer can be connected either 1:1 or 1:2. In the 1:1 configuration, the input resistance is around 34 kΩ, with an input sensitivity of 1.5 V for full output power. With the 1:2 connection, the input resistance drops to 8.5 kΩ, but the output stage can be fully driven with only 0.75 V. Suitable wire bridges are provided on the circuit board. If the amplifier is wired with a Cinch socket, the input sensitivity remains at 1.5 V.

At the input to the amplifier, the 2.2-µF bipolar electrolytic capacitor C1 ensures low-impedance coupling and at the same time blocks any dc component of the signal. Resistors R3 and R4 block coupled-in RF interference. Valve V1a provides the main amplification, with RF oscillations being suppressed by C2. Valve V1b is wired as an impedance converter, and it serves as a low-impedance source for the signal applied to paraphase circuit valve V2. The grid of valve V2a is driven directly without a coupling capacitor, since the dc potential has been brought to the proper level by suitable selection of component values. The control grid of V2b is connected to the same dc potential as the grid of V2a via R14. Capacitor C10 provides a short-circuit path to ground for any ac components that may be present at the grid of valve V2b, which is driven by the ac signal voltage via its cathode. Valve V2 must provide the full amplitude of the output voltage, since the output valves V3 and V4 only provide the signal with the current needed to achieve the desired power. The latter two valves act as cathode followers, in the same way as transistors are used as emitter followers in the output stage of a transistor amplifier.

This also gives an answer to the frequently hotly debated question, 'How do different types of valves sound?' Since the output valves of a PPP design do not contribute any-



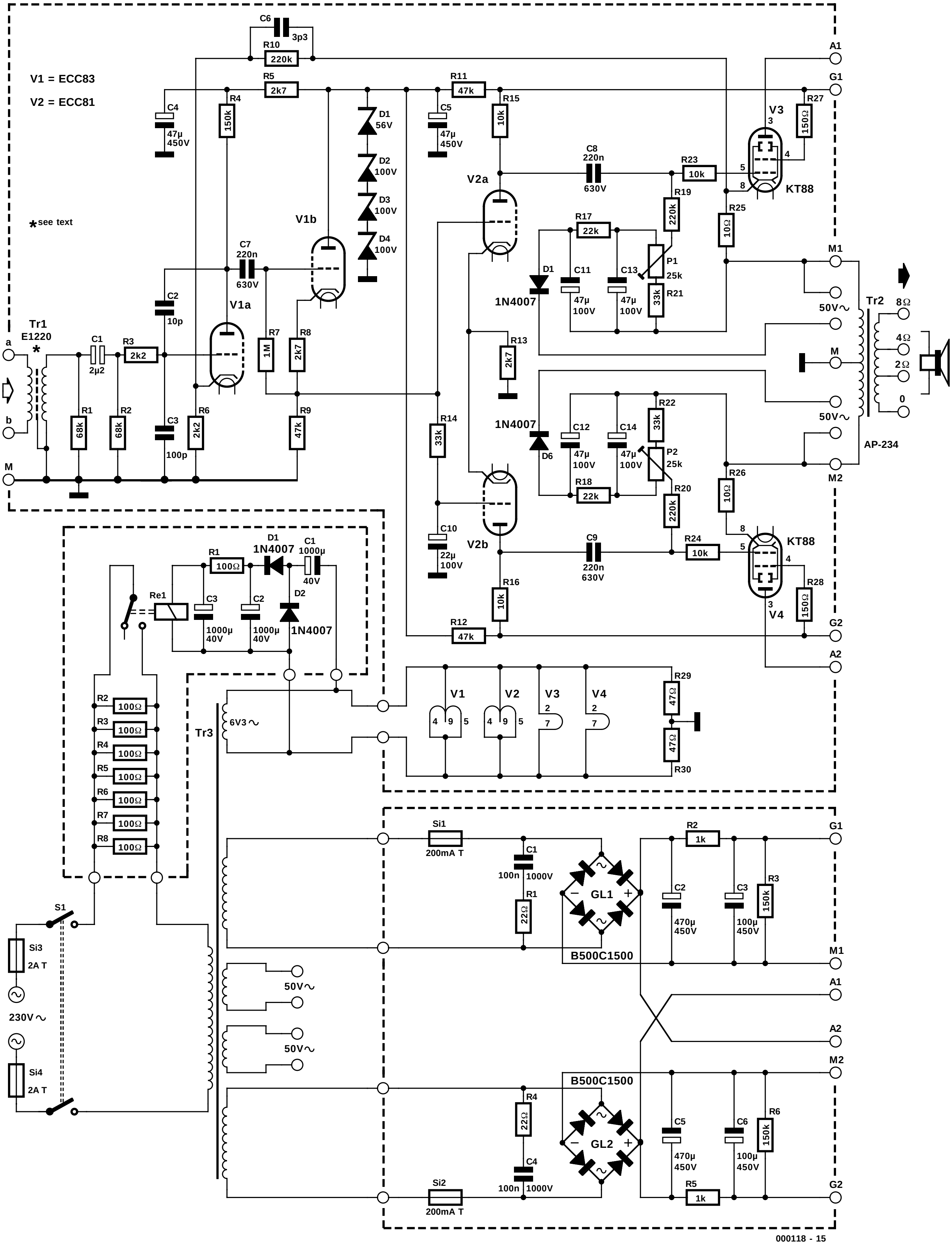
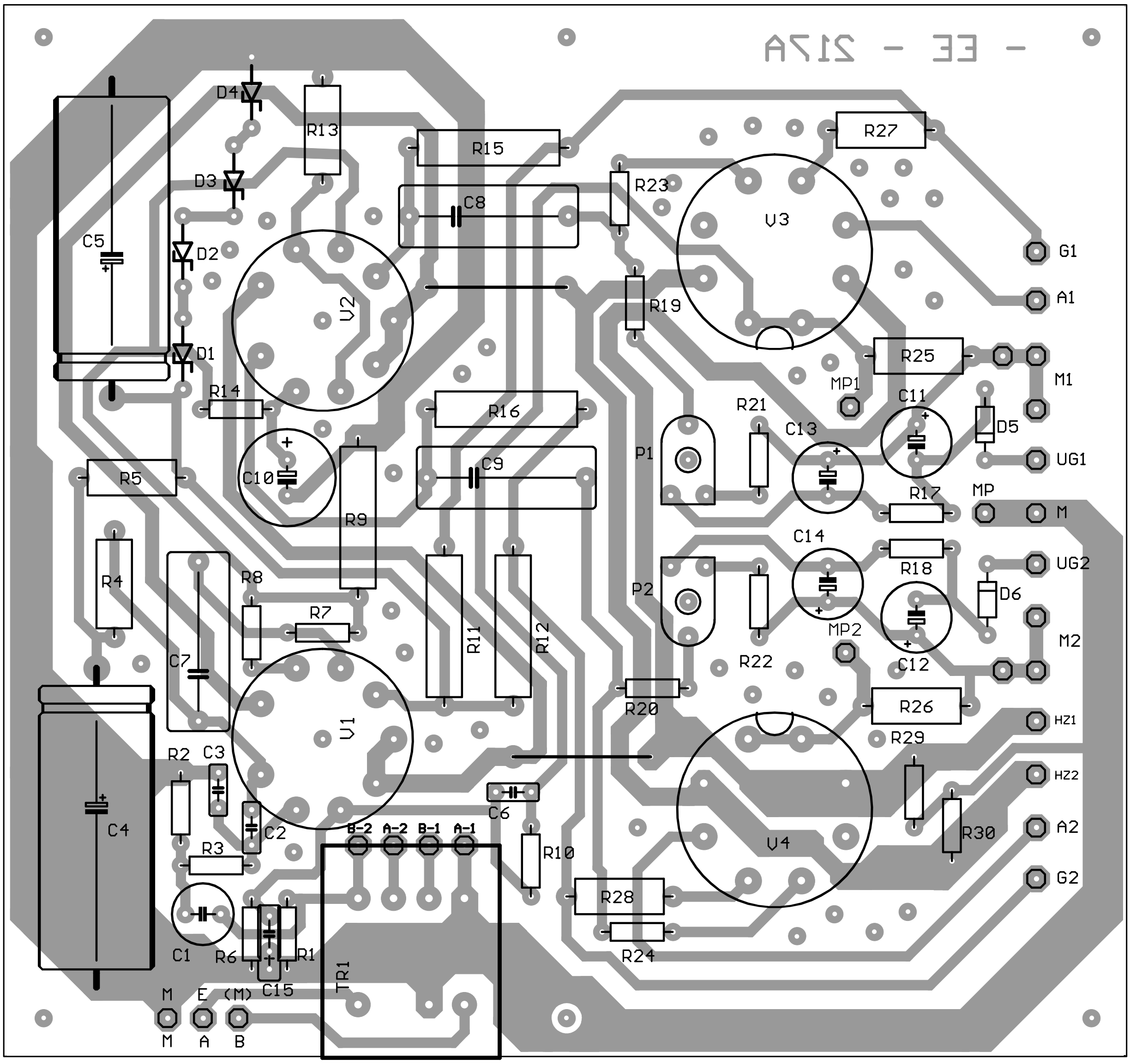


Figure 5. A circuit according to the building-block principle: the equipment can be configured as a monobloc or stereo amplifier, with different types of output valves.



COMPONENTS LIST

Power Output Stage with KT 88 or 6550A

Resistors:

(unless otherwise stated, use metal film types, 0.7W, 1% tolerance;
MO = Metal oxide, 5% tolerance)

R1,R2 = 68kΩ
R3 = 2kΩ2
R4 = 150kΩ, MO, 2W
R5 = 2kΩ7, MO 2W
R6 = 2kΩ2
R7 = 1MΩ
R8 = 2kΩ7
R9 = 22kΩ, MO, 2W
R10 = 390kΩ
R11,R12 = 47kΩ, MO, 2W
R13 = 2kΩ7, MO, 2W
R14 = 33kΩ
R15,R16 = 10kΩ, MO, 2W
R17,R18 = 10kΩ
R19,R20 = 220kΩ
R21,R22 = 33kΩ
R23,R24 = 3kΩ3
R25,R26 = 10Ω, MO, 2W
R27,R28 = 270Ω, MO, 2W
R29,R30 = 47Ω
P1,P2 = preset 25kΩ

Capacitors:

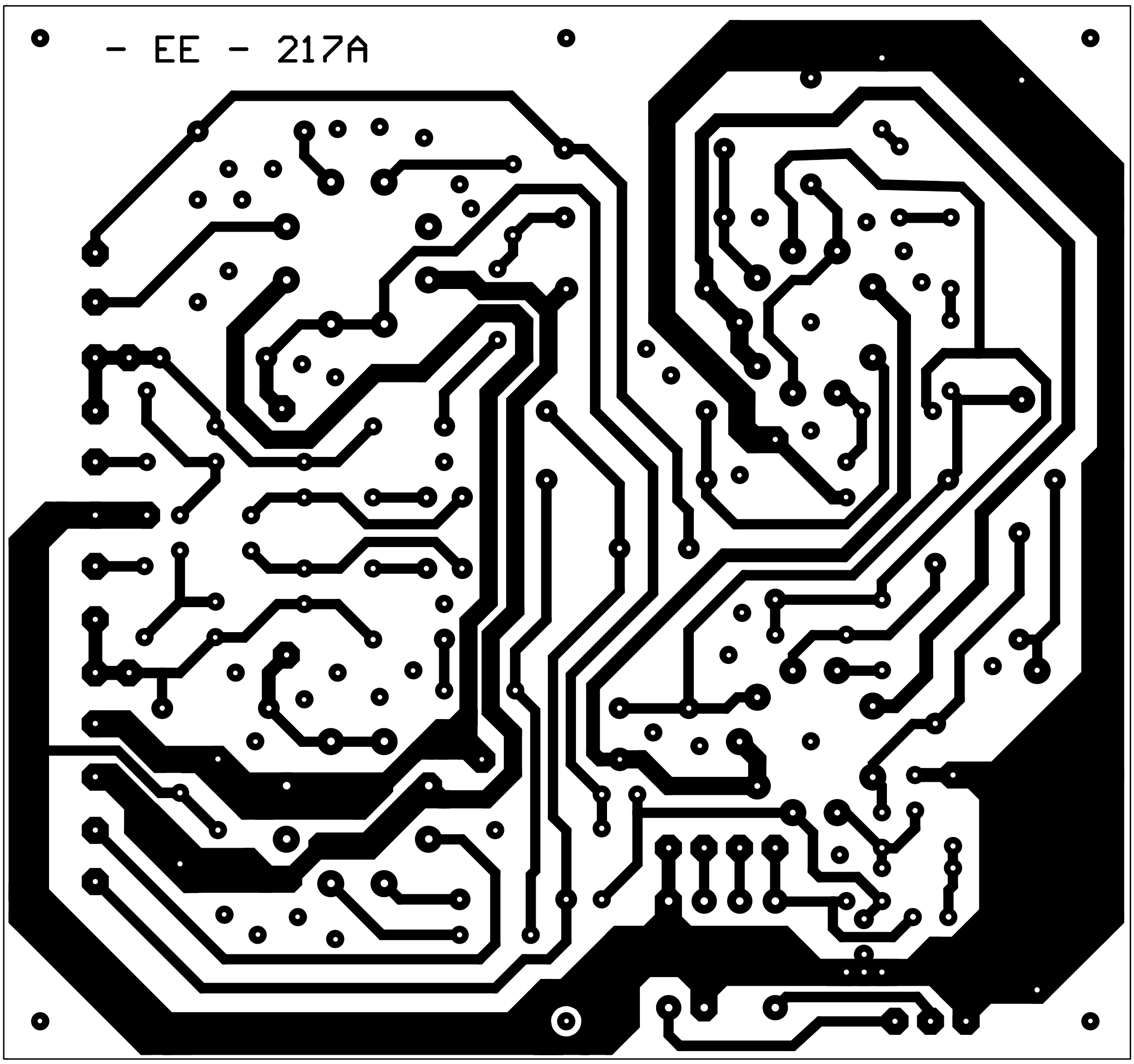
C1 = 2µF2 50V bipolar
C2 = 10pFceramic
C3 = 100pF ceramic
C4,C5 = 47µF 450V axial
C6 = 10-33pF (fit only if parasitic oscillation is noticed)
C7,C8,C9 = 0µF22 630V, MKS 4
C10 = 10µF 100V, lead pitch 5mm
C11-C14 = 47µF 100V, lead pitch 5mm
C15 = 1nF, MKH, lead pitch 7.5mm (see text)

Semiconductors:

D1 = zener diode 56V 1.3W
D2-D4 = zener diode 110V 1.3 W
D5,D6 = 1N4007

Miscellaneous:

V1 (Rö1) = ECC83
V2 (Rö2) = ECC81
V4 (Rö4),V5 (Rö5) = KT88 or 6550 A
Tr1 (Ü1) = E-1220 (if necessary)
Tr2 (Ü2) = AP-234
2 off 'Noval' (9-way) valve socket, PCB mount
2 off 'Octal' valve socket, PCB mount
Tr1 = Mains transformer NTR-P\7 (Mono) or NTR-P\5 (Stereo)



COMPONENTS LIST

Power Output Stage with EL 34

Resistors:

(unless otherwise stated, use metal film types, 0.7W, 1% tolerance;
MO = Metal oxide, 5% tolerance)

R1,R2 = 68kΩ
R3 = 2kΩ2
R4 = 150kΩ, MO, 2W
R5 = 2kΩ7, MO, 2W
R6 = 2kΩ2
R7 = 1MΩ
R8 = 2kΩ7
R9 = 47kΩ, MO, 2W
R10 = 220kΩ
R11,R12 = 47kΩ, MO, 2W
R13 = 2kΩ7, MO, 2W
R14 = 33kΩ
R15,R16 = 10kΩ, MO, 2W
R17,R18 = 22kΩ
R19,R20 = 220kΩ
R21,R22 = 33kΩ
R23,R24 = 10kΩ
R25,R26 = 10Ω, MO, 2W
R27,R28 =150Ω, MO, 2W
R29,R30 = 47Ω
P1,P2 = preset 25 kΩ

Capacitors:

C1 = 2µF2 50V bipolar
C2 = 10pF ceramic
C3 = 100pF ceramic
C4,C5 = 47µF 450V axial
C6 = 10-33pF (fit only if parasitic oscillation is noticed)
C7,C8,C9 = 0µF22 630V, MKS 4
C10 = 10µF 100V, lead pitch 5mm
C11-C14 = 47µF 100V, lead pitch 5mm
C15 = 1nF, MKH, lead pitch 7.5mm (see text)

Semiconductors:

D1 = zener diode 56V 1.3 W
D2,D3,D4 = zener diode 100V, 1.3W
D5,D6 = 1N4007

Miscellaneous:

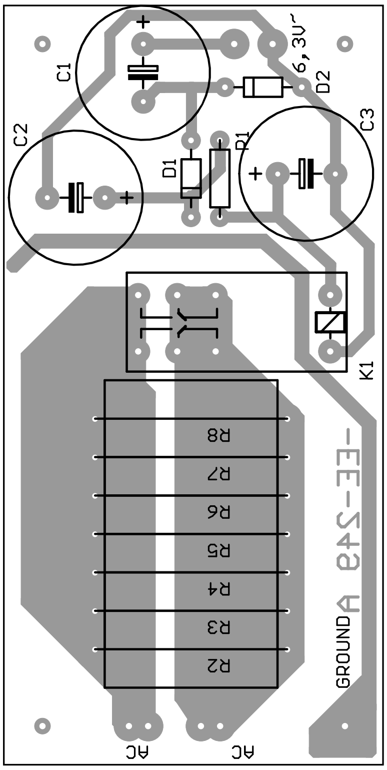
V1 (Rö1) = ECC83
V2 (Rö2) = ECC81
V4 (Rö4),V5 (Rö5) = EL34
Tr1 (Ü1) = E-1220 (if necessary)
Tr2 (Ü2) = AP-234
2 off 'Noval' (9-way) valve socket, PCB mount
Tr1 = Mains transformer NTR-P\6 (Mono) or NTR-P\3 A (Stereo)

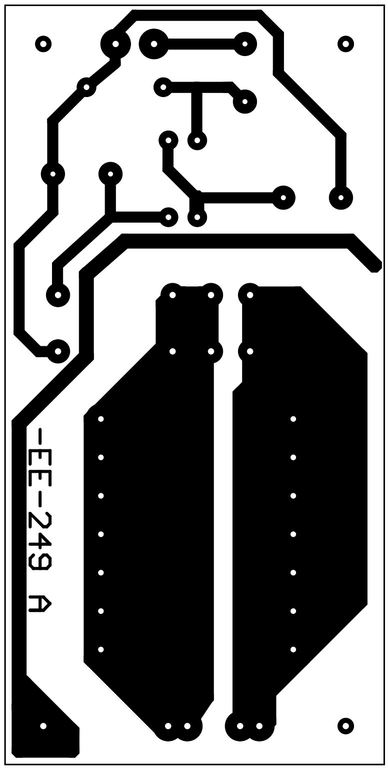
thing to the voltage gain, they have a secondary influence on the sound. With a conventional Class AB push-pull output stage, the characteristics of the output valves affects the sound much more, due to their voltage gain. The component values shown in the schematic diagram have been selected for the KT88, with which a power of 45 W per channel can be achieved. A different set of component values is also shown in the components list for use with the EL34. With this type of valve, 35 W per channel is achievable. Capacitor C15 is an important component. The frequency response drops slightly at high frequencies, due to unavoidable stray circuit capacitances. With C15, the negative feedback can be slightly reduced at the uppermost frequencies. A value of around 1 nF (guideline) should be used for C15, depending on the circuit construction. The frequency response of the completed amplifier must anyhow be checked. When C15 has the proper value, the frequency response at 20 kHz will be absolutely flat, and it may drop by around 1 dB at 100 kHz. However, it is very important to avoid making C15 too large, since otherwise the high-frequency response may become exaggerated.

High voltage and filament voltage

The wiring of the second block, the power supply, appears to be fairly complicated. Since the valves in the output stage are cross-coupled with each other, two separate secondary windings are needed in addition to the filament winding. The grid bias voltage G is generated from the two 65-V windings. The 50-V windings are intended to be used with the EL34, which can manage with a lower grid bias voltage. With 50 V, there is still a good adjustment range for the two trimpots. Each of the screen grids receives its voltage from the opposite half of the power supply. This is necessary to allow the output-stage pentodes to actually operate as such. When either one of the valves is driven fully on, the voltage between the anode and the cathode drops to a low value. If the screen voltage were to also drop by an equivalent amount, the output signal premature would be prematurely limited by the valves themselves.

Each of the screen grid voltages is filtered by 1 kΩ in combination with 100 µF. The driver valve V2 is fed from theses two voltages. The supply voltage for V1 is tapped off via R11 and R12, and additionally filtered by C5. The supply voltage for V1 is stabilised and filtered using the Zener diodes D1–D4. This is particularly important, due to the fact that V1a provides the main amplification. A stable voltage at this location contributes to the consistency of the amplification characteristic.





COMPONENTS LIST
Power-on Delay

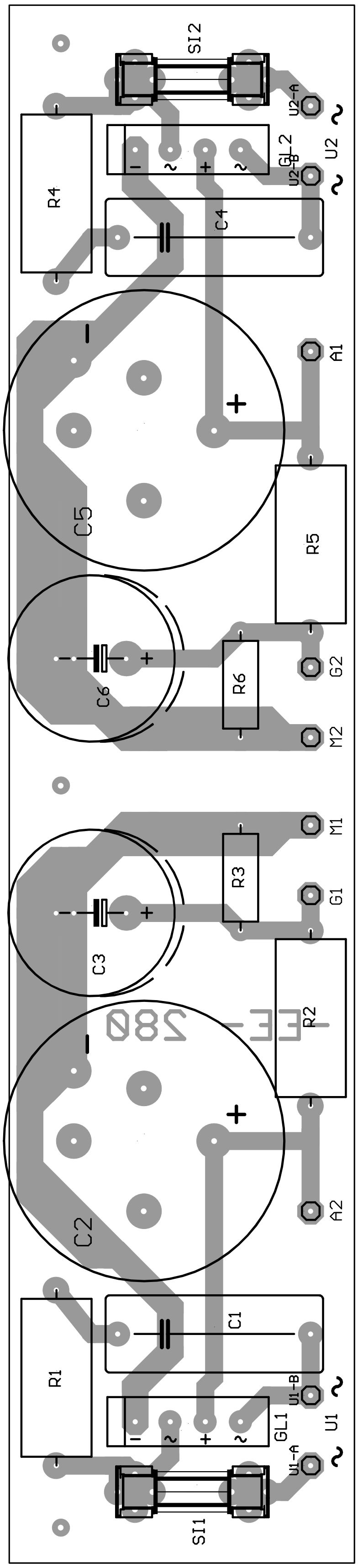
R1 = 100 Ω
R2-R8 = 100Ω, MO, 4.5 W
C1,C2,C3 = 1000µF 40V, lead pitch 7.5mm
D1,D2 = 1N4007
Rel1 = 12V coil, 2 x change-over gold-plated contact, 8A, PCB mount, (Celsea E3208)

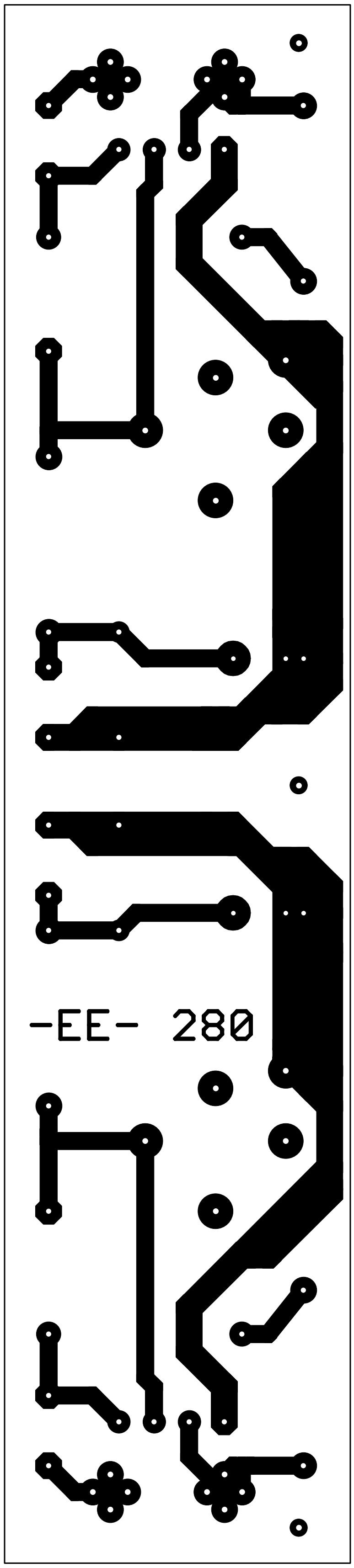
There are RC networks located in front of the high-voltage rectifiers to suppress spikes generated by the rectifiers. If they are not suppressed, these very narrow spikes, with their very broad frequency spectrum, are objectionably audible in the tweeter. If necessary, the values of these components may be modified.

The high voltages are individually and separately fused. When an output valve draws its final breath (and these valves fail the most often), it often draws a high current surge in the process. The fuse prevents any additional damage and interrupts the anode current. Fusing is also a good idea to protect against possible failure of a rectifier or electrolytic capacitor.

There are two versions of the power supply — one that can provide power to a monobloc, and another that can handle a stereo power amplifier. In order to save space and money, different values of electrolytic capacitors are used in the two versions. The compact stereo version employs a 220 µF/47 µF combination, while each half of the monoblock version (which has much more room for the power supply) is fitted with 470 µF/100 µF. If you make your own enclosure, you can always build a stereo power amplifier with two internal monaural assemblies or fit only the larger capacitors. In principle, the design shown here is a flexible set of building blocks.

The 6.3-V filament voltage is connected symmetrically to ground via a pair of 47-Ω resistors. This is essential, since the allowable potential difference between the filaments and the cathodes of the valves is limited. In addition, this drastically reduces the amount of hum noise coupled in via the filaments.





COMPONENTS LIST

Power supply, mono

Resistors:
R1 = 22Ω, MO, 4.5 W
R2 = 1kΩ, MO, 4.5 W
R3 = 150kΩ, MO, 2W
R4 = 22Ω, MO, 4.5W
R5 = 1kΩ, MO, 4.5W
R6 = 150kΩ, MO, 2W

Capacitors:
C1 = 0.1µF 1000V, MKP 10
C2 = 470µF 450V (PCB mount)
C3 = 100µF 450V, lead pitch 10mm
C4 = 0.1µF 1000V, MKP 10
C5 = 470µF 450V (PCB mount)
C6 = 100µF 450V, lead pitch 10mm

Miscellaneous:
Gl1,Gl2 = B500C1500 (500V piv, 1.5 A continuous), flat case
2 off fuse, 0.2 AT (time lag)
2 off fuse holder with cap, PCB mount

Power supply, stereo (one channel)

Resistors:
R1,R2 = 22Ω, MO, 4.5 W
R3,R4 = 1kΩ, MO, 4.5 W
R5,R6 = 150kΩ, MO, 2W

Capacitors:
C1,C2 = 0.1µF 1000V, MKP 10
C3,C4 = 220µF 450V, axial
C5,C6 = 47µF 450V, axial

Miscellaneous:
Gl1,Gl2 = B500C1500, (500V piv, 1.5 A continuous), flat case
2 off fuse, 0.2 AT (time lag)
2 off fuse holder with cap, PCB mount

Switch-on delay

The third block, the switch-on delay circuit, is driven by the filament voltage. The mains transformer is built with an MD core and has very low-impedance windings. This transformer can deliver very high peak currents, and like toroidal transformers, it draws a strong magnetisation current pulse when switched on. In addition, high-capacitance, high-voltage electrolytic capacitors are connected to the secondaries directly after the rectifiers, and when the power supply is switched on these capacitors are empty and must be charged to around 430 V. On top of this, the filaments of the valves represent very nearly a short circuit when cold. Just like incandescent lamps, they exhibit PTC resistance characteristics. Without switch-on current limiting, the branch circuit fuse (or circuit breaker) would blow (trip) when the amplifier is switched on.

Using a voltage-doubling rectifier, the necessary 12 V potential for the relay is generated from the filament voltage. At the

Technical data and measured results

Input sensitivity (for P = 43 W):		0.92 V_{rms}
Maximum output power (THD + N = 1%):		43 W
THD + N	(B = 80 kHz, 1 W, 1 kHz):	0.17 %
	(B = 80 kHz, 1 W, 20 kHz):	0.48 %
S/N (A-weighted) at 1 W:		72 dB
Damping factor (at 1 W/1 kHz):		2.67 (output impedance 3 Ω)

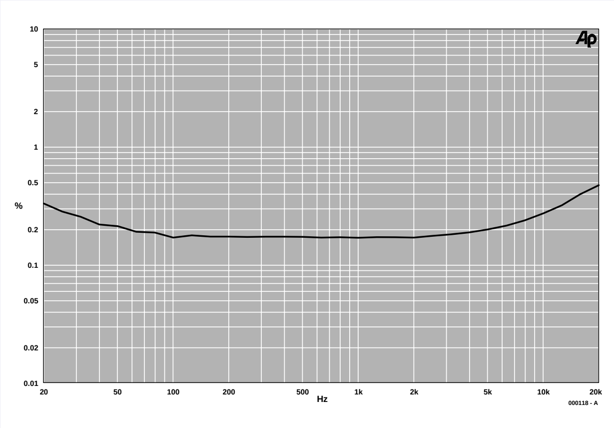


Figure A. THD + N versus frequency at 1 W

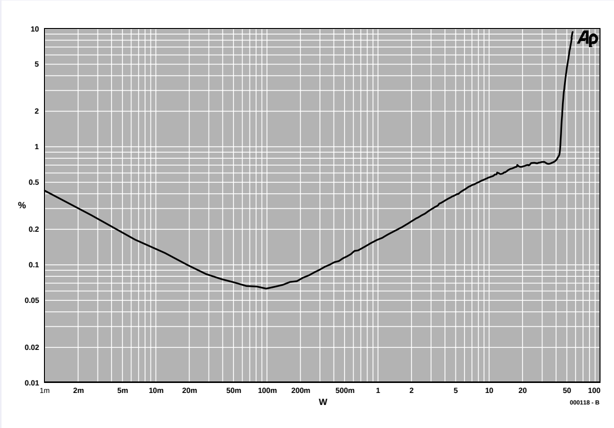


Figure B. THD + N versus output level (minimum value at 100 mW: 0.063 %)

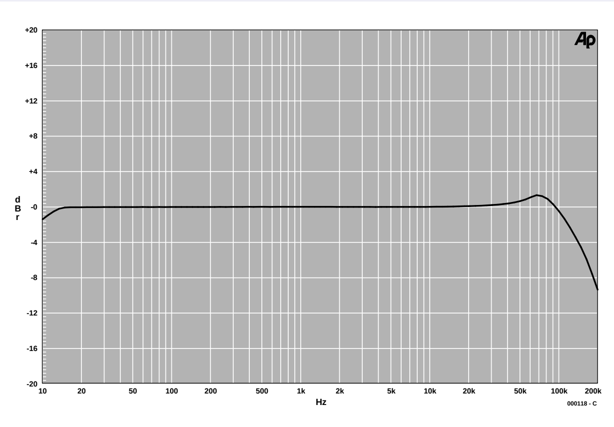


Figure C. Amplitude versus frequency (at 25 W).

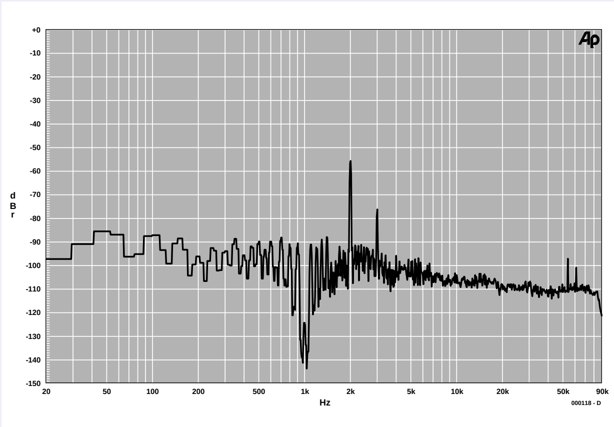


Figure D. Frequency spectrum at 1 W (amplitude of the 2nd harmonic approximately –56 dB)

moment when the unit is switched on, the relay contacts are open, so the current is limited to around 16 A by the seven power resistors connected in parallel. The relay is activated after a delay of around one second, after all the critical processes have already taken place, and it shorts out the resistors. A printed circuit board relay with two gold-plated, parallel switchover contacts has been selected here, in order to achieve high contact reliability and thus a high level of operational reliability.

Construction with flair

At the end of this article, you will find the components lists for the various equipment configurations and types of output valves. Please stick to the components specified in these lists and use only the specified components; this will help ensure that everything goes as intended.

A seamless-welded, polished and bright-nickel-plated chassis has been selected for the enclosure. This gives the finished amplifier an attractive appearance. An aluminium chassis avoids magnetic distortions, which for example can result from transformer excitation. The polishing and bright nickel plating (not chrome plating) give the

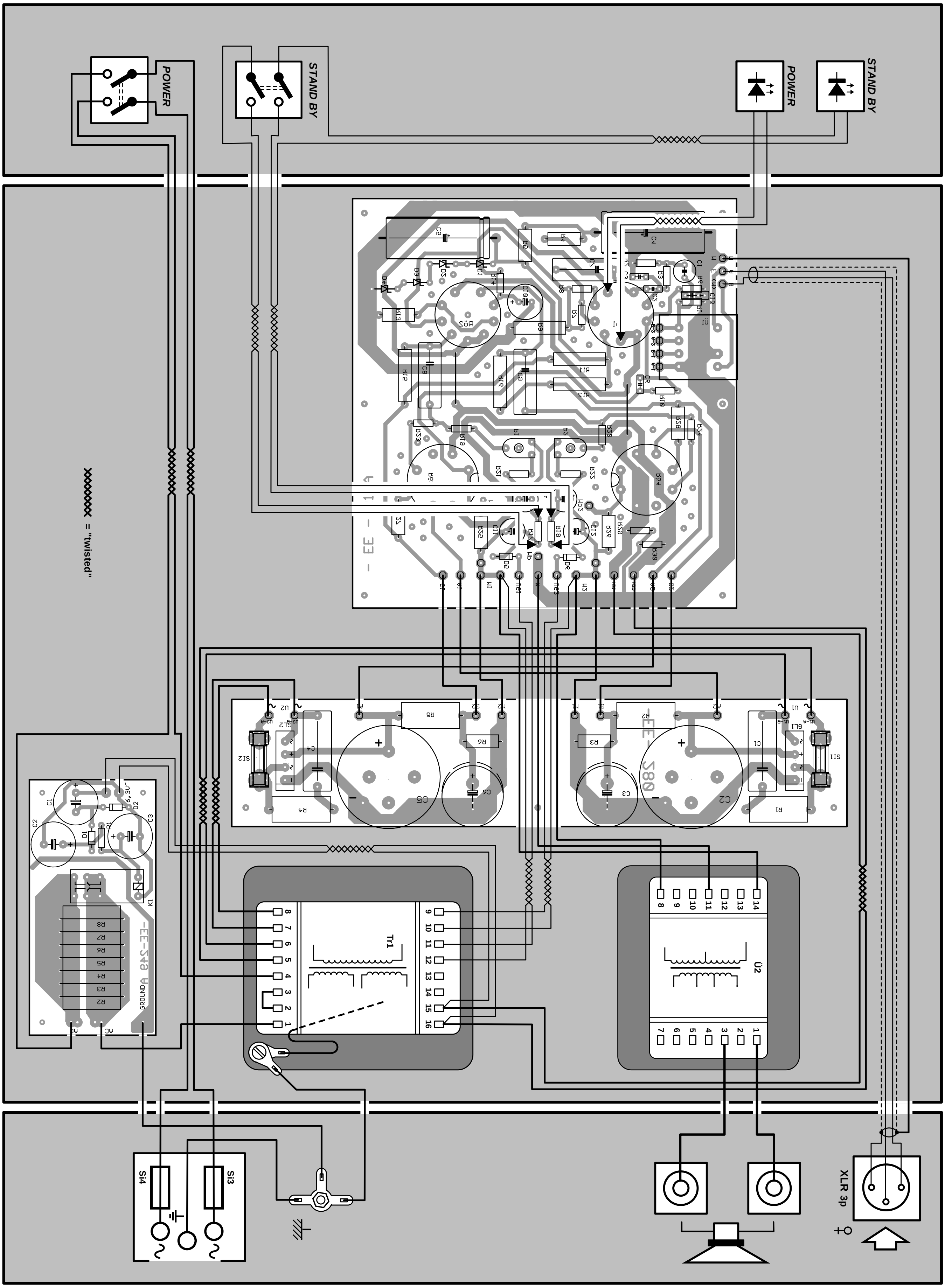


Figure 6. This wiring diagram is a nearly indispensable construction aid.

chassis an elegant appearance. Nickel optically emphasises the warm tone and appearance of the valves, in contrast to the coldness of chromium with its blue cast.

A metallic enclosure that is bonded to the safety earth (E) lead is necessary for reasons of screening and electrical safety. All sockets must be electrically insulated from the enclosure, and all circuit boards must be fitted such that they are electrically insulated from the enclosure. The safety earth lead is connected directly to the enclosure. The mains transformer has a static screen between the primary and secondary windings; this must also be connected to the safety earth lead. The signal ground is connected the enclosure at only one point, at the amplifier input, for potential equalisation. You should wire the amplifier with pairwise twisted stranded wires, using the cross sectional areas specified in the wiring diagram shown in **Figure 6**. Make a copy of this drawing before you start, and mark the connections using a coloured pen as make them. This will help you to avoid forgetting a connection or making a false connection!

Once you have finished the wiring and checked everything, you can carry out the electrical test. Start by removing the fuses for the anode voltages. The filaments should glow visibly after approximately two minutes. After this you should check whether the negative grid bias voltages are actually present at the valve sockets and can be adjusted using the trimpots. Use a voltmeter to measure the potential between M1 and the grid of V3 and between M2 and the grid of V4. Then adjust the trimpots so that the maximum negative voltage is applied to each of the two grids.

Now you can switch off the amplifier and install the fuses for the anode voltages. After switching on the amplifier, use the voltmeter to measure the voltage drops across the cathode resistors R25 and R26. The voltages across these resistors are a measure of the quiescent currents. Working back and forth, adjust the quiescent current each valve to the specified value. This must be repeated several times, since the changed load on the power supply causes the quiescent current of the already adjusted valve to change slightly. Once the quiescent currents have been set, you can use a sinewave generator, dummy load resistor and oscilloscope to check the frequency response and power output. At the very end, you will have to once again adjust the quiescent currents to the specified values. The quiescent currents must always be adjusted with no signal applied. Following this, the equipment is ready for use.

The sample amplifier was built as a set of monoblocks and as a stereo power amplifier (cover picture). With the monoblocks, there is no channel crosstalk, which yields very good localisation, spaciousness and brilliance. The stereo version is more economical, since the cost of the enclosure has to be paid only once and the power supply is less expensive.

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PCB Service

As with the Valve Preamplifier published last year, ready-made printed circuit board for the PPP Valve Power Amplifier are not available through the Publisher's Readers Services. Ready-made boards, with or without components, are available from the author:

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FAX: (+49) 7324 2553
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We can only answer questions or remarks of general interest to our readers, concerning projects not older than two years and published in *Elektor Electronics*. In view of the amount of post and email received, it is not possible to answer all correspondence, and we are unable to respond to individual wishes and requests for modifications to, or additional information about, *Elektor Electronics* projects.

How much current from the USB?

Dear Sir — in the article 'USB Audio-DAC' (December 2000), you state on page 45 that the current consumption of the circuit (60 mA), is such that it was not considered appropriate to power it from the USB port, which has a maximum spec of 100 mA in this respect. According to information I have available, a USB device may draw up to 500 mA. For example, the Canon CanoScan N650U gets its complete power requirement of 2.5 watts (500 mA x 5 V) from a USB port. This works just fine, even if an additional USB camera is connected. I would therefore be interested to know where you got the 100 mA spec from?

M. Radde (by email)

The specification '100 mA' is the current a USB port is always capable of supplying. During initialisation, a higher current is allowed. Once initialised, a USB device requiring more current may draw more current after reporting its requirement to the operating system. The limit of an USB port is 500 mA, and no further devices may be powered once this current is reached.

Depending on their current requirement, USB devices fall into one of three classes:

1. low bus powered
2. high bus powered
3. self powered

Devices from the first two classes draw all necessary current from the USB port. Self powered devices, on the other hand, have an internal power supply and require little or no power from the USB. A unit load of 100 mA has been defined for the USB. Devices from the first class (low bus powered) are only allowed to draw one unit, i.e., 100 mA. The same goes for high bus powered devices (class 2) when in the pre-configuration state. Up to 5 load units are allowed in class 2 devices when configured.

The amount of current an USB device is allowed to draw from a port depends on five factors:

A. Device status
– power-on: max. 1 load unit
– configured: 1 load unit (100 mA) for low-power devices, or up to 5 load units (500 mA) for high power devices provided they have been allowed to do so by the OS
– suspended: not exceeding 500 mA including current through pull-up to D+ or D–.

B. Hub type used to connect the device. Up to 500 mA allowed using a hub with internal power supply. If the hub is bus powered, this is reduced to 100 mA. A hub is a PC is normally considered self-powered, while a hub in a portable device is bus powered. When a device draws to much current, it may happen that the USB goes along by supplying more current than is strictly allowed. None the less, it is recommended to provide for an internal power supply.

C. Configured and running: current consumption should not exceed the level reported by the device (by means of the enumeration in the configuration descriptor) during the configuration phase.

The same restrictions in respect of current demand apply to devices with an internal powered supply (self powered). Good separation is essential between the USB and the external power supply. In the event of the internal supply failing, the device may not present a heavier load the USB than 'promised' during configuration (i.e., not more than one load unit of 100 mA when configured as self-powered or low power bus powered).

CD Recording Quality

Dear Editor — roughly a year ago you had an article in Elektor Electronics about CDs that were supposedly recorded with overdriven signals. An oscilloscope was used to look at the output signal of a CD player playing Jean Michel Jarre's 'Oxygen 7-13'. It was clear that the peaks of the waveform were flat where the signal was greater than what could be reproduced by the CD. This is also known as clipping.

Using an oscilloscope in this way can't really tell us how much the CD player affects the signal quality. Between the CD and oscilloscope will be an oversampling digital filter, a digital-to-analogue converter and finally an analogue filter. The most accurate way to check a CD is to extract the track digitally and store it as a WAV file. This will be an exact digital copy of the CD, assuming there were no read errors. The creation of the WAV file shouldn't be a problem since it is a function that is found in most current CD 'rippers'. A computer program can then be used to analyse the contents of this WAV file.

A WAV file consists of a header followed by the thousands of samples that make up the digitised sound. The 16-bit format used on CDs gives these samples a range from -32768 to 32767. In order to have a closer look at these WAV files I have written a program that can analyse them. As the first step, the program creates a histogram of the WAV file. This counts how often each sample value occurs in the WAV file, and gives a display as shown in Figure 1.

The values near zero occur most often; the more you go to the extremes of –32768 and 32767, the less often they occur. If there was any clipping at the recording stage, all strong signals will be rounded to the extremes of -32768 and 32767, which results in two peaks at the ends of the histogram. This way you can see if there is any clipping present and at what level (it is possible

CORRECTIONS & UPDATES

Free Downloads now supplied as zip files

Having obtained a licence to distribute .zip files via the Free Downloads section of the Elektor Electronics website, we are pleased to announce that all previously supplied .exe self-extracting archive files have been replaced by .zip equivalents. The program needed to unpack these files is called Winzip and may be downloaded free of charge from http://www.winzip.com.

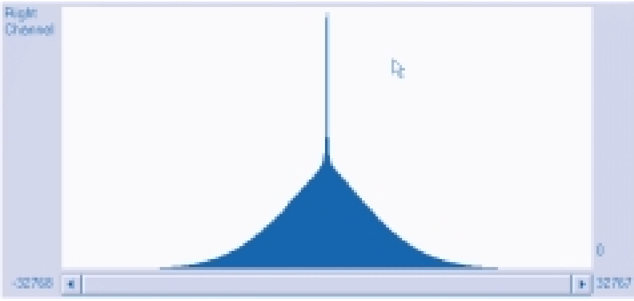
MCS BASIC-52 V1.3

On page 22 of the article, the sentence 'The PROG commands are also redundant, so they have been deleted' should read 'The FPROG commands…'

MIDI Lights & Slide Control

The SAB80C535 used in this project may is not easily obtainable and may be replaced by the more commonly available and fully compatible type SAB80C515-N.

1



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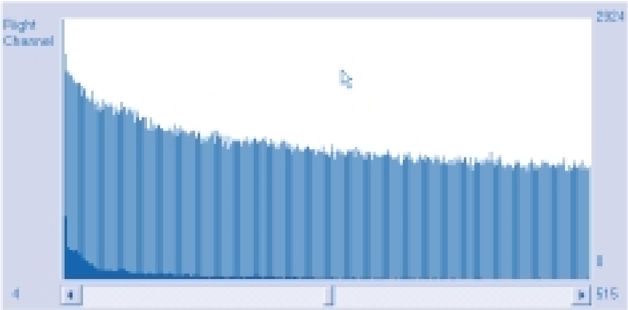

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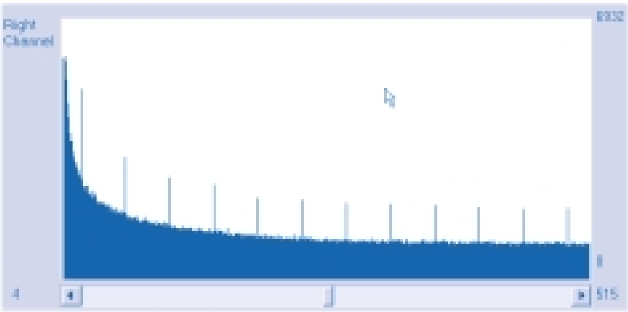


for it to occur at different values than -32768 and 32767, due to badly set up A/D converters).

Next, the program displays the actual waveform stored in the WAV file. This window has a search facility that searches for clipping values, as determined by the previous histogram function.

I have used this program to analyse the track "...Baby One More Time" by Britney Spears, from her CD "...baby one more time". It was obvious that there was a lot of clipping in this track, but the display returned by the histogram was the most surprising.

This is shown in Figure 2. It contains the number of occurrences in the WAV file for samples with values from 4 to 515. After every two values that occur over a 1000 times, there follows a value that rarely or never occurs at all. You could only get such a histogram when a 16-bit recording is re-recorded after amplification by a factor of 1.5. Samples with values of 1, 2, 3, 4, 5, 6, 7 etc. are multiplied by 1.5 to give 1.5, 3.0, 4.5, 6.0, 7.5, 9.0, 10.5 etc. These values then have to be rounded to integers before being written to the CD. This gives the result 2, 3, 5, 6, 8, 9, 11 etc. You'll see that every third value is missing from this series of numbers. By looking at the waveform where clipping occurs, it is possible to deduce by how much the recording of the CD was overdriven. I estimate this to be a factor of about 1.5.

From this I conclude that the original mix, which was recorded perfectly, was remixed with an amplification of 1.5, causing clipping in the new recording.

The track "Gloria, lonely boy" on the CD "Metamorphose" by Jean Michel Jarre was then analysed, in order to see if his latest CD had signs of clipping. Here too it seemed that the recording was overdriven. But once more it was the histogram that returned some fascinating information (Figure 3).

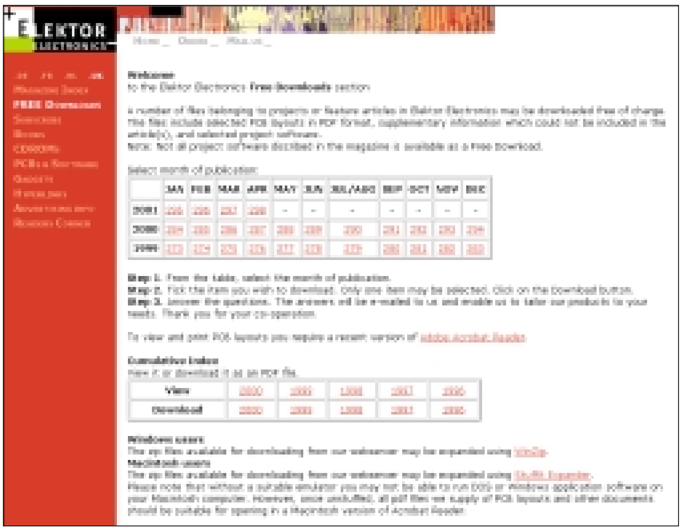
The smallest and largest samples found in the recording were -32022 and 32021 respectively (0.977 of the maximum range). These were also the levels at which clipping occurred. From the histogram it can be seen that every fortieth value is twice as large as its neighbours are. From these observations you can conclude that this CD is a remix of an original CD recording. The samples have been multiplied by 0.977 before being converted back into CD format.

This raises the question why this meaningless conversion was carried out on the recording. The only reason I can think of is that the remix was made to hide the fact that the recording was overdriven. Anybody who uses an oscilloscope to look for clipping would set the trigger level at the largest possible signal level, about 99.9% of the maximum signal possible. The remix would cause the signal to remain below this level so the oscilloscope would never trigger and the clipping would remain undetected. The person making the remix would have known that clipping occurred and was determined to hide this fact. The correct procedure would have been to make a slightly softer remix from the master tape and avoid any clipping that way.

The record industry is currently promoting the new Super Audio CD (SACD), which has an improved quality. They would be better off stopping the messing around with the recordings. These fudges also sound bad in SACD format and making careful recordings in CD format will result in bigger improvements than the change to SACD format.

A. Kappert, Netherlands

We've said it before and we'll say it again: We think that the price of a CD is high enough to expect the best quality recordings. Initiatives such as these by quality conscious music lovers should therefore be applauded. Mr. Kappert has been kind enough to make his analyser program available to Elektor Electronics readers free of charge. The program can be downloaded from our website at www.elektor-electronics.co.uk. It can be found as 'CD-analyser' on the Free Downloads page, May 2001 items.



Introduction to TCP/IP & Embeddded Internet (1)

learn about remote control via the Internet

By P. Stuhlmüller

What has been for some time possible with a 'big' system can now also be realised with a small system: microcontrollers have become Internet-capable, since they have learned how to 'speak' TIC/IP. In this article, the author first describes a few important fundamentals of Internet communications that are relevant to our target application.

Embedded Internet

This is the first of a three-part series. The first article, in this issue, discusses the fundamentals of Internet communications, the second article (which will appear in next month's issue) deals with programming a TCP/IP stack, and the series will culminate in the Summer Circuits issue (July/August) with the first '*Elektor Electronics* Internet Project'. The subject of this project is a universal mini-webserver (Web-I/O) with the following noteworthy features:

- 80186 processor core
- integrated Ethernet interface
- 2 serial Ports
- I²C interface
- 16 digital inputs
- 16 digital outputs
- liquid crystal display
- expansion connector for custom hardware development
- DOS-like multitasking operating system
- TCP/IP stack with PPP, HTTP, FTP, SMTP and Telnet
- many free sample programs available on the Internet
- easily programmed using various DOS compilers

Don't miss this special project!

For years, the control and monitoring of complex systems and processes from a central control station has been a standard and proven practice. In the present era of high-speed global networking, the increasingly smaller role played by spatial separation can be regarded as fully logical. Nowadays, all-encompassing data networks at local and global levels are part of the infrastructure of every modern industrial society. For a long time already, data lines have been used to link travel agencies to airlines, ticket agencies to promoters and information systems to their operators, although this is not always externally evident. Communications satellites and terrestrial transmitters are also controlled remotely, of course, and many industrial and research facilities would not be conceivable without remote control and remote maintenance, due to the dangers that their emissions pose to humans. The list of systems that are dependent on long-distance data exchange could be extended almost at will. The methods and processes of data communications are generally adapted to their intended uses, and transmission security is often an important consideration in the implementation of the transmission paths.

The idea that the worldwide data and information transmission medium called the Internet is also suitable for

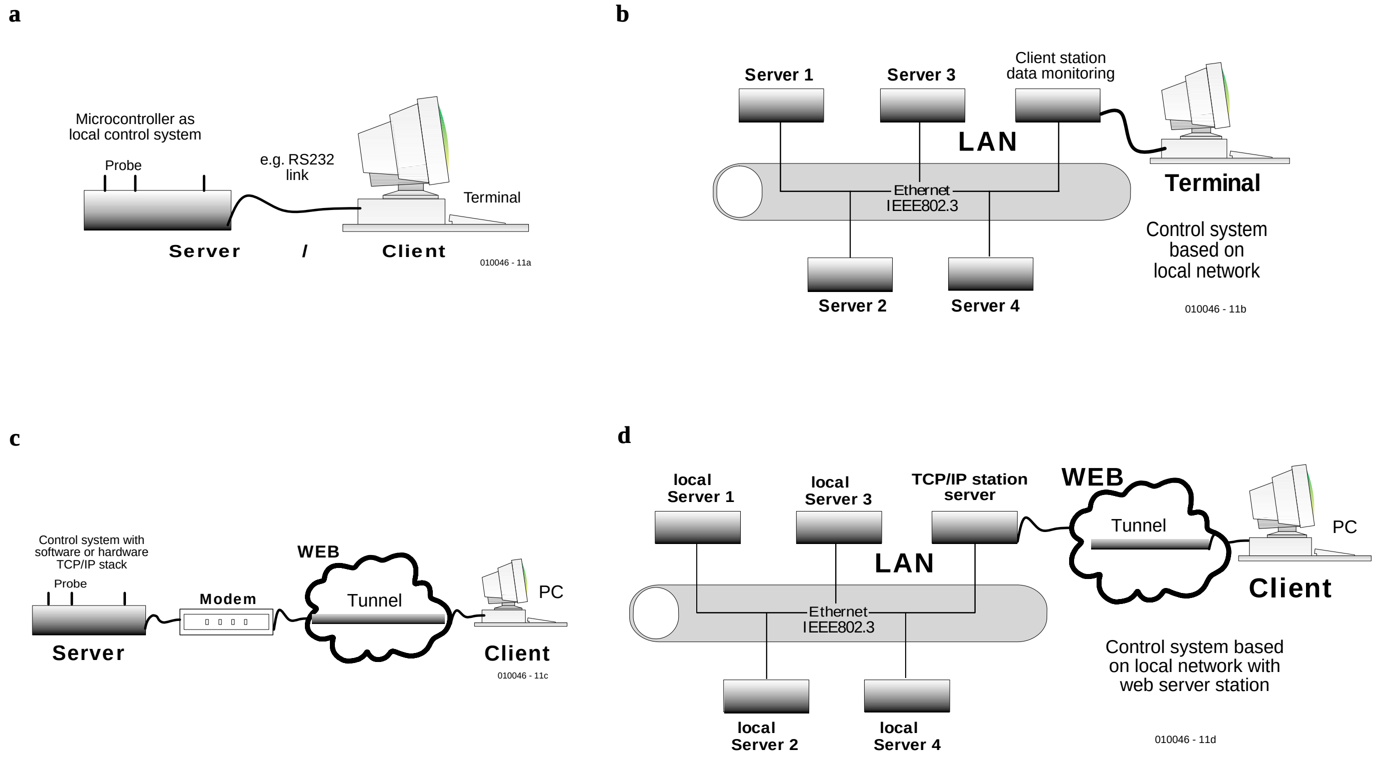


Figure 1. Client–server system as an isolated system (a), as part of a LAN (b), as a Web-capable individual system (c) and as part of a Web-capable LAN (d).

many remote control and monitoring tasks is certainly not new; in fact, such applications were envisaged at the very beginning. The systems that are supposed to communicate with each other via the Internet only have to speak the language of the Internet: they must master a 'stack' of protocols that is collectively referred to as 'TCP/IP'. In the rest of this article, we take a closer look at the individual aspects of TCP/IP that are important in our context.

Client–server model

The meaning of the term 'client–server model', which originates from network technology, can be quickly explained using a practical example. Let's assume that we have a measurement system that must be operated from a more or less remote location and that the measured values (data) must be evaluated at this remote location. Here the measurement system is the server (data supplier) and the control and evaluation facility is the client (data consumer). Four different types of client–server configuration are sketched in **Figure 1**. The simplest type, as shown in Figure 1a, actually needs no further explanation. In Figure 1b, several servers are networked together by means of a local area network (LAN), and they are collectively linked to the control and evaluation centre (with the client). Figures 1c and 1d correspond to the configurations shown in Figures 1a and 1b, with the decisive difference that here the transmission paths run through the Internet in the form of 'tunnels'.

A measurement data system that communicates via the Internet, and in which Internet capability is already present in the system itself (in other words, this capability is 'embedded' in the system), has the following advantages:

– It is possible to access the system via the Web from every corner of the world (and even from outer space).
– Without regard to place and time, the system can be monitored and controlled by specialists who only have to concentrate on these tasks.
– Security mechanisms, such as data encryption and passwords, prevent unauthorized access and improper evaluation of the data.
– At the operator interface level, the system behaves exactly as though it were installed locally.
– Naturally, there are also a few disadvantages:
– The cost and complexity of the software for the system are significantly higher than for a locally installed system.
– The implementation of Internet capability is generally only possible with high-performance hardware, which significantly increases costs.
– Data transport via the Internet results in operating costs (provider charges) that would not accrue with a local installation.

Paths through the Internet

In order to be able to transport measurement data or any other desired date via the Internet, the data must be packaged into suitable 'shells'. This is necessary if only because the transport paths in the Internet are utilised by many servers and clients *in common*, for predominantly economic reasons. The data thus do not travel as a single bundle; instead, they travel in small individual packets from the server to the client via a relatively large or small number of intermediate stations. The

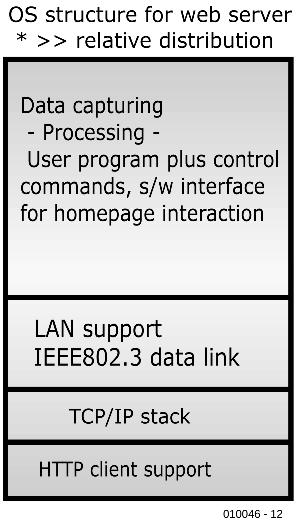


Figure 2. Software modules of a mini-webserver.

requirements document that defines how the utility data are to be split into partial shipments and packaged is called TCP/IP, or in full, 'Transport Control Protocol / Internet Protocol'.

Figure 2 shows schematically how the individual tasks are distributed in a client–server data acquisition system that communicates via the Internet. Aside from the hardware and software for acquiring and preparing the measurement data, it is necessary to have a LAN support module, which primarily coordinates the local networking of the measurement equipment, a TCP/IP support module that enables data transport via the Internet, and an HTTP support module that looks after additional necessary or desirable functions. Since the measurement data are divided into packets that travel along the data highway along with many other unrelated packets. Each packet must be identified by an unambiguous and unique recipient address. For both the client and the server, the handling and processing of addresses is a key task that must be mastered by every system that communicates via the Internet.

Internet addresses

Every Internet address consists of four blocks of digits in which the numbers from 0 to 255 may occur, which corresponds to a word width of 32 bits. The address is composed of two parts: the network address, which is the same for all stations connected to a single network, and the individual host address. Internet addresses are divided into five classes, labelled A through E, which have different schemes for splitting the available 32 bits into the network address and the host address. **Table 1** summarises these classes.

For local networks that are constantly connected to the Internet, the address must be registered with InterNIC (the Internet Network Information Center). Other types of Internet users, who only intermittently communicate via the Internet (using a modem connection, for example) can be assigned an address dynamically, which means for the duration of the connection.

TCP/IP service levels

The design of the software structure of a client–server Web model boils down to a division into a number of service levels. The Reference Model, which has been given the name 'Open System Interconnection' (OSI), is a representation of the software modules that are mandatory for a client–server Web model. In the case of a mini-webserver, the Reference Model reflects all of the activities that the microcontroller must carry out in order to feed the utility data into the Internet via a modem or other data link in the proper form for the Internet. This structure is illustrated in detail in **Figure 3**. The TCP/IP module has the most important role for data transport via the Internet. It adds the recipient address to the data packets that are to be sent, and in addition it adds 26 values that have a prescribed sequence and represent various delivery features. The structure shown in Figure 3 is vertically oriented, so it runs from the top to the bottom.

Another important aspect of the client–server Web model is the communications management function, which for example governs whether the client or the server initiates communications. Normally, the role of the 'caller' falls to the client, for specific reasons (e.g. protection against manipulation), but the role assignment can also be reversed. Of course, this must first be implemented in the software.

Into and through the Net

The computers that are involved in the Internet, and their operating systems, belong to a wide variety of different worlds: in addition to DOS and Windows computers, Unix, SunOS and AS400 systems, to name only a few, there are also large computers, some of which are familiar only to specialists. Independent of the type and size of the system, however,

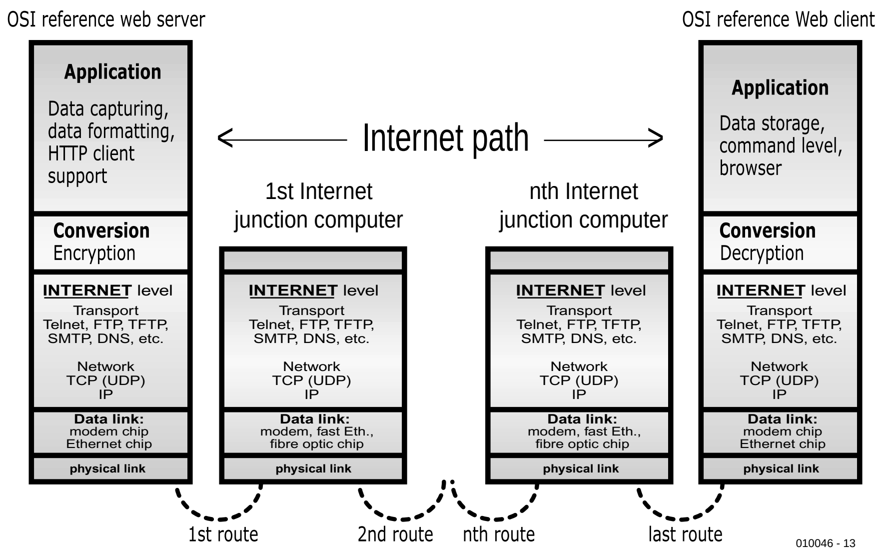


Figure 3. OSI Reference Model for Web-based control systems.

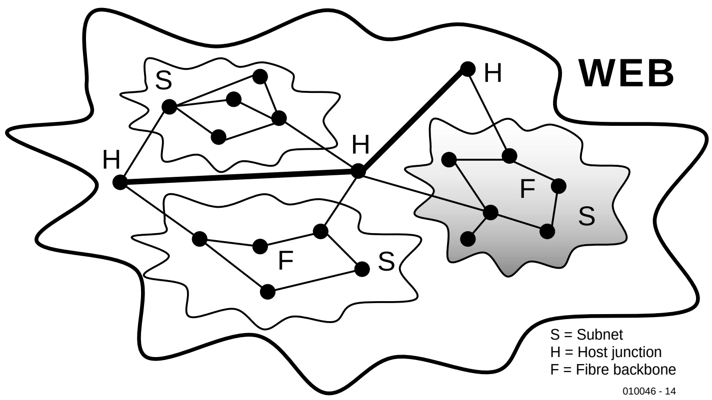


Figure 4. Schematic representation of data transfers in the Internet.

Table 1. Division of the Internet addresses in groups A through F

Class	Network	Hosts	Address range
A	1 – 126	16 777 214	0.0.0.1 to 127.255.255.255
B	128 – 191	65 534	128.0.0.0 to 191.255.255.255
C	192 – 223	254	192.0.0.0 to 223.255.255.255
D	224 – 239		Reserved for multicast addresses.
E	240 – 254		Reserved; presently unused.

many procedures are the same for all of them. A microcontroller-based mini-webserver with its own operating system connects to the Internet in exactly the same manner as a mainframe machine, to the extent that the *input conditions* are the same. Both systems must go through the same log-on procedures if (for example) they are linked to the external data world via the So bus of an ISDN connection.

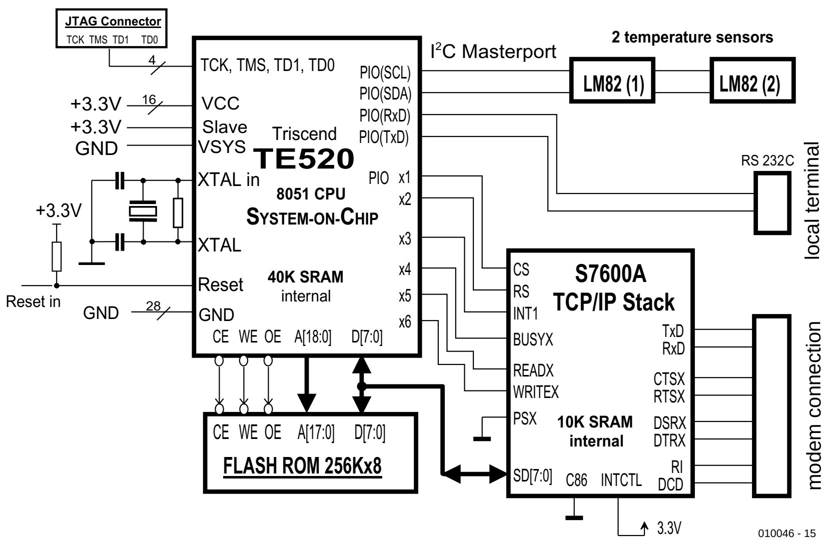


Figure 5. Basic circuit diagram of an isolated mini-webserver using the Seiko S7600A as the TCP/IP hardware stack.

The access procedure via the ISDN So bus is standardised as follows:

– A connection to the node computer of the provider is set up using the Telnet network protocol. This connection initially has the nature of a terminal connection.
– The data packet is sent to the node computer using the platform-independent FTP protocol (File Transfer Protocol). This link is a point-to-point connection.
– From this point onwards, TCP/IP (or more precisely, IP) is involved in the data communications. The node computer of the provider evaluates the file header and specifies the path to the next node computer. This routing function is looked after by a software module that works with routing tables. The routing tables can be generated manually (static routing), but they can also be continually updated to match actual circumstances (dynamic routing). Besides this, it is possible to have the routing be based exclusively on hardware. The 'IP tunnelling' method is used for transporting data packets from one node computer to the next one.
– At latest after the packet has been handed over to the first node computer, a confirmation is sent back to the sending computer. The processes in the node computer are rather complex, so here we will describe only the most important feature: for every possible path to the recipient, a maximum data packet length is determined, and this is a component of the input conditions for the link in question within the network. This is because each link has a limitation on the maximum length of data packets that can be transmitted, which means that if the data packet is too long, it must be split into smaller pieces. Since the transmission speeds of the various paths through the network that are employed can vary considerably, there is no guarantee that the partial packets will arrive in the correct order at the recipient. It is the task of the recipient software to reconstruct the original file from the partial packets.

For clarification, a section of the Internet is sketched out in **Figure 4**.

TCP/IP header

From Tables 2a and 2b, we can see which parameters are contained in the TCP and IP parts of the file header. All of the header words have a width of 32 bits, and they are arranged in the file header so that they can be evaluated as quickly as possible. The programmer of a client–server Web system must without exception use the values in the file header that are applicable to the system that he or she has developed! If the client–server

Table 2a. TCP entries

Field Designation	Length (bits)	Meaning
Source Port	16	Sender address of the process or data service of a higher-level protocol. Together with the IP address, pointer to a procedure socket. Data packet at the Telnet port of the recipient. The Telnet port communication sender/recipient is NOT a component of the Sender IP address.
Destination Port	16	Socket process or data service of the recipient computer. The sockets on the sender and recipient sides remain active as long as the connection exists.
Sequence Number	32	Data flow; sequence number of the first byte in the data packet.
Acknowledge Number	32	Data flow; confirmation of the receipt of all sent bytes.
Data-Offset	5 – 15 usually 5	Number of 32-bit words in the TCP from this word onward, including the data set (always a multiple of 32 bits). Reserved 6 For future applications.
Condition-Control-Flags	6	URG: urgent pointer; ACK: the Acknowledge number means important/unimportant (1/0); PSH: handover to a higher-level protocol (yes/no); RST: terminate existing link (yes/no); SYN: to the recipient: set up connection (yes/no); FIN: sender indicates the end of the connection.
Windows-Limit/Size	16	Flow control between the sender and the recipient of the data block in the original datagram.
Checksum	16	Checksum of the 96-bit pseudo-header (32+32+8+8+16 bits) IPSA + IPDA + empty protoident + TCP segment.
Urgent-Pointer *****	16	Positive offset of the sequence number.
Service-Options ******	Var.	variable Type, length and options data in the datagram.
Padding	8	Fill data to ensure that the header has a 32-bit format.
Data Set	32	

* Indicates that the TCP data are highly urgent. The pointer always points to the end of prior data. If the value of the pointer is added to the sequence number, the result is the sequence number of the last urgent TCP data set. Data of this sort are interrupts of breaks that are transmitted by an opposite party.

** Field length; depends on the type, length and special data. Every TCP implementation must support all defined TCP options.

Table 2b. IP entries

Field Designation	Length (bits)	Meaning
Version	4	Version 4 IP header version, currently 4.
IHL	4	Internet Header Lenght.
Service Type	8	IP datagram services:, delay, throughput, reliability.
Total Length	16	Datagram length, IP header + data block.
Identification	16	Identifier; assignment of the fragment in the datagram.
Flags	3	DT: fragment, MF: last fragment / no fragment. Fragment Offset 13 Position of the data fragment relative to the start of the data block in the original datagram.
Time to Live	16	Remaining lifetime of the datagram.
Protocol *	16	ICMP, IGMP, TCP, EGP, UDP, XNS-IDP, ISO-IP, OSPFIGP.
IP Header Checksum	16	Checksum for the IP Header (ones-complement).
IP Source Address	32	Source address of the server (sender).
IP Destination Address	32	Target address of the client (recipient).
Options **	7 min.	Classes: Control and Debugging, IT Timestamp.
Padding	8 - 24	Fill data to ensure that the header has a 32-bit format.

* ISO IP (ISO Internet Protocol), UDP (User Datagram Protocol).

** Copy Flag in the fragmentation process.

Web system is a component of a local area network (LAN), then normally only the station that sets up the link to the Internet has to run TCP/IP. If all stations were to have direct access to the Internet, the software overhead would be quite significant.

TCC/IP in a chip

Everything that is necessary for a typical mini-webserver has already been illustrated in Figure 2. For an isolated mini-webserver, the necessary functions could basically be implemented using a suitable microprocessor and the associated software. An interesting alternative to this classical solution is provided by Seiko Instruments in the form of an IC that renders programming a TCP/IP stack redundant. Inside the S7600A chip, the TCP/IP stack is already 'cast' in hardware in a ready-to-use form. Here Seiko Instruments has implemented the current version (Version 4) of the TCP/IP protocol suite in an IC with UDP (User Datagram Protocol) and PPP (Point-to-Point Protocol). One limitation that should mentioned is that UDP is somewhat simpler than TCP and not quite as reliable, although the working speed of UDP is higher.

The S7600A incorporates an integrated serial interface (UART), which can be connected to a modem via an RS232 line driver. The 8-bit parallel port of the S7600A is compatible with standard microcontrollers, and if necessary it can easily be adapted. Figure 5 shows the basic diagram of a mini-webserver using the S7600A. The TE520 is a configurable microcontroller with an 8051 core, 40 kB of SRAM and 2048 freely programmable logic cells. Two LM85 temperature sensors are shown connected to the I^2C bus by way of example. What they measure can be queried from continents away via the Internet!

Another solution is based on the well known Zilog Z80 microprocessor, which nowadays can work at clock frequencies up to 80 MHz, can address up to 16 MB of peripheral and memory space and also offers on-chip DSP functions. With this device, the associated 'Embedded Web Server Software Suite' and a Z02915 single-chip modem, we have a mini-webserver with astonishing capabilities.

Among the other current hardware solutions, the Intel SA1110 should be mentioned. With a rich assortment of on-chip peripherals, this IC together with a V.90 modem chipset is a powerful alternative solution.

In the next instalment of this series the focus is on programming a typical mini-webserver, while in the third instalment the mini-webserver will be presented as an *Elektor Electronics* construction project.

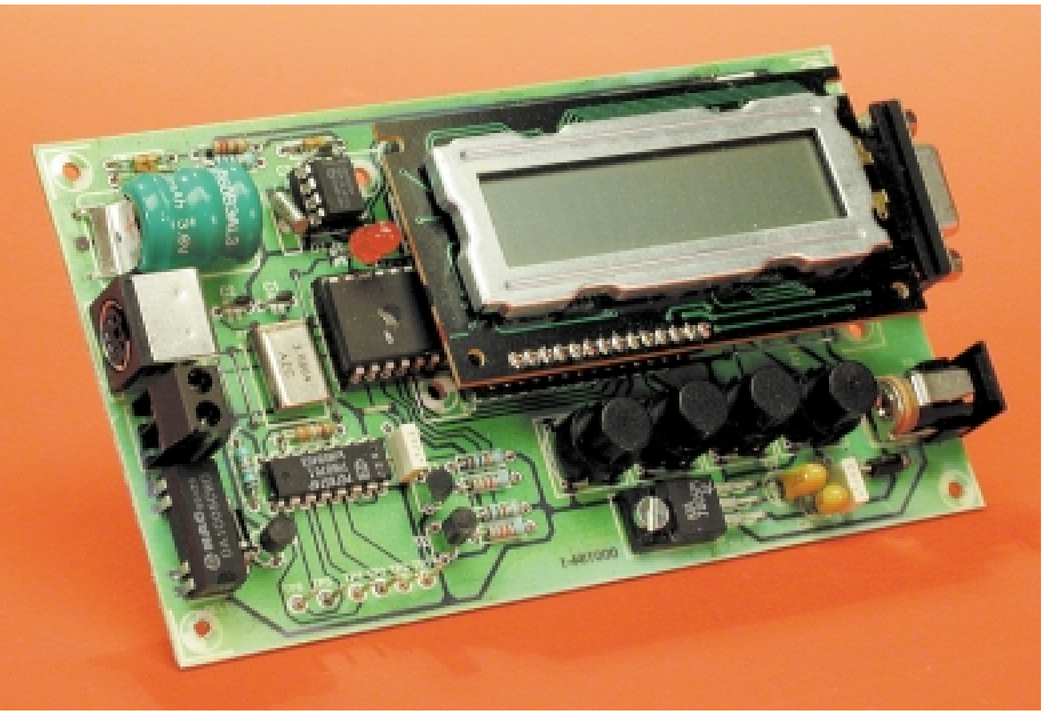
(010046-1)

Major-Domo

a 16-channel microcontroller-driven switching clock

Design by H. Vos

The switching clock described here can control up to 16 channels and has 20 memories for individual switching times, each of which may apply to multiple days of the week. This clock has a very easy to operate, 4-button user interface. Monitoring of the outputs via an RS232 port is also possible. The 'brains' consist of an AT90S8515 from Atmel.



There exists no shortage of switching clocks. Both 'analogue' and 'digital' (with LCD) models are available with widely varying features. So, what are the merits of this design that makes it deserve the light of day? Well, the particular feature that separates this clock from the masses, is that it does not control just one channel, but can be expanded to no fewer than 16 channels. In addition, this clock has a serial interface, which enables every on and off switching event for each channel to be monitored on a PC using, for example, HyperTerminal.

The circuit

An initial quick glance at **Figure 1** makes it clear that a central place has been reserved for the AT90S8515 from Atmel (IC1). The entire program resides within the microcontroller. The frequency of the microcontroller clock is set to 3.6864 MHz, using X1. This frequency permits exact baud rates to be generated for the UART, which is used for the serial communication with a terminal or PC.

The four control buttons S1-S4 for the switching clock (a detailed description of the operation follows later) are directly connected to port B of the controller. R5-R8 provide a defined input level at the inputs of port B when the pushbuttons are not activated. The software takes care of debouncing the pushbuttons.

The display consists of a standard 2 × 16 character LC-display, which is connected directly to PA0-PA7 and PC4, PC5, PC6. Port PA is used as a bi-directional data bus. PC4, PC5 and PC6 are the control signals for the display (RS, WR and E respectively). Potentiometer R11 and resistor R12 form a potential divider that provides a bias voltage for the contrast control of the display. The

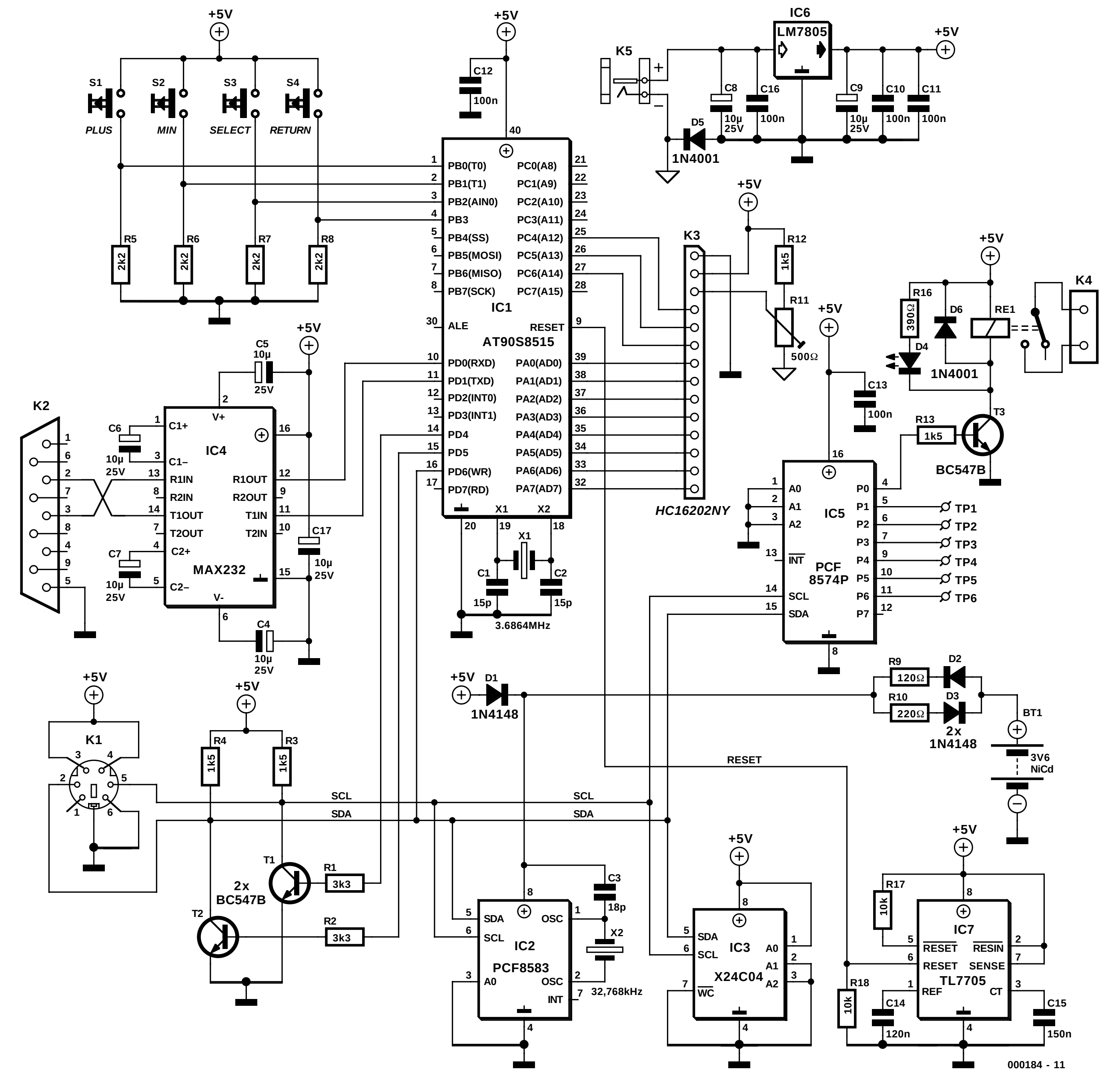


Figure 1. The switching clock is designed around the AT90S8515 microcontroller.

desired contrast is adjusted with potentiometer R11.

An integrated 'supply voltage supervisor' (IC7) from Texas Instruments was chosen to be the reset signal source for the controller. This TL7705 contains a 'power-on-reset' and a 'brown-out' circuit. The IC has a RESET as well as a $\overline{\text{RESET}}$-output, both of which are of the open collector type. R17 and R18 serve as pull-up and pull-down resistors respectively.

Capacitor C15 is the time defining element for the reset pulse. Because the AT90S8515 has a negative reset input, pin 5 of IC7 is used as the reset source.

The communications interface for a PC or terminal is the well-known MAX232 (IC4). Capacitors C4-C7 are the external components that are necessary for the internal DC/DC-converter, which generates the ±10 V power supply.

I^{2}C

The common thread that runs through the entire switching clock is the I^{2}C bus. The I^{2}C bus requires three controller pins (the SDA line is bidirectional). These pins are PD4, PD5 and PD6, where PD4 is used as SCL, and PD5 and PD6 are SDA-send and SDA-receive respectively. Transistors T1 and T2 function as (inverting) buffers for what ultimately becomes the I^{2}C bus. The controller software generates the correct I^{2}C bus logic levels. This implementation does not permit the use of the INT signal that some I^{2}C devices pro-

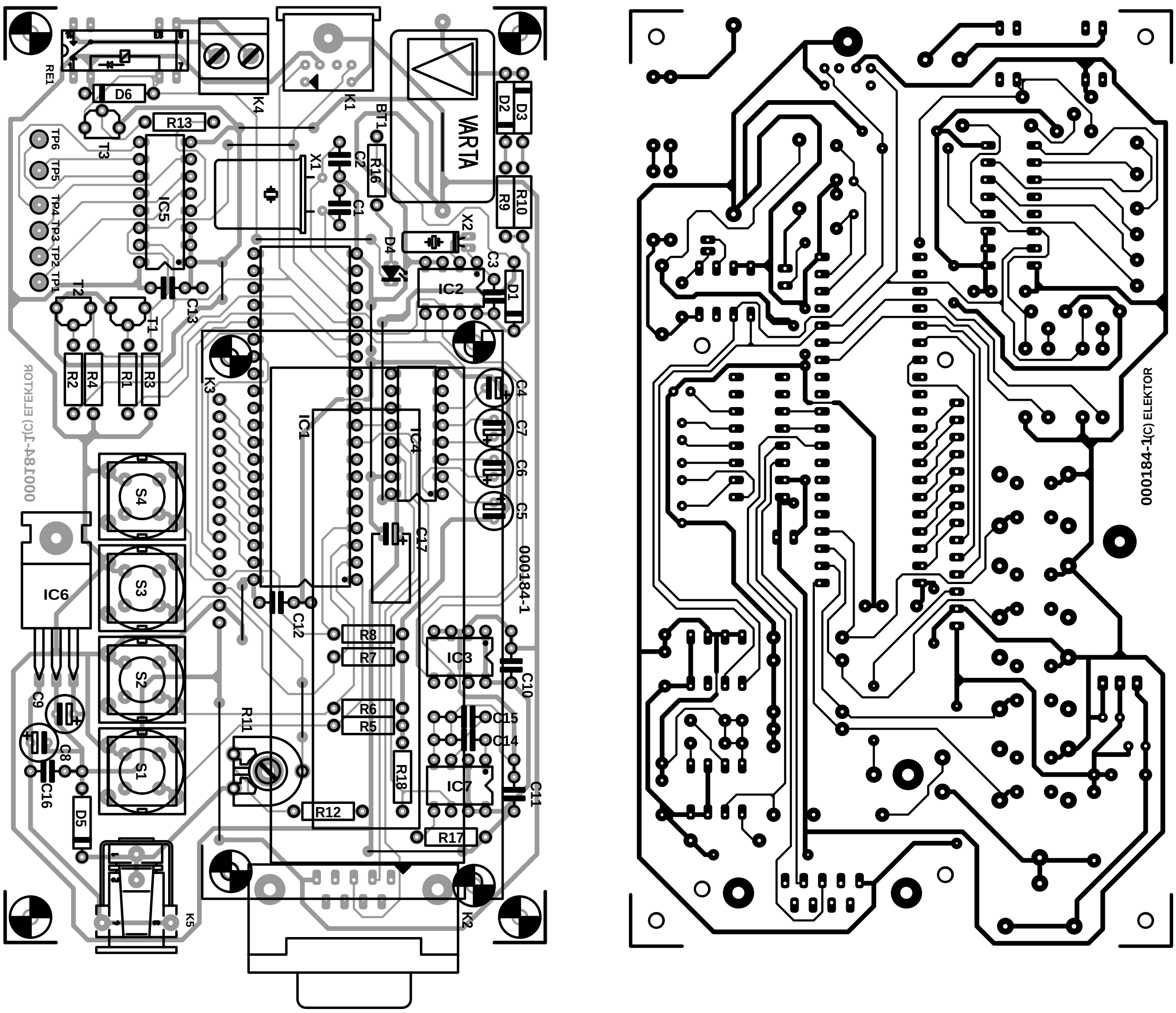


Figure 2. The printed circuit board houses the entire circuit, including connectors, display and pushbuttons.

COMPONENTS LIST

Resistors:
R1,R2 = 3kΩ3
R3,R4,R12,R13 = 1kΩ5
R5-R8 = 2kΩ2
R9 = 120Ω
R10 = 220Ω
R11 = 500Ω preset H
R16 = 390Ω
R17,R18 = 10kΩ

Capacitors:
C1,C2 = 15pF
C3 = 18pF
C4-C9,C17 = 10µF 25V radial
C10-C13,C16 = 100nF
C14 = 120nF
C15 = 150nF

Semiconductors:
D1,D2,D3 = 1N4148
D4 = LED, 3 mm, low current
D5,D6 = 1N4001
T1,T2,T3 = BC547B
IC1 = AT90S8515-8PC, programmed, order code **000184-41**
IC2 = PCF8583P
IC3 = 24C02
IC4 = MAX232
IC5 = PCF8574P
IC6 = 7805
IC7 = TL7705CP

Miscellaneous:
Bt1 = 3.6V NiCd battery with solder tags
X1 = 3.6864MHz quartz crystal
X2 = 32.768kHz quartz crystal
S1-S4 = pushbutton, 1 make contact, e.g., D6-0 (ITT/Schadow)
Re1 = 5V reed relay, DIL case, 1 make contact, e.g., Clare MSS21A05B
K1 = 6-way mini DIN socket, PCB mount
K2 = 9-way sub-D socket (female), angled pins, PCB mount
K3 = 14-way SIL pinheader for connecting LCD module
K4 = 2-way PCB terminal block, lead pitch 5mm
K5 = mains adapter socket, PCB mount
LCD module, 2x16 characters, e.g., Sharp type LM16A211
PCB, order code **000184-1** (see Readers Services page and website)
Disk, contains hex file and source-code: order code **000184-11**

vide. Also, no other masters may be connected to this I²C-bus. The controller software supports single-master operation only!

The I²C bus is available externally through connector K1. This allows additional switching modules to be connected. The real-time clock (RTC) functionality is provided by a PCF8583P (IC2). This IC contains all the circuitry for a complete clock and calendar. X2 and C3 are required for the correct functioning of the oscillator.

While mains power is switched on, the RTC is powered via D1. Simultaneously, NiCd battery Bt1 is then charged via D3 and R10. During periods when the mains is not available, Bt1 provides the power supply to the RTC via D2 and R9. This ensures that the clock continues to run and that the date and time are properly maintained. Diode D1 prevents the remainder of the circuitry from being inadvertently powered by the NiCd battery.

IC3 is a 256-byte EEPROM, which is used to store the switching times. IC5 is a switching unit whose address lines A0, A1 and A2 are connected to ground, which causes this unit to have the lowest address. LED D4 indicates the switch state of Relay Re1. The relay contact is closed when D4 is on. The contact itself may be used freely.

Software

The software for the switching clock is written entirely in AVR Studio, which can be obtained from the Atmel website. The software consists of seven modules: SKLOK.ASM, DATADEF.ASM, DIVFUNC.ASM, I2CFUNC.ASM, MENU.ASM, 8583SET.ASM and SWKLOK.ASM. The latter is the main module. This has to be indicated as such in the project-manager of AVR-Studio. The source code has been liberally commented and has been kept as simple as possible. It is, however, impossible to discuss the details of the software in its entirety.

On start up, the list of switching times is read from the EEPROM. Every switching event consists of five bytes (hours, minutes, days, status/channel, 2's complement of the previous four bytes). When the fifth byte is correct it is assumed the previous bytes contain valid data. Reading from the EEPROM continues until an invalid 2's complement value is retrieved. All these values are copied to an array and stored in the SRAM of the controller. (To reduce the I²C activity on the bus.) The main loop of the software compares the times in the array with the time in the RTC. If one or more array entries match, then the corresponding I/O module is set to the desired state. These status changes are also passed on to the RS232 port of the microcontroller. Whenever a change is made to the switching time, both the array, as well as the contents of the EEPROM, are updated.

Finally, a brief description of the functionality of each module:

SKLOK.ASM is the main module. This details all the interrupt vectors, program variables and data definitions. It also contains the main loop of the program.
DATADEF.ASM contains all the text strings, which may be shown on the display. These strings may be translated to another language. The length of the text must remain exactly the same (adding extra spaces if necessary), and every string has to be terminated with a '00'. The software uses this to test for 'end of string'.
DIVFUNC.ASM contains the routines that are called from various modules.
The file **I2CFUNC.ASM** contains all the routines that are required for controlling the I²C-bus. This software does not support the multi-master protocol. The microcontroller is in this case the only master. This version of software does not support interrupts either. The software is, however, perfectly adequate for most applications.
The file **MENU.ASM** is self- explanatory. This module is called from the main module and it in turn calls the underlying program modules. The design is such that is easy to add menu options.
8583SET.ASM is one of the modules that is called from within the menu. This controls all the programming of the RTC. For details refer to the data sheet of the PCF8583.
SWKLOK.ASM is also called from the menu. This module takes care of the initial entering or changing of the switching times.

Practical matters

An ordinary 9 V mains adapter connected to K5 will suffice to power the switching clock. There are no particular requirements since voltage regulator IC6 will provide a stabilised 5 V.

Figure 2 shows the printed circuit board that has been designed for the switching clock. Although the circuit contains rather a large number of ICs, we have been able to keep the size of the PCB down, as you can see. There are no difficult details associated with the construction of this circuit. If you build it according to the parts list everything will, in all likelihood, go smoothly. Because the various connectors and four pushbuttons are fitted directly on the PCB there is almost nothing required externally.

As already indicated on the schematic, the PCB contains only a single relay stage (T3, Re1). Of course, the other outputs of IC5 (TP1-TP7) can each be connected to an external relay stage. For details regarding the operation of the I²C-bus and the various components you are referred to the 'I²C Peripherals' data book from Philips (order number 9397 750 00306). When connecting I/O modules a considerable distance away from the switching clock, it is necessary to buffer the I²C bus using an 82B715 I²C Bus Booster (refer to the June 1994 issue of *Elektor Electronics*, for example).

The programmed controller, the printed circuit board as well as a diskette with both the hex file and source code may be purchased from *Elektor Electronics*. The latter two may also be downloaded from our web site at www.elektor-electronics.co.uk.

Operation

The four pushbuttons S1-S4 have the following functions: Return, Select, Down and Up. After switching the clock on, the display always indicates the time and date which is read from the RTC:

```
Time: 14:44:59
We 11 Oct 2000
```

By pushing the Select button, the switching clock menu becomes available. This menu offers the following choices:

```
- 'RTC-setting'
```

- 'Switch clock'
- 'Module list'
- 'Manual'

Using the Up- and Down-buttons one of the four options can be selected. The chosen option is entered by pressing the Select button.

Setting the RTC

After selecting the RTC setting, you have to successively enter the hours (24-hour clock), minutes, month, day of the month, day of the week, and year.

Use the Up and Down buttons to set the correct value. Pushing the Select button takes you to the next field. Once all the values have been entered you are returned to the menu.

Switching Times

The clock can store up to 20 different switching times. When adjusting the switching times, the number of the event (0-19) is displayed in the top right hand corner.

Shown below 'Time:' is the actual time when the switching action will occur. Following that, indicated with asterisks, is/are the day(s) for which this action is planned (MTWTFSS = Monday, Tuesday through Sunday). By pressing the +- and –-buttons you can choose to switch on a particular day, every day of the week, working days only or weekend only. An underscore ('_') indicates that the switching event is inactive on that particular day.

Once the correct days have been selected we have to choose which output (Channel) has to be switched and whether it has to be turned on or off. Channel 0 means relay Re1, Channels 1-7 are found on P1-P7 of IC5. Channels 8-15 are only available if an additional PCF8574P is connected to the I^2C bus.

Module list

The main intention for this menu option is to display which (switching) modules are present on the I^2C bus. The display may indicate the following:

```
8574     *——
8574A    ———
```

This may come in useful to check if all the modules are properly identified or whether there is a problem with the I^2C bus, for example.

Manual

This makes it possible to switch a channel on or off manually, bypassing the timer.

The display indicates the same thing as the switching clock channel/status setting. An asterisk indicates whether the module is present or not.

Serial connection

The switching clock can be connected to a PC using a standard 9-way cable; a terminal program such as HyperTerminal in Windows may then be used to keep a log of switching activity. Whenever an output changes state, the clock will send a line of text to the PC, indicating the time of day, name of the day (2 characters), the date, channel number and the new state of the channel. The serial port on the PC has to be set as follows: 9600 baud, 8 bits, no parity, 1 stop bit, no flow control. (000184-1)

Online Component Shopping

track down those elusive parts

By Martin Cooke

Two years ago we published an article lamenting the passing of many of the smaller suppliers of components and noting the rise of Internet shopping. Some companies have since folded and others have sprung up to serve the needs of the intrepid electronics enthusiast. We thought it timely to pen an update on the current situation.

RS components…Not just for trade customers.

How often have you been put off building a project when you discover that it specifies really exotic components that no supplier has heard of?

It used to be that finding data sheets were a problem not only for the hobbyist but also sometimes for professional engineers. With the advent of the Internet most component manufacturers make this information freely available on their web-sites and there are now many independent sites dedicated to supplying technical data sheets, all the information you need is just a download away.

Tracking down and ordering components using the Internet should also be much speedier and more convenient than using traditional methods. It seems however that many people are cautious about giving out their credit/debit card details on-line for fear that they will fall into unscrupulous hands. Figures for the year 2000 suggest that only a tiny proportion of credit card fraud occurred over the net. Remarkably, your card details are far more likely to be hi-jacked in the high street than intercepted on the internet.

If anything does go wrong with an Internet transaction you are protected by UK criminal and civil laws if the company is UK based.

Methods of payment

The most popular method of payment for purchases over the Internet

Maplin…The most prolific high street component retailer.

is with a credit/debit card. Credit card companies stipulate a minimum age of 18 years whereas debit cards can be issued at 16 if you fulfil the banks criteria (no dosh, no card). Recently some banks have introduced credit cards to encourage customers to make more use of the net by offering cashback of 2% on net transactions from its Internet shopping partners. Another innovation to increase consumer confidence is the one-use credit card number — for each purchase you use a credit card number that is only valid for one transaction.

E-wallets help to make shopping more convenient by keeping your credit card information in a secure area of your browser and transferring the necessary card information (encrypted) at the moment of purchase.

Most overseas web traders accept the major credit cards and in this case each transaction is shown on your monthly statement along with its conversion into sterling.

Overseas customers living outside the EU zone and ordering goods from companies inside the EU will of course not expect to pay VAT but must pay an import duty (this varies from country to country) and additional postage charges will be incurred.

Companies online

An increasing number of component retailers are turning to the Internet.

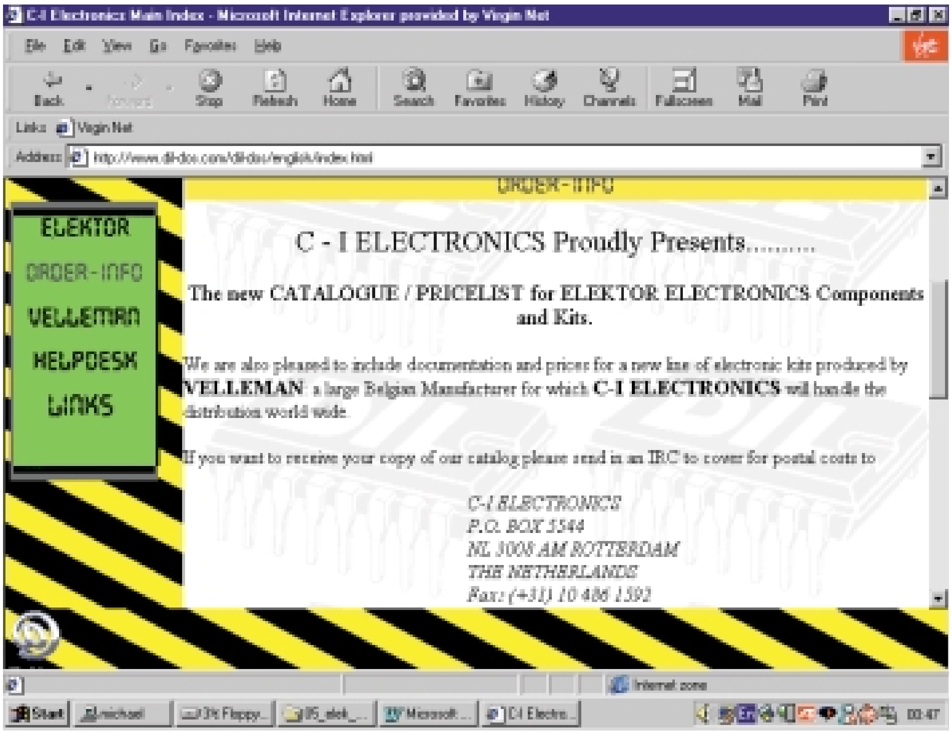
Among them is **Maplin Electronics**. Maplin has been in existence now for over 25 years has almost 60 shops throughout the UK and Ireland and an on-line catalogue (www.maplin.co.uk). It stocks around 20,000 products and offers on-line product availability check and same day despatch. Orders valued at less than £30 incur a small handling charge — otherwise postage is free.

RS Components have always been suppliers to the trade but have been catering for the needs of individuals through its **Electromail** branch since 1985. Their website is at www.rswww.com.

Leeds-based **Farnell** carries an enormous stock of products and has an impressive web site with on-line shopping and a 24hr manned order hot line. It supplies to the trade and also to individuals. There is a minimum order value of £10.00 on credit card orders and delivery is free. They can be found at www.farnell.com.

If you are specifically searching for parts for *Elektor Electronics* projects then a good source is **C-I Electronics** based in Rotterdam,

Farnell….An impressive website and excellent service.



C-I Electronics…Have been specialising in supplying Elektor Electronic kits and parts for over ten years.

The Netherlands. This company specialises in supplying complete kits and individual items. Although C-I Elecrtronics does not trade on the net, its website lists kits of parts for *Elektor Electronics* projects and Velleman kits. Prices are quoted in Euros so with the current favourable exchange rate to UK sterling the prices represent good value. This company carries a vast stock of components and accepts the main credit cards. Orders are made on a downloaded order form and sent conventionally via snail-mail. Enquiries should be made by fax or e-mail. More information is available from their website at www.dil-dos.com.

Viewcom Electronics are another firm which specialises in the supply of kits, PCB's and parts for *Elektor Electronics* projects along with a broad spectrum of components supplying to both the trade and individuals. This company does not presently have a website but can be reached via email at sales@viewcom.force9.co.uk.
All major credit/debit cards are accepted.

Sky Electronics (formerly Cricklewood Electronics Ltd) offer a wide range of products and supplies to the trade and individuals. Online ordering is not possible at the time of writing. They can be reached at www.sky-electronics.co.uk

Greenweld Limited supply good value components and kits along with surplus stock items. Online ordering and catalogue browsing can be found at www.greenweld.co.uk.

Electrovalue Ltd stocks an extensive range of general components and in particular wound products, pot cores, ferrites etc they stock EPCOS (formally Siemens) products and supply to both the trade and individuals, a minimum order charge of £5.00 is only applicable if you pay by credit card. On-line ordering is currently not possible. Find them at www.electrovalue.co.uk.

Distel are a company specialising in the supply of surplus stock. It has loads of interesting products on its web site at www.distel.co.uk so if your thinking of building a robot to bash the brains out of hypno-disc in next season's Robot Wars then this may be a good place to start.

Eurodis is a large European component distributor and has a UK presence through **HB Electronics** in Bolton. They are geared to supplying to the trade and have a minimum order value of £150.00. Their website is at www.eurodis.com.

Spoerle is also a large European distributor and is now part of the Arrow group. Branches are found in the UK and most European countries. They carry a wide range of products and do not have any minimum order value restriction. The company does not sell on-line at present but you can reach their website at www.spoerle.com.

Conrad Electronics is based in Germany and has a multilingual web site with prices in Euros, it offers full catalogue and on-line ordering and a handy Euro to sterling converter on its website at www.conrad-electronic.com.

Happy hunting

On balance, the Internet is a good facility for the electronics enthusiast but it pays to be cautious if you are thinking of ordering from a new supplier. Remember the golden rules:

– Always look for the company address, country and telephone information on the website.
– Ensure that a secure server is used before you give details of your credit card. (look for the key icon in Netscape Navigator or the little yellow padlock when using Microsoft Explorer).
– Check the costs of postage and packing (p&p), and any minimum order charge.
– Search newsgroups (www.deja.com/usenet) or sites such as ePinions (www.epinions.com) to check for any bad user feedback connected with the company.
– Check your monthly statement.

If you are still unsure it is worth giving the company a ring. A five-minute call can reveal far more about an organisation than any elegantly crafted web page.

Not least, PCB's, pre-programmed controllers and software for *Elektor Electronics* projects, along with technical books and much more can be ordered on-line from our own website at www.elektor-electronics.co.uk.

(000146-1)

Fuzzy Logic (2)

Part 2 (final): Fuzzy Design

By Owen Bishop

OandA.Bishop@bigpond.com

There are a number of programs written for PCs and other computers that allow the user to design and debug fuzzy systems. **Figure 1** shows a screen produced by *fuzzyTECH*™, which is published by Inform. This is the software we used to design the dishwasher controller described in Part 1. We will look at its operation stage by stage. The rearmost window shows the Project Editor, which displays the sections of the controller. From left to right, these are the Input (Load), the Rule Block, and the Output (Washtime).

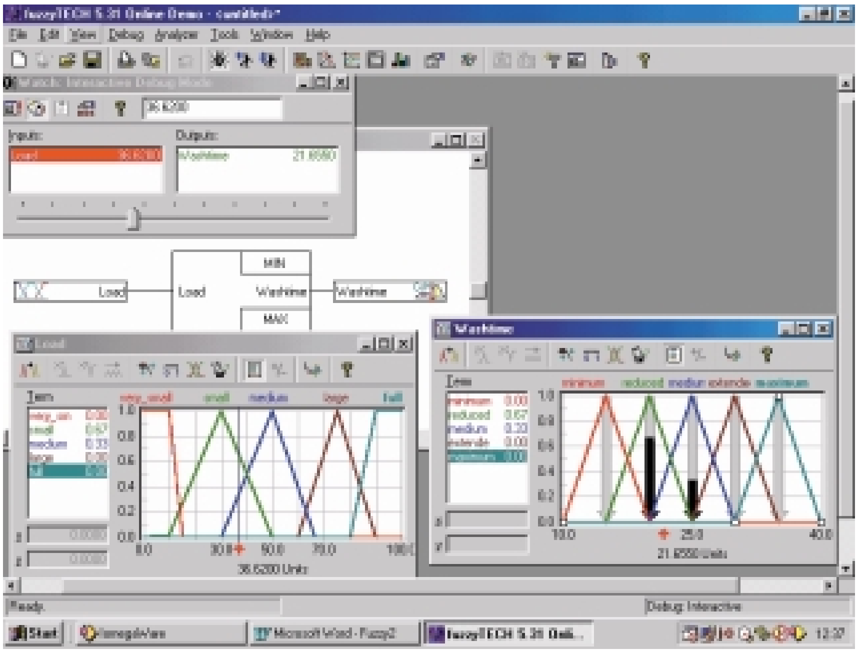


Figure 1. A fuzzy controller for a dishwasher being designed with fuzzyTECH‘.

Crisp input

The input to the controller is set by a slider in a Debug window at top left. The input panel at lower left shows the input membership functions of the dishwasher as depicted in Figure 3 of Part 1. The load input has been set to 36.62 units. The input panel shows a vertical line at 36.62, which cuts the functions for ‘small’ and ‘medium’ at 0.67 and 0.33 respectively. Rules 2 and 3 are fired:

```
2  IF load is ‘small’,
   THEN washtime is ‘reduced’.
3  IF load is ‘medium,
   THEN washtime is ‘medium’.
```

Crisp output

Output is calculated using a centre of area method and is displayed in the output window at bottom right. The ‘reduced’ and ‘medium’ output membership functions are being activated with degrees of applicability 0.67 and 0.33 respectively, as shown by the broad black arrows. The resulting crisp output value is indicated by the arrow at the bottom of the panel. As can be seen on the right of the debug panel, the crisp output is 21.655 minutes. Summing up, a load of 36.62 is to be washed for 21.655 minutes.

The slider on the debug panel can be set to any load value in the range 0 to 100, and the corresponding washtime is immediately displayed as the output. The displays in the input and output windows are updated at the same time. This is an extremely simple example that has only five input and output variables, and only five rules but it illustrates the essential features of fuzzy controllers.

Two inputs

To exploit the abilities of *fuzzyTECH*™ more fully, though by no means demonstrating all of its many and varied functions, we provide the washing machine controller with a second input variable, *dirt*. This could be measured (again on an arbitrary scale) by a photometric technique that measures the cloudiness of the washing water after the machine has been running for a fixed period, such as one minute. Let this be expressed on the scale 0 to 10, and suppose that there are seven linguistic variables for dirt. These are ‘almost clean’, ‘slight’, ‘low’, ‘medium’, ‘medium-high’, ‘high’, ‘very high’. The membership functions are shown in the panel at lower left in **Figure 2**.
The screendump also portrays the load input window (top right), the output window (bottom right) and

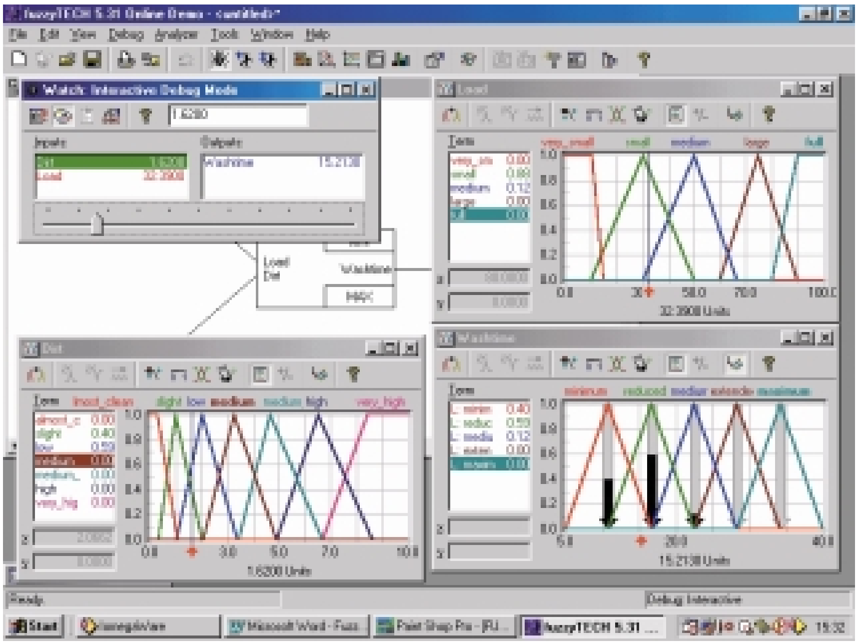


Figure 2. In this simulation, the washtime of the dishwasher depends both on the load and on the amount of dirt on the dishes.

debug window (top left).

As before, the computer generates a rule block, which contains all possible rules for connecting two input variables (at 5 and 7 levels) with one output variable at five levels. There are 175 such rules of the form:

```
IF <load variable>
   AND <dirt variable>
   THEN <washtime variable>
```

The user deletes unwanted rules and simplifies some of them. For example, if the dirt level is 'very high', it is advisable to wash for 'maximum time', whatever the load size. are 'almost clean. Eventually we reduced the list to 31. In general terms, these gave increased washtime for larger loads and greater dirt. In the figure, load is set to 32.3900 and dirt to 1.6200. The load window shows that the load is in the 'small' and 'medium fuzzy sets, with indices 0.88 and 1.2 respectively. Dirt is in the 'slight' and 'low' sets with indices 0.40 and 0.59. Four of the 31 rules are fired:

```
9  IF load is 'small'
   AND dirt is 'slight'
   THEN washtime is 'minimum'.
10 IF load is 'small'
   AND dirt is 'low',
   THEN washtime is 'reduced'.
15 IF load is 'medium'
   AND dirt is 'slight',
   THEN washtime is 'reduced'.
13 IF load is 'medium'
   AND dirt is 'low',
   THEN washtime is 'medium'.
```

Three output variables are involved in the fired rules. The output panel at the bottom right of Figure 2 shows that output consists of 'minimum' (0.40), 'reduced' (0.59) and 'medium' (0.12). Defuzzification, gives a crisp output of 15.2130 minutes of washtime.

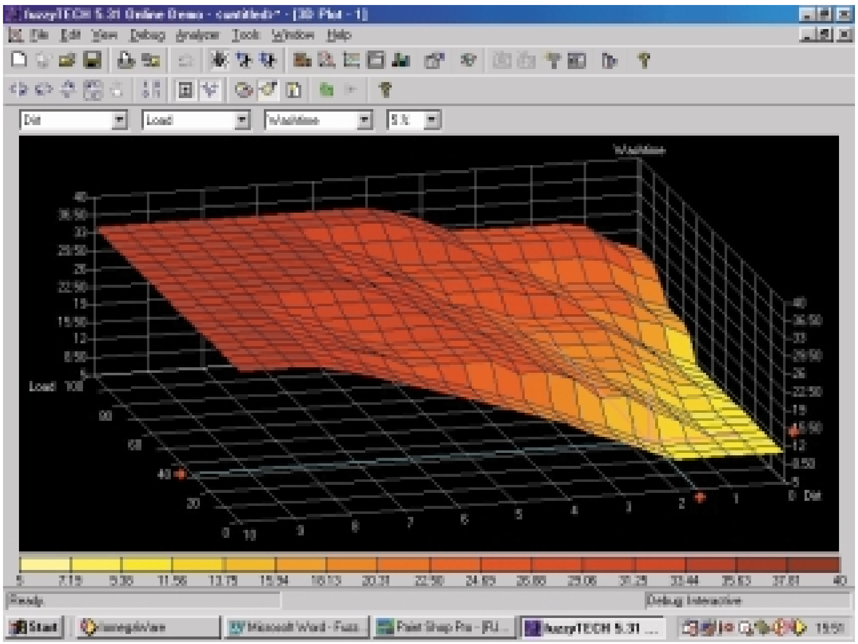


Figure 3 The response surface of the dishwasher controller shows a suitably smooth relationship between input and output.

3-D Plot

The relationship between the variables can be plotted in 3-D, as in **Figure 3**. The overall response of the system can be seen as a reasonable one, washtime increasing with both load and dirt. The rule block may be set up using expert knowledge, although in this case it was a matter of the author's 'common sense'. The 3-D display allows the user to inspect the relationships visually. In this example, it is important to ensure that the response is a smooth one. Other applications might require gaps and discontinuities.

Crisp and fuzzy logic

The rather primitive dishwasher controller described above uses 31 fuzzy rules to define its behaviour. To achieve the same degree of smooth control with crisp logic rules requires possibly hundreds of rules. As an illustration of the problems, consider the behaviour of the washer at one particular load and dirt level. The rule would require crisp inputs and would provide a crisp output. A typical rule might read:

```
IF load is above 40
   AND dirt is above 1.5,
   THEN washtime is 25.
```

This rule is not fired if the load is 39 and the dirt is 1.6. It is not fired if the load is 41 and the dirt is 1.49. Yet, in either case, it is obvious that the washtime should be 25, as before. Since there is no known underlying mathematical relationship that connects load, dirt and washtime, we are forced to compose reams of binary rules to cover the situation. This is the result of trying to force binary rules to a case which is not binary in its nature. In practice, most of life is fuzzy and crisp rules are inappropriate.

The advent of fuzzy logic does not mean that crisp logic has been supplanted. There are many instances in fields such as flight control, chemical processing, and boiler control where conventional control systems are satisfactorily employed. Usually these are processes where the basic mathematical relationships are known exactly or closely enough. Then we can implement the traditional techniques of proportional control (P), usually with the refinements of integral (I) and differential (D) control. However, even in these fields, fuzzy systems are beginning to challenge tradition. One of the elements in a boiler control system is the flame that heats the water. This is far from binary in nature. On the contrary, it is extremely fuzzy, so perhaps fuzzy logic could better control the boiler than the traditional systems.

It is possible to implement systems similar to the conventional P, P+I, P+D and P+I+D systems using fuzzy methods. The crisp error signal (the difference between the output and the set point of the system) can be fuzzified into, say, five fuzzy sets for use in proportional control:

'large negative', 'small negative', 'nil', 'small positive', 'large positive'.

We can similarly fuzzify the rate of change of the error signal for P+D (differential) control systems.

Applications

Fuzzy logic took a long time to gain acceptance among theoreticians and engineers of the west, especially in USA. In fact, there was considerable active opposition to the ideas. It did much better in the east, notably in Japan. There were soon a large number of practical applications that really worked, were commercially successful, and proved the advantages of the method.

Fuzzy logic is used today in a wide range of consumer goods. Washing machines using fuzzy control have been marketed by numerous companies, including Hitachi, Matsushita, Daewoo, Goldstar, Samsung, Sanyo and Sharp. These machines adjust the washing program according to the size and dirtiness of the load, the type of fabric, and the water level. They control the amount of water used, its flow rate and the wash time. At later stages in the cycle, they control rinsing and spin-drying. It is claimed that washing machines with fuzzy control complete the wash more quickly, with significant savings of electric power.

Fuzzy controllers are successfully employed in other items of domestic electrical equipment. Applications include TV sets, microwave ovens, fan heaters and air conditioners. In general, the result is superior performance coupled with power economy. An interesting application has been found in camcorders. One of the problems with these is that a hand-held camera is very much subject to camera shake. A camcorder from Matsushita eliminates this in an ingenious way. Each frame of the recording is compared with the previous one. In normal recording some **but not all** of the parts of the image will be expected to move. If the fuzzy logic detects that all parts of the image have moved about the same distance in the same direction, this is evidence of camera shake. Accordingly, the new image is displaced to compensate for the movement and the effects of shake are considerably reduced.

Fuzzy logic has been successful on the larger scale too. A notable example is the subway system in the Japanese city of Sendai. In 1987, this was one of the first major examples of fuzzy logic control. The system controls many aspects of the subway, including the acceleration, velocity and deceleration of the trains, route control, public address systems and ticket sales. The result has been improved safety, reduced power consumption and reduced travel times.

The success of fuzzy logic is such that its use is rapidly increasing. Today's hardware, crisp on the outside, is more than likely to have a soft, fuzzy centre.

Software

At the time of writing, more detail may be had from Inform, Pascalstrasse 23, D-51000, Aachen, Germany. Their web site at http://www.fuzzytech.com offers a free download of the restricted version of *fuzzyTECH*™ that was used in the preparation of this article.

(010026-2)

DMX512 Revealed

we tell all…

By B. Bouchez

Since the very beginning, *Elektor Electronics* has tried to help its readers profit from new technologies by means of original and professional circuits. In that tradition, we are presenting a series of circuits (DMX to voltage and MIDI to DMX) that are specifically intended to be used to control lighting systems for theatres, discos and outdoor events — all based on the DMX512 system.

The DMX512 system, which is already well known to those who are professionally involved in theatre lighting, is beginning to find widespread use. The objective of this article is to take a look behind the scenes with regard to the various technical aspects of this connection protocol, after first briefly reviewing the history of remote control techniques for lighting systems.

Once upon a time…

Starting with the appearance of the first theatres in ancient times, lighting has played an important role. In Roman and Greek ruins, it can be clearly seen that even then, architects took the ‘technical’ aspects into consideration. However, it is not possible to determine with certainty whether the emplacements for the ancient light sources served only practical ends (to allow performances to be given after dark) or

were used to create effects on the stage. We should not forget, though, that the Greeks and Romans had considerable experience with special effects, which would tend to support the second supposition.
Once theatre performances started to take place increasingly often in enclosed spaces, the need for lighting became clearly evident, if only to fix the attention of the audience on the stage.
The first remotely controllable lighting sources appeared in the middle of the 19th century. The lamps of that era used natural gas. The term 'light organ' originates from that time and comes from the appearance of the control panel, which with its multitude of pipes and valves resembled the front of a pipe organ.
Gas lighting was a far from ideal solution, and it was replaced by electric lighting as soon as this was made possible by the technology of incandescent lamps and carbon-arc lamps.
The first electrical control panels consisted simply of a set of autotransformers connected to a number of incandescent lamps (it is not possible to control carbon-arc lamps electrically). These installations were large, heavy and rather dangerous, since the majority of the 'live' elements were exposed to touch, as was common with control panels at that time.
This system, which came into being in the 1920s, was used exclusively until the 1960s, when the first thyristors appeared.
Thanks to these semiconductor devices, the weights and dimensions of control panels could be significantly reduced. An additional factor was that all that was needed to remotely control the thyristors was a simple voltage, so it was possible to devise intricate switching circuits that could be used to create effects that had previously been impossible, such as the simultaneous chained dimming of a multitude of lights (just imagine trying to operate several mechanical regulators in unison!).
Before continuing with the history of remote control, let's take a closer look at these thyristor controllers, which technicians call 'dimmers' or 'power boxes'. They dominated the scene from the 1960s until the mid-1980s. Thryistors work on the principle of phase control, and they create audible disturbances (the lamps 'sing') as well as a large amount of electromagnetic interference, which can create disturbances in sound systems. In order to curb these disturbances, thyristor dimmers are fitted with chokes, which increase the weight of the circuitry and reduce the speed with which the lights can be brought to full power.
Since the middle of the 1980s, transistors that can work at mains voltages have been available. This made it possible to produce high-frequency dimmers. The chokes could thus be made considerably smaller and thus lighter (if they were not simply eliminated). An additional benefit is that there is almost no interference with the sound system (although the same cannot be said with regard to electromagnetic interference!).

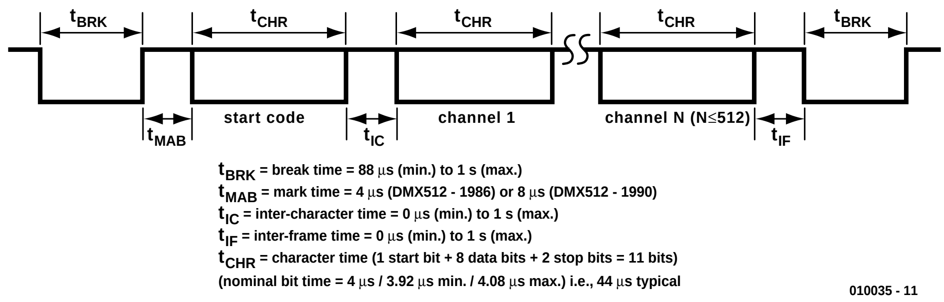


Figure 1. Structure of a data block.

From remote control to multiplexing

In addition to the other advantages that came with the use of power semiconductors, it was possible to operate the electronic controllers remotely. Installations that were fitted with these semiconductor controllers could be operated remotely using voltages or currents generated by a control panel made up of a number of potentiometers.
This type of remote control gave rise to a number of concepts that are still used, such as 'scene presets' (a set of lighting commands sent to the stage as a group), 'fades' (progressive transitions from one preset to a different preset), 'master' (the simultaneous control of the levels of a group of presets using a single potentiometer), 'blackout' (fading a complete preset to the null level) and so on.
It should be noted that even today, the basic ideas of stage lighting are still based on the concepts that were established at the time of the first remote control panels for lighting systems. In the catalogues of manufacturers of stage equipment, you will still find fully analogue control panels at the bottom end of the price range.
One of the principal shortcomings of analogue remote control using a voltage (or current) is the need for a separate conductor for each control channel. With a portable installation, such as is used for tours, re-attaching all the connectors for a large number of lights can turn into a real nightmare. In addition, repeated plugging and unplugging of connectors does not improve their reliability.
Beginning in the early 1990s, many developers of lighting systems turned their attention to

Table 1. Earlier communications protocols

Protocol	Developed by	Remarks
ADB62.5	ADB (Belgium)	Comparable to DMX512, but works at 62.5 kb/s. No longer used in new commercial products.
AVAB	AVAB (Sweden)	Widely used in Northern Europe.
CMX	Colortran (USA)	Was the basis for DMX512. Still frequently used in the USA.
Micro2	LMI (USA)	Secure protocol, control by means of returned data.
PMX	Clay-Paky (Italy)	Slow protocol (9600 baud); compatible with the PC serial port. Still used in Clay-Paky and Pulsar products.

this problem. Several different solutions were proposed to reduce the number of connections between the dimmer racks and the control panel. One manufacturer, Strand Lighting, developed a protocol based on multiplexing and gave the name 'CD80'. With this system, several tens of commands could be transferred via a standard microphone cable. Before this protocol could be fully developed, a standard that was largely inspired by it was developed by the USITT (United States Institute for Theater Technology). This standard defines an analogue multiplex protocol, called AMX912, that is based on two pairs of leads. One of these carries a variable voltage ranging between 0 V and 5 V (which represents values from 0% to 100%), while the other one carries a digital clock signal (0 V = '0', > 4 V = '1').
The synchronisation of the dimmers with the transmitter (the control panel) is realised by sending one high-level clock signal with a longer duration than the rest. Each receiver copies the value of the analogue signal while the clock signal is high, usually be means of a sample and hold circuit. The high level of the clock signal must have a duration of at least 50 µs. AMX (which stands for 'Analogue MultipleX) owes its name to the fact that the maximum number of channels that can be controlled before the receivers must again be synchronised is 192.
The USITT recommended the use of a balanced transmission line for the clock signal, in order to allow large distances to be bridged. This is a bit strange, since this recommendation did not apply to the analogue signal pair, with the result that AMX192 is sensitive to ground loops and noise. This standard was only used privately by a few manufacturers and was never truly successful. On top of this, the analogue protocols of other manufacturers profited from their commercial strengths and spread more quickly. Some of these protocols, such as S20 (ADB) and D54 (Strand Lighting), are still in use.

Digital is coming!

In the middle of the 1980s, several manufacturers of lighting equipment, faced with the demands of professionals and being convinced of the future prospects of the microcontroller, decided to integrate this new device in their equipment. As a result, a number of communication protocols appeared, each with its own advantages and disadvantages. You should also realise that these protocols were completely incompatible with each other. Nevertheless, some of them are still used at the present time. **Table 1** presents a brief summary of these protocols.

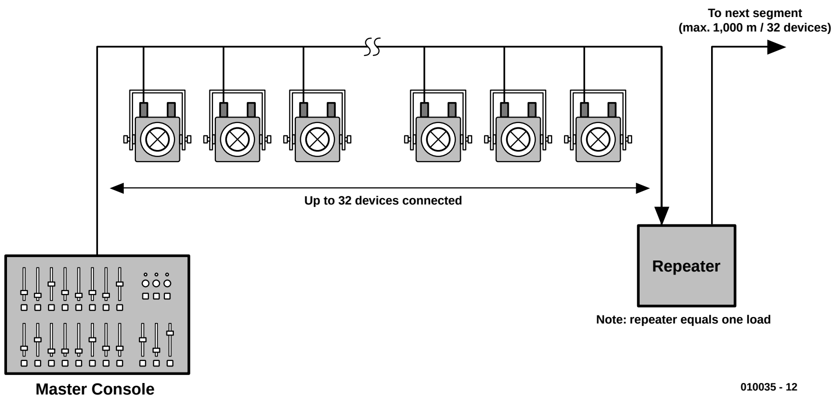


Figure 2. The electrical limitations of RS-485.

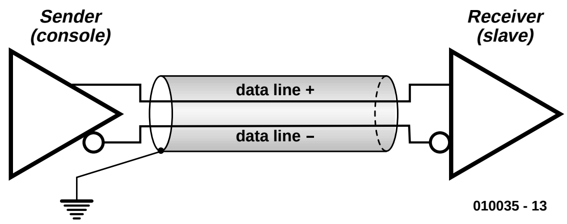


Figure 3. Principle diagram of an RS-485 connection.

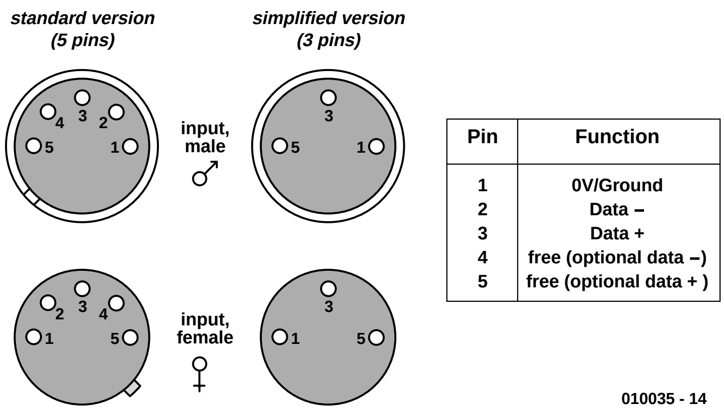


Figure 4. XLR connector pinout.

And then came DMX512

At the beginning of 1986, the USITT and certain manufacturers held a meeting in order to specify a fully open digital protocol.
In order to enable existing microcontrollers to be used, they settled on an asynchronous serial connection with 8 data bits, 2 stop bits and no parity, with a data rate of 250-kbaud.
The control principle is simple: bytes that contain the values for the associated channels are sent successively via the serial link. The value 00h is equivalent to 0% and the value FFh is equivalent to 100%. You must take into account that in the DMX standard, there is no fixed relationship between the received DMX value and light intensity (or movements, such as the orientation of a mirror or the rotational speed of a gobo).
After all the values have been sent once, the cycle starts again from the beginning. It must be noted that the reliability of DMX is simply based on

the continuous repetition of this cycle, with essentially no check being made to verify the correct reception of the values (this is called a 'simplex' link). Due to the absence of a return signal, DMX is not suitable for controlling pyrotechnic devices or heavy mechanisms (such as hydraulic lift platforms), for reasons of safety.

In the DMX standard, there is no value specified for the minimum number of channels that can be controlled. This number can lie between 1 and 512 (hence the name DMX512). Clearly, the fewer channels there are to be controlled, the shorter is the cycle time (and the faster the reaction time of the controlled equipment). A number of 'master' DMX devices, such as the interface that will be described later on, also allow the number of controlled channels to be changed for each new cycle.

The first byte of each series of bytes sent via the DMX line is not responsible for controlling the intensity of a lamp. This byte is called the 'Start Code', and it is intended to be used for possible extensions. Presently, the only valid standard value for this byte is 00h. Normally speaking, every 'standard' DMX device is supposed to ignore a frame in which the Start Code is not equal to 0. However, some manufacturers assign a non-standard value to the Start Code in order to cause their equipment to execute special programs (tests, resets, demos etc.). Take our advice: if you want to avoid panic situations during a performance, refrain from 'playing around' with the Start Code, especially when using equipment from several different manufacturers (we speak from experience!).

Now you know how data for the various channels is sent out. The problems that the receiver is faced with are how to determine where the command series starts (and ends), how to recognise the Start Code and how to count the number of bytes that have been sent in order to identify its associated channel. Synchronisation is achieved by forcing the serial link into a state that it can never attain in actual operation, regardless of what values are sent. This special state is called Break, and it is achieved by holding the link at the '0' level for the duration of at least two frames (88 µs). Given the fact that at least two stop bits are located between every pair of bytes that are sent, it is impossible for such a situation to be reproduced during a normal transmission, even if two successive frames have the value 00h (eight bits with the value of '0').

When the receivers in the communications link recognise a 'gap' generated by Break, they are synchronised and thus know that the following byte is the first byte of a new data block. According to the DMX standard, the Break time can have any value between 88 µs and 1 s. In principle, after the expiry of a one-second interval the receivers can assume that the link has been broken, and they must then switch over to a predefined state (which is usually 'everything off').

The problem that we now face with regard to Break relates to the signal that announces the start of a new byte to the receiver, and which is called the start bit. This bit has a value of '0', since the value of the line when it is at rest (the 'idle state') is '1'. In order to allow the start bit to be properly recognised, the serial link must remain 'high' for a certain time following the Break. It is the job of 'Mark after Break' (MaB), which comes after the Break time, to ensure that this happens.

In the first version of the standard, published in 1986, the MaB time was specified to lie between 4 and 8 µs, but experience proved that this time was too short for the electronic circuitry of that time. In 1990, a revised version of the standard appeared in which this interval was increased to a value of at least 4 µs and at most 1 s. Equipment that supports this version (which in practice is far from being the most commonly used type) is declared to be compliant with 'DMX512 (1990)'.

If we combine all of the elements described above, we come up with the data-block timing diagram shown in **Figure 1**.

The transmission medium

Now that we have looked at the 'cars' (data blocks) that travel over the electronic DMX 'road', let's have a closer look at their characteristics of the road (in technical terms, this is called the 'physical layer'). If we enumerate the features of the protocol, we see that we have a connection that supports communications over the longest possible distance, a capacity of 250 kb/s and the need to simultaneously control a certain number of different devices. It is clear that our old friend, the RS-232 standard – with a maximum distance of 15 m between two points, a theoretical maximum speed of 19,200 baud and the electrical characteristics of a point-to-point link – is totally unsuitable for this task.

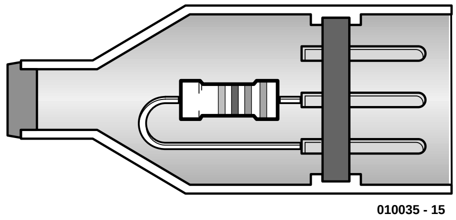


Figure 5. A homemade DMX terminator.

The DMX512 standard employs a different physical layer, which has been standardised under the name RS-485. This type of connection is derived directly from the RS-422 standard (which is primarily used for the serial ports of Apple computers), but it allows multiple receivers to be connected in parallel to a single transmitter (which is in fact the difference with RS-422).

The possibilities that RS-485 offers in comparison with RS-232 are extraordinary. Judge for yourself: the maximum speed is 10 Mbaud (that's right, 10,000 kbaud) over 100 m, or 1 Mbaud over 1000 m. The 250 kbaud of the DMX512 standard thus represents nothing to fear for components that comply with the RS-485 standard. In addition, the electrical characteristics are such that 32 receivers can be attached to the same cable. This is referred to as 32 unit loads (ULs).

Let's take a closer look at the UL concept. DMX installations quite often experience problems for the simple reason that the number of ULs connected to the transmitter is more than the upper limit of 32. This occurs most commonly with equipment in the lower price range in which optocouplers are directly connected to the DMX line, so that the receiver represents more than one unit load. We know that the RS-485 transmitters are linear analogue amplifiers that, quite properly, have current limiting circuitry. If the load exceeds the expected maximum value, the current limiting goes into action, which translates into the output stage of the chip becoming warmer. The majority of modern ICs have thermal protection, which completely cuts off the output stage if the chip temperature rises excessively. In the case of a DMX connection, this causes the communications to be interrupted for several minutes (until the chip has cooled down and communications are resumed). This type of behaviour generally provokes nervous breakdowns among users. Now you know what to expect if you ignore the limit of 32 connected devices.

What if you want to use more than 32 devices on the same line? Not to worry, you can buy repeaters from the makers of lighting equipment. A repeater counts as one unit load, but it can in turn be loaded with an additional 32 unit loads. (Nowadays, there are even chips that can handle 256 unit loads.)

You may wonder how RS-485 is able to transmit data over such a long distance. The answer is that it employs the differential mode, which means that for the receiver the important factor is the difference between the voltages on the two lead of the pair, rather than the difference between a voltage and a reference level as in RS-232 (see **Figure 3**.) One benefit of this approach is that it frees us from interference problems (the voltages on both conductors are displaced with respect to the reference level, but the difference between the voltages remains the same, so the receiver is not disturbed). A second benefit is that since no current flows in the ground conductor (the return current flows via the second lead of the pair), there is no common-mode voltage as with RS-232.

The previous paragraph is actually just an introduction to a discussion of the cable that must be used for the DMX link. The connector that is recommended by the standard is a 5-way XLR type, which is called a 'microphone connector' in the standard. This short phrase has seduced countless technicians into employing audio cable with a single screened twisted pair of conductors. This is just plain wrong! A special data cable that is suitable for RS-485 must be used (consult the catalogues of cable manufacturers), especially if the total length of the DMX connection is more than a few tens of metres.

It is essential that cable that is used contains a twisted pair of leads and is screened. The data leads must be connected to the proper pins of the XLR connector, as shown in **Figure 4**. The screen, which is usually grounded, is connected to the shell of the connector. The data leads must not make contact with the screen under any circumstances. Otherwise, in the best case the result will be an improperly functioning installation, and in the worst case the output stages of the RS-485 transmitters will be destroyed.

You may have noticed that there are two types of XLR connectors: a three-way type and a 5-way type. The DMX standard actually refers exclusively to the 5-way type and notes that outputs (mixer panels, master interfaces) must have female connectors and inputs (dimmers, lamps) must have male connectors. In short, most slave devices have two connectors, male at the input and female at the output.

So where does the 3-way connector come in? This is just a question of money. The 5-way connector is not used as much as the 3-way connector in the audio world, so it is more expensive. According to the standard, two of the pins re reserved for an optional link, which is almost never encountered in practice. Consequently, some manufacturers decided to equip their equipment with 3-way connectors, which does not cause any problems – at least not in theory, but in fact the well-

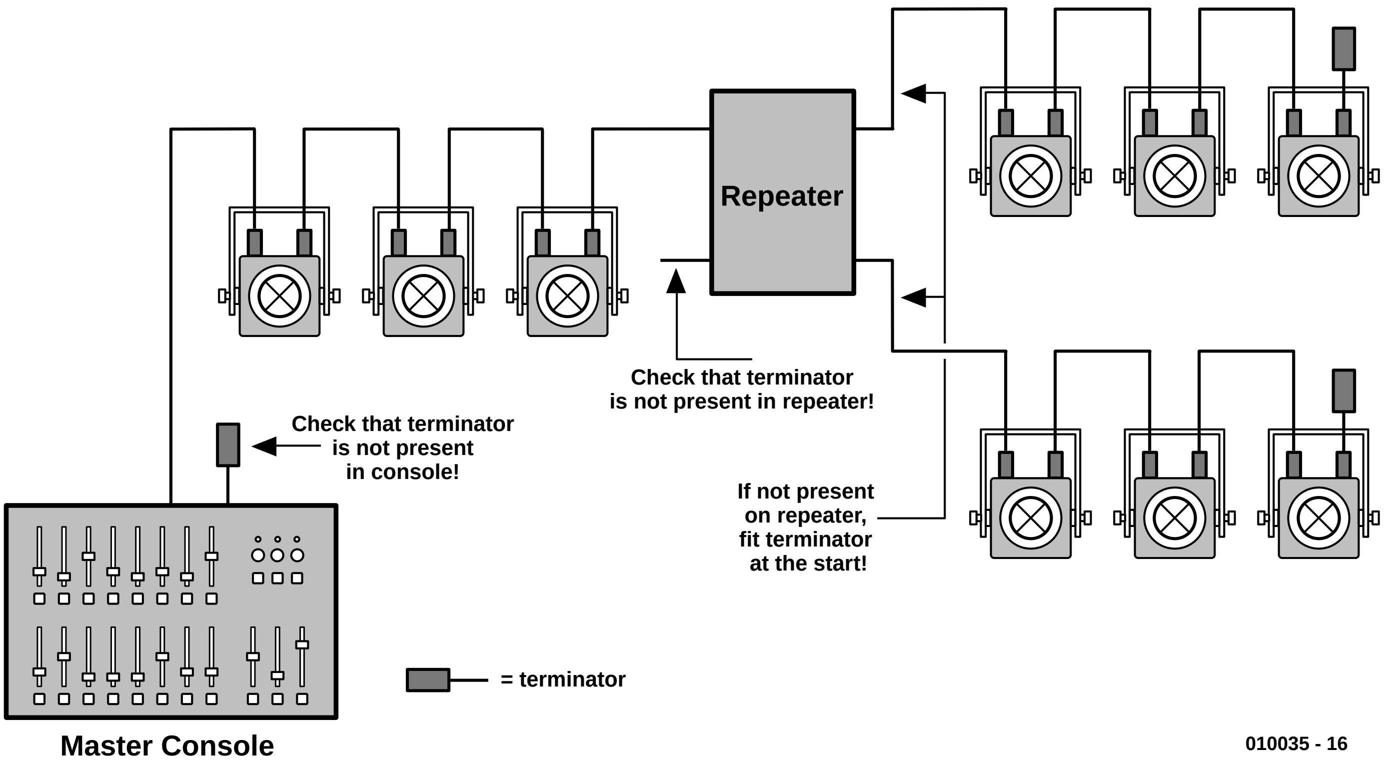


Figure 6. Terminator locations in a DMX512 installation.

known Danish manufacturer Martin came up with the brilliant idea of reversing the functions of pins 2 and 3 (data+ and data–) in their first generation of DMX-compatible products. What makes matters worse is the fact that a few years ago, they decided to go back to the standard connection!
It is thus highly recommended to ensure that you have a few 3-way/5-way adapters on hand – you may need them. In addition, make sure that you have one or two crossover cables for old-style Martin equipment. Recently we have read that in certain equipment, the two spare pins are used for a second DMX line, which is most often used as a return link. Take our advice: read the instructions carefully!

Don't forget the terminators!

We're nearly finished with the theoretical discussion of DMX, but there's one more thing to consider: terminators. The cables that we have described cannot simply block the transported data so that it all accumulates at a particular location. They need to be fitted with terminators, which are electrical components that are attached to the ends of a transmission line in order to eliminate reflections. This is a rather complicated subject, and a full explanation is certainly outside the scope of this article. However, you can imagine that the signals 'rebound' from the end of the cable, which creates high-frequency electromagnetic waves that arrive at the circuit inputs.
In order to explain what happens in practice, let's assume that a logic '1' is travelling along the DMX cable. When it reaches the end of the cable, this pulse will be reflected and travel back towards the transmitter, which we assume to have just sent out a '0'. The problem with very fast connections, such as DMX (don't forget that we are dealing with 250-kHz rectangular signals, which means a very wide spectrum), is that the transmission time for the signal (which travels at around 80% of the speed of light in a copper cable) is approximately the same as the pulse duration. Our logic '1', which is travelling back along the cable, thus has a chance of meeting the logic '0' that has just left the transmitter.
If this happens, the result is not a short circuit, but instead either a cancellation or an overshoot (depending on the topology of the bus) that completely distorts the transmitted signal. The receivers cannot make head or tail of the resulting signal and go into the wait state, so the DMX cable does not work properly and appears to be broken. In certain cases, false Breaks can be generated, which results in total chaos.
Such situations can be easily prevented by simply 'adapting' the transmission line. Transmission line theory shows that reflections can be eliminated by attaching a load resistance equal to the characteristic impedance of the medium. The characteristic impedance depends on the geometry of the medium (in our case, the cable). For instance, cables that are used for cable television have a characteristic impedance of 75 Ω. This does not mean that the cable has a resistance of 75 Ω (which would be pointless), but rather that a resistance of this value must be placed at the end of the link to avoid reflections. As an experiment, you can disconnect the termination resistor in a television tuner and observe the consequences on the screen: the picture appears to be repeated several times, each time with an offset. This is caused by the fact that the picture signal is reflected several times at both ends of the cable. RG58 cables, which are used for radio reception, are dimensioned for an impedance of 50 Ω. In the case of RS-458 cables, the characteristic impedance is typically 120 Ω.
How can you make such a terminator? It's very simple; all you have to do is to solder a 120-Ω resistor (in practice, any value between 100 Ω and 150 Ω can be used) across pins 2 and 3 of an XRLR connector (see **Figure 5**).
The terminators must be placed at the two extreme ends of the DMX cable (they are useless at any other location). If you are working with a simplex DMX connection, the terminator at the transmitter can be omitted if necessary. Make sure that there are not more than two terminators connected to a single section (one at each end), as otherwise the amplifiers that provide the transmitted signal will overheat, since the resistors are in parallel. I can hear you thinking, 'OK, but what about branch circuits? They also have ends.' The answer is simple: branch circuits are unnecessary, except where you have placed a repeater (see **Figure 6**). It goes without saying that a repeater must be regarded as the end of a cable section, so a terminator must be fitted at this location (but bear in mind that many commercially available repeaters have built-in terminators).

Coming soon: Ethernet

It's been a long journey through the land of DMX, but now we know what is important for its proper operation.
However, our story about remote control of lighting systems would not be complete without a few words concerning the future of this technology. DMX is powerful enough to control the majority of lighting installations, up to a considerable capacity. For large installations, however, such as are found in television studios and on certain concert stages, DMX turns out to be inadequate. In addition, it has been frequently been noted that DMX does not have a return line for monitoring the execution of its instructions, which means that this technique is not allowed to be used in certain areas, such as pyrotechnics and hydraulic equipment. (In such cases, the preferred solution is MIDI Show.)
For some time now, a new standard based on Ethernet has been making itself known. This standard eliminates all problems related to speed (Ethernet works at 10 Mbaud, 40 times as fast as DMX) and the absence of a return signal (Ethernet is by design a duplex system). On top of this, Ethernet cards for PCs and Mac computers are extremely inexpensive and are directly supported by the most important operating systems (Windows, Unix, Linux, MacOS and their relatives), which naturally makes them much easier to use.
As you can see, the world of lighting and special effects is highly dynamic, with the best contemporary techniques being used to pamper our eyes.
The practical side of the subject will be dealt with shortly in *Elektor Electronics*.

(010035-1)

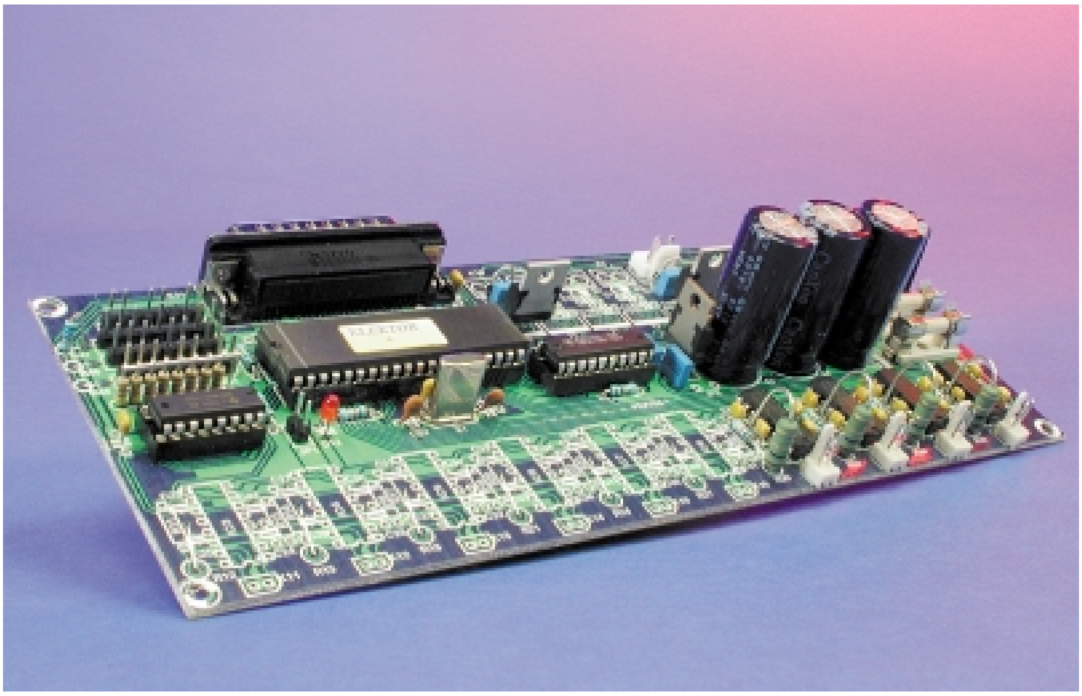
PCB Drilling Machine (3)

Part 3: PCB construction

Design by T. Müller (Radix GmbH)

www.radixgmbh.de

The controller board contains all the electronics for the plate motor and four drill arms. As a matter of course, the board population is in accordance with the actual number of drill arms you have in use.



All the components of the circuit can be built onto a single-sided printed circuit board. The PCB and component layout are shown in **Figure 1**. The power supply (as far as the transformers) is also built onto the circuit board: it provides a +30 V supply for the solenoids and stepper motor drivers, +15 V for the drill motors, and +5 V for the logic, regulated by IC1. The layout is very straightforward: along one long edge of the board we have the stepper motor drivers, each with a pair of terminal posts (K11-K20) for connection to the stepper motor windings. Since comparatively high currents flow through the ICs, they should be soldered directly to the board without sockets.

Along the other long edge we find the terminals for the jumpers and switches, the 25-way sub-D connector, the output stages for the solenoid drivers and drill motors (connected via the small PCB connectors K3-K10), and finally the power connectors K1 and K2. The microcontroller, the GAL and the shift register are in the middle of the board, and naturally good quality IC sockets should be used for them. One toroidal mains transformer with 2·12 V/ 80 VA secondary windings is used for each pair of drilling arms. If four arms are fitted, then the windings of the two transformers should be wired in parallel to connectors K1 and K2.

Fitting the components should present no difficulties. One point to note is that although all the ground connections are linked, the two marked heavy gauge wire links on the circuit board next to the fuses must be soldered in. This is to avoid long current paths over the circuit board and the interference problems this causes. The ground plane is split between the sub-D connector and the nearest corner of the circuit board: with the type of drilling motor used this gives the best EMC performance. With other types of motor it may be necessary to bridge the gap.

The reset input JP1 can be brought out and used as a kind of emergency switch. Using this switch will certainly confuse the system, however, since the controlling PC

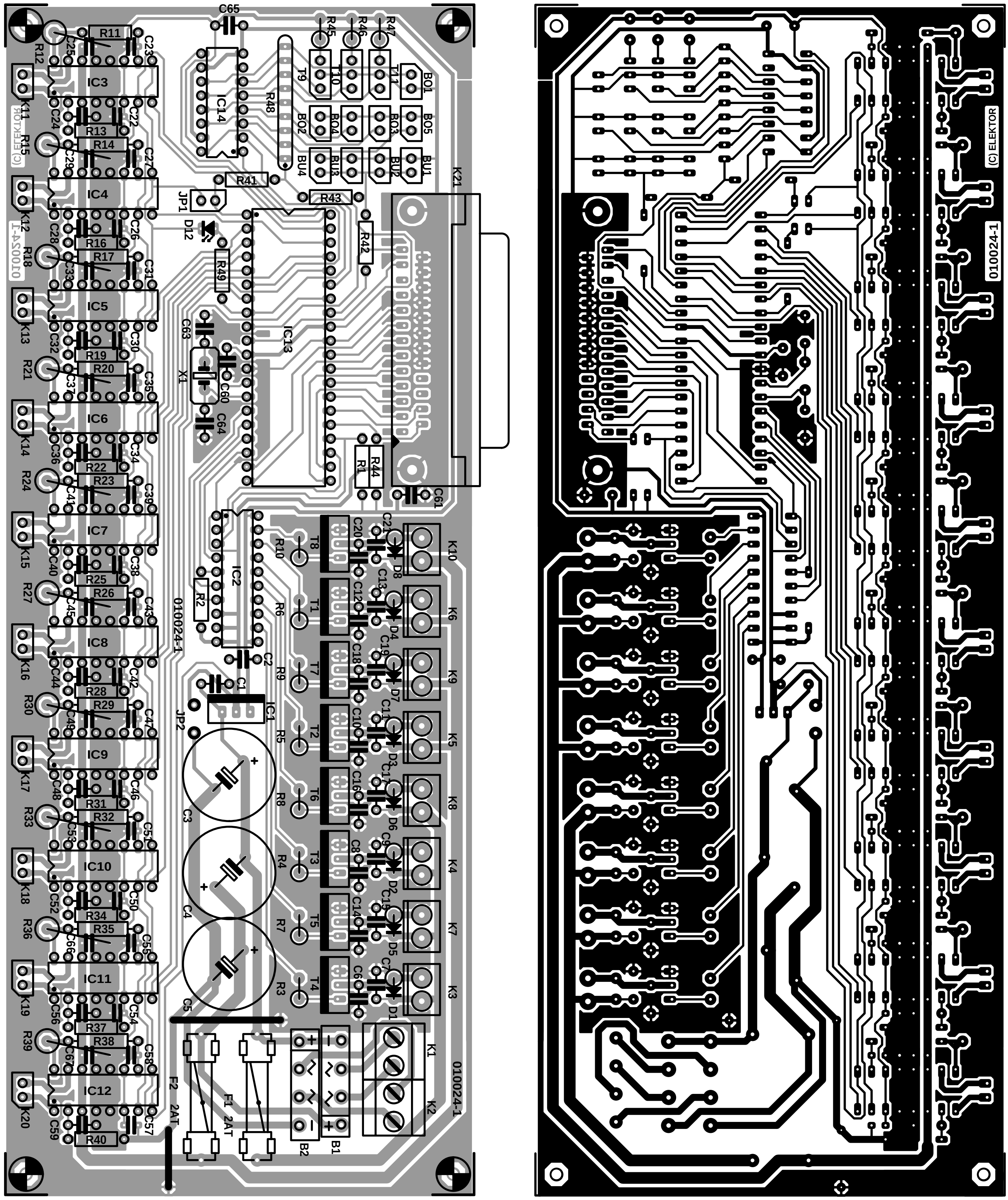


Figure 1. Track layout and component mounting plan of the controller board.

will not know when it has been operated; but in any case an emergency switch is not a bad idea.

(010024-3)

In the next instalment of this series we get down to the nuts and bolts of the project. The various parts of the PCB drilling machine will be brought together, with illustrations — and lots of glue and grease!

Compact PCB drilling machine

After discussions with several readers during the HobbyTronic Fair in Dortmund, Germany, and several e-mail enquiries and telephone calls, we have decided to describe briefly once more the operation of the PCB drilling machine, along with its advantages and capabilities.

Basic features of the PCB drilling machine

Our project is a CNC machine which carries out operating commands calculated with the help of a computer. Conventional CNC machines are constructed using linear components, such as worm drives, ball races, linear guides and many other — not inexpensive — specialist components. The construction of a linear machine appears very straightforward: simply select the required components in the appropriate sizes, fit them together, and set the computer going. But the devil is of course in the detail: all the parts must be precisely mounted, parallel, true, and free of play, or else things will grind and jam.

Our PCB drilling machine is, as far as function and aims are concerned, exactly like an X/Y/Z-machine. It can handle circuit boards up to 200 mm by 300 mm with an accuracy of 0.03 mm, which is entirely adequate for our application. Because of its lightweight construction it can move extremely quickly, at almost 80 mm/s. In construction it is fundamentally different from linear machines. Everything is reduced down to two rotating, vertical axles. The only specialist components required are good-quality ball bearings to support these axles without play. In appearance, the machine resembles a record player: in the middle is a rotating table, on which the workpiece — the circuit board — is fixed. Instead of a cartridge we have a drill, which can be moved up and down. And in exactly the same way as the tone arm can reach any point on a record, the drill can get to any point where drilling is required on the circuit board.

As well as the rotating table, there is space for more tool arms, which can all operate practically simultaneously on the same circuit board. The advantages are that the job gets done quicker, and no tool changes are required.
The tool can be a drill, which can be used for drilling holes for the leads of components, or a milling tool, with which the conductors can be milled away. Drilling data is taken from one's preferred circuit board design program in Excellon format: all programs are capable of outputting this format. Milling data is sent to the machine in the HPGL plotter control language, allowing any desired cutting path to be followed. The machine can be connected directly to the printer port of an ordinary PC without extra hardware.

The machine is not designed for heavy-duty milling operations. If you want to produce milled aluminium front panels, you will have to look elsewhere. The key feature of the machine is its high speed, which can only be obtained with a lightweight construction. It is sturdy enough for working on printed circuit boards, but not sturdy enough to withstand the high sideways forces that arise when milling metals. The machine has to be small and light enough so that it finds a place not just in the workshop, but actually on the experimenter's bench

COMPONENTS LIST

Basic version, w/o tool arm

Resistors:
R1,R2,R41,R43,R44 = 10kΩ
R11,R14 = 1kΩ
R12,R15 = 1Ω, 1W
R13,R16 = 56kΩ
R42 = 220Ω
R45,R46,R47 = 560Ω
R48 = 8-way resistor array 47kΩ
R49 = 1kΩ5

Capacitors:
C1,C2,C60,C61,C65 = 100nF, 5mm lead pitch
C3,C4,C5 = 3300µF 50V radial
C22,C25,C26,C29 = 100nF, 2.5mm lead pitch
C23,C27 = 820pF, 2.5mm lead pitch
C24,C28 = 820pF, 5mm lead pitch
C63,C64 = 15pF

Semiconductors:
B1,B2 = GBU6B
D12 = LED, red, 3 mm dia.
IC1 = 7805
IC2 = GAL 16R8 (programmed, order code **010024-31**)
IC3,IC4 = PBL3717A
IC13 = PIC16C64-20P (programmed, order code **010024-41**)
IC14 = 74HC165

Miscellaneous:
K1,K2 = 2-way PCB terminal block, lead pitch 5mm
K11,K12 = 2-way pinheader, 0.1in. lead pitch
K21 = 25-way sub-D plug, angled pins, PCB mount
F1,F2 = fuse, 2A (T) time lag with PCB mount holder and cap
BU1-BU4,BO1-BO4,JP1 = 2-way SIL pinheader
JP2 = jumper
X1 = 20MHz quartz crystal
PCB, order code **010024-1**, see Readers Services or www.elektor-electronics.co.uk

COMPONENT LIST

(external components)

1 mains transformer for 2 drill arms: toroidal 2 x 12V 80VA
T9,T10,T11 = SFH309-F4 or LTR4206E
IR-LED: TSIP4400
Stepper motors type KH2JM2-851

COMPONENTS LIST

(for Tool Arm #1, replicate for other arms)

Resistors:
R8,R10 = 100Ω
R17,R20 = 1kΩ
R18,R21 = 1Ω, 1W
R19,R22 = 56kΩ

Capacitors:
C16,C20 = 100nF, 5mm lead pitch
C30,C33,C34,C37 = 100n, 2.5mm lead pitch
C17,C21 = 220nF, 5mm lead pitch
C31,C35 = 820pF, 2.5mm lead pitch
C32,C36 = 820pF, 5mm lead pitch

Semiconductors:
D6,D8 = BUV27-200
T6,T8 = BUZ11
IC5,IC6 = PBL3717A

Miscellaneous:
K8,K10 = 2-way pinheader, lead pitch 3.96mm
K13,K14 = 2-way pinheader, lead pitch 2.54mm

The content of this note is based on information received from manufacturers in the electrical and electronics industries or their representatives and does not imply practical experience by Elektor Electronics or its consultants.

New Oscillator ICs

for operation from 45 to 650 MHz

By G. Kleine

With the introduction of the MAX260x series of oscillator chips from Maxim it is possible to build a stable oscillator circuit with predictable characteristics. An external inductor is all that is necessary to set the centre frequency of the oscillator.

More and more of today's electronic systems have the need for a good quality, reliable and compact frequency source e.g. radio system local oscillators and phase lock loops (PLL) or microprocessor and data acquisition circuits. The MAX260x series of oscillator chips from Maxim combines an integrated voltage controlled oscillator (VCO) together with on-chip varacter diode and differential output drivers. The differential output can be used as a symmetrical HF output or as two single ended outputs with good decoupling as typically required in PLL applications.

The design of an oscillator circuit for use at HF, VHF and UHF (30 MHz to 1 GHz) is not a trivial exercise. The designer needs to take into account many factors. The circuit must be guaranteed to oscillate under all conditions with extremes of component tolerances and provide an output with good frequency stability and output swing independent of the output loading. If the frequency of oscillation is to be controlled by an external analogue control voltage the oscillator will be a voltage controlled oscillator (VCO) here the design becomes a bit more involved where the control voltage/frequency characteristic must also be considered. A typical discrete VCO is shown as **Figure 1.** Design problems become more acute as the operating frequency of the oscillator increases. Parasitic properties of the components begin to have significant influence on the performance of the design. Layout of the oscillator, inductance of the PCB tracks and other unwanted coupling effects in the circuit must also be taken into account. The oscillator frequency can also be affected by factors such as output loading (also known as *load pulling*) or changes in its supply voltage (also known as *pushing*). A good low noise isolated voltage regulator for the oscillator will go a long way to solving this last problem and will also improve the oscillator phase noise performance. The final stage of the design process for a discrete VCO would involve prototype construction and testing under extremes of temperature.

The integrated solution from Maxim

Maxim has introduced a family of integrated oscillators suitable for fixed frequency or narrow band VCO operation. Five different models cover the frequency range from 45 to 650 MHz. These devices allow very simple and compact implementation of an oscillator circuit. (see **Table 1**).

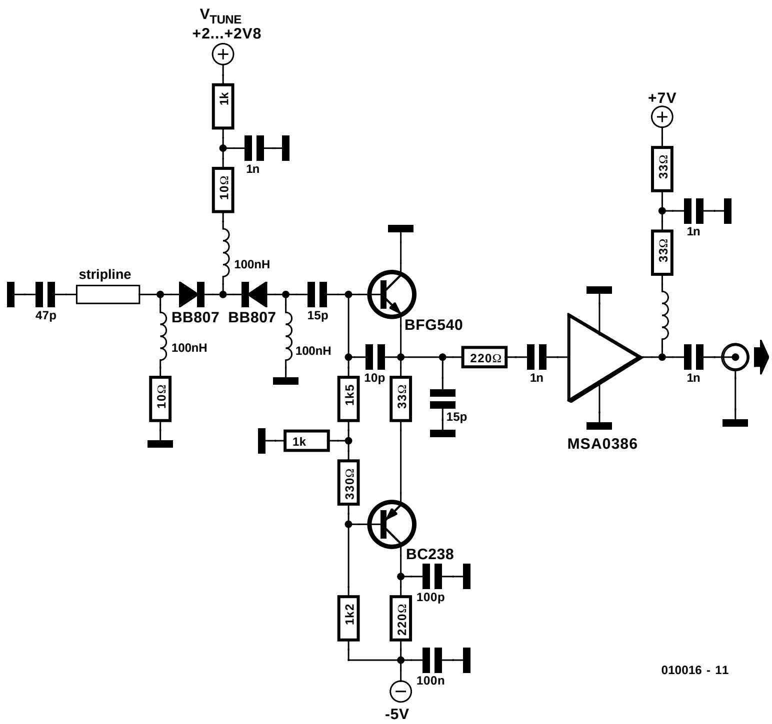


Figure 1. Circuit diagram of a typical discrete VCO.

Table 1. Frequency and inductor range.

Type	Frequency range	Inductance range	min. Q
MAX2605	45 ... 70 MHz	680 ... 2200 nH	35
MAX2606	70 ... 150 MHz	150 ... 820 nH	35
MAX2607	150 ... 300 MHz	39 ... 180 nH	35
MAX2608	300 ... 500 MHz	10 ... 47 nH	40
MAX2609	500 ... 650 MHz	3.9 ... 15 nH	40

The feedback capacitors and varacter are all integrated on the oscillator chip. To produce a correctly operating VCO it is only necessary to fit an external inductor. The width of the tuning range is such that with 2% tolerance inductors no board-level adjustments or trimming of the VCO is necessary. Once the correct value of inductance is fitted to the oscillator, the VCO is guaranteed to tune to the desired operating frequency.

The MAX260x family operate with a supply voltage of between +2.7 V and +5.5 V, requiring a current of less than 5 mA. A look inside the chip **(Figure 2)** shows that the oscillator design is based on the familiar Colpitts configuration. The feedback capacitor at the base-emitter junction together with the external inductor determine the frequency of oscillation. The built-in varacter diode in the feedback loop allows for the correction of component tolerances of ±3 % by applying an external trimming voltage of between +0.4 V and +2.4 V. The circuit is designed for reliable and stable operation throughout its operating temperature range. The phase noise performance quoted in the data sheet will only be achieved if the Q of the external inductor is greater than that shown in **Table 2.**

A circuit showing a MAX260x configured as a VCO is shown in **Figure 3**. The external inductor is the single component necessary to set the oscillator frequency. The inductance seen at pin 1 (including the PCB track inductance) determines the frequency of oscillation. If the inductor value necessary for your application does not coincide with a value in the standard range then it is possible to use two inductors to achieve the desired value. Choose the biggest value of inductor to have a value just less than the value required, making sure that it meets the minimum Q constraints given in **Table 1**. The second inductor will be a much smaller value and can also have a much lower value of Q. It will not have a great influence on the overall Q of the two inductors. A thin film SMD inductor would be suitable here or at higher frequencies it would be possible to use a PCB track run of the correct length as an inductor.

Shown in **Table 3** is the frequency adjustment range corresponding to a control voltage range of +0.4 V and +2.4 V together with the phase noise performance offset 100 kHz from the carrier.

To ensure that an oscillator can be built that will not require trimming it is important that the tolerance of the two inductors must be less than 2% of their nominal value. At frequencies above about 150 MHz the layout of the circuit and PCB track inductance begin to have a significant influence on the circuit performance. **Table 4** shows standard values of inductors together with the frequency of operation of the VCO. These values should be seen as a starting point for inductor values

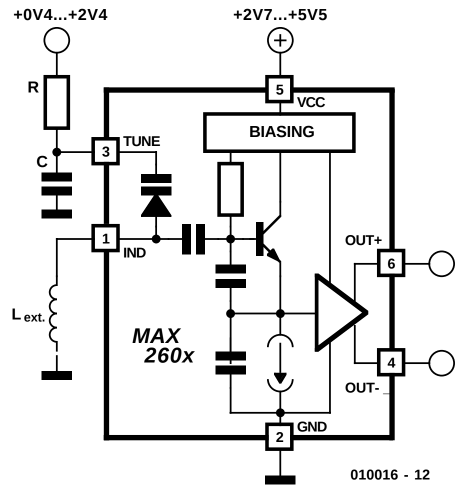


Figure 2. Block diagram of the MAX260x family.

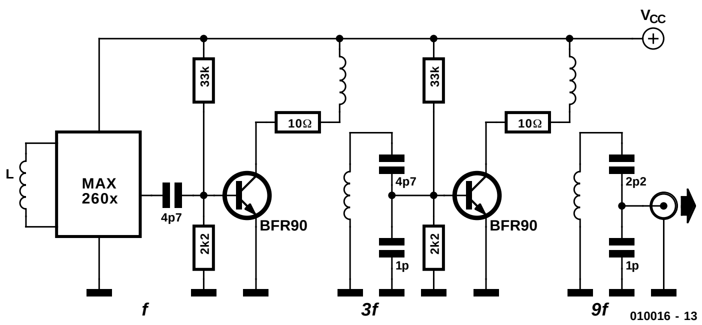


Figure 3. Typical application circuit of the MAX260x.

Table 2. Technical Data for the MAX260x family.

Type	Vcc	Current consumption	V_{TUNE}	Output power (single ended)	Pushing
MAX2605		1.9 mA			60 kHz/V
MAX2606		2.1 mA			120 kHz/V
MAX2607	+2.7V to +5,5V	2.1 mA	+0.4 to +2.4 V	typ. –10 dBm	220 kHz/V
MAX2608	(max. +6V)	2.7 mA	(max. Vcc +0.3 V)		480 kHz/V
MAX2609		3.6 mA			720 kHz/V

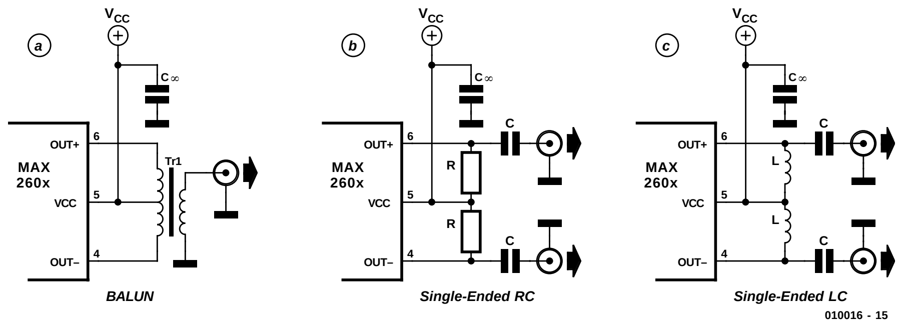


Figure 4. Possible output configurations: a) BALUN, b) RC, c) LC

and the optimal value will be determined once the circuit layout has been built and tested.

Coil selection

Coils suitable for use with the MAX260x series can be obtained from Neosid, Toko and Coilcraft. Toko inductors are distributed by B.E.C. Ltd or Roxburgh Ltd while Coilcraft Products are supplied by Coilcraft Europe. Neosid are supplied by Sterling Components Ltd. **Table 5** shows the recommended series from these suppliers. All the series are SMD inductors. The air-cored coils have plastic formers to enable them to be placed onto the PCB automatically by machine. The air spaced coils have a higher Q than the chip inductors and give a better phase noise performance for the oscillator.

The coil Q factor

Table 1 gives the minimum Q factor necessary of the inductor to guarantee oscillator start-up and ensure that phase noise of the oscillator falls within the specification given in the data sheet. The Q factor or 'goodness' of the inductor is defined as the ratio of the stored energy to the lost energy and can be expressed as:

$$Q = R_p / X_p$$

R_p is the equivalent parallel resistance (the loss resistance) and X_p is the parallel reactance including all the parasitic effects like winding capacitance. It can be seen that for a high Q inductor the losses must be minimised so that R_p is much greater when compared with X_p. This can be achieved by using air spaced or ferrite cored inductors with low eddy current losses.

In contrast to the high Q inductors that we need for the resonant circuit of this oscillator, there are also other types of inductors that are used to suppress unwanted interference in electrical circuits. This type of inductor has a deliberately low Q so that they produce a greater damping effect on unwanted wide-band interference. Using a high Q inductor for interference suppression would have the effect of creating a resonant circuit, increasing interference rather than damping it.

The output stage

The MAX260x series have differential outputs (OUT+, OUT-) both outputs are driven by open collector transistors so it is essential that any circuitry connected to these outputs has a load connected up to Vcc. **Figure 4** shows some of the possible output configurations. The differential outputs can be used directly or can be connected to a balun (balun = balanced to unbalanced converter) to provide a single ended output (**Figure 4a**). Using the differential output mode gives maximum output power with minimum unwanted harmonics. Alternatively, the two outputs can be used independently as in **Figure 4b.** One example would be in a PPL prescaler application where one output drives the frequency divider chain and the other is used as the VCO output signal. The isolation of the two outputs ensures that unwanted noise and side bands from the frequency divider is not superimposed on the VCO output signal. Output impedance matching can be performed with LC circuits as shown in **Figure 4c**.

Figure 5 shows how the two outputs would be employed in a PLL prescaler application. One output drives the PLL divider stage and filter while the second output is used as the local oscillator input to the mixer stage. If the mixer and divider stage were driven by the same output then we would find that interference from the PLL divider circuitry produces interference in the form of sidebands spaced at multiples of the phase detector switching frequency appearing in the wanted signal.

Applications

The MAX260x series of oscillators were designed for use as fixed fre-

Table 3. VCO Frequency Limits and Phase Noise

Type	Frequency limits @ V_{TUNE} = +0.4 V to +2.4 V	Phase noise @ 100 kHz
MAX2605	–2,25 % - +2,25 %	–117 dBc/Hz
MAX2606	–2,5 % - +2,5 %	–112 dBc/Hz
MAX2607	–2,75 % - +2,75 %	–107 dBc/Hz
MAX2608	–2,8 % - +2,8 %	–100 dBc/Hz
MAX2609	–3,0 % - +3,0 %	–93 dBc/Hz

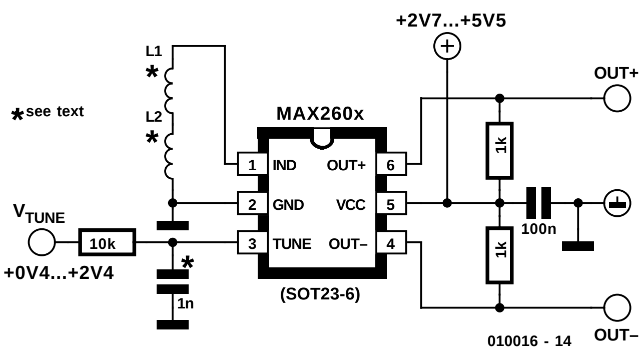


Figure 5. Using the two outputs in a PLL application.

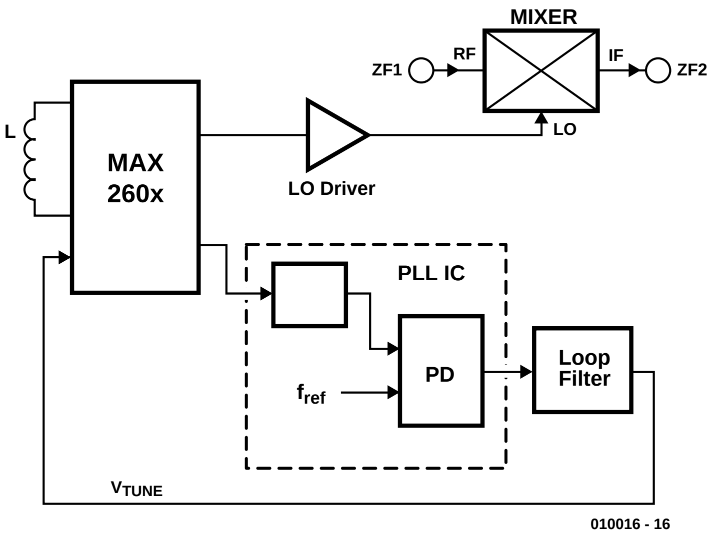


Figure 6. MAX260x with a frequency multiplier.

quency or for narrow band VCO applications. For example conversion oscillators in radio receivers operating at a fixed frequency and connected via a PLL to a reference frequency.

Another application is as a narrowband FM modulator. The operating frequency can be set by a dc voltage applied to the VCO control input, modulated by a superimposed ac signal.

For frequencies above 650 MHz a frequency multiplier circuit can be used (**Figure 6**). Here a transistor stage configured as a class C amplifier is driven by the oscillator output. It switches at the fundamental frequency of the oscillator but because it is producing a square wave it also generates odd harmonic frequencies. The following transistor circuit contains a filter tuned to the harmonic of interest this can usually be the third or fifth (higher harmonics generally have too much attenuation to be of use). Tuning to the fifth harmonic will enable the circuit to operate up to a maximum frequency of 3 GHz with the Maxim series of oscillators. The multiplied frequency is passed through another filter and then to the output.

Table 4. Output frequency and corresponding inductor values.

MAX2605		MAX2606		MAX2607		MAX2608		MAX2609	
L	f	L	f	L	f	L	f *	L	f *
1500 nH	48 MHz	680 nH	75 MHz	150 nH	150 MHz	33 nH	300 MHz	10 nH	500 MHz
1200 nH	55 MHz	470 nH	95 MHz	100 nH	190 MHz	22 nH	350 MHz	8.2 nH	550 MHz
1000 nH	60 MHz	390 nH	100 MHz	68 nH	220 MHz	15 nH	420 MHz	6.8 nH	590 MHz
820 nH	67 MHz	220 nH	130 MHz	47 nH	270 MHz	12 nH	470 MHz	4.7 nH	630 MHz

(= The layout is critical at these frequencies!)*

Table 5. Coils suitable for use with the MAX260x family of VCOs.

Manufacturer	NEOSID	COILCRAFT	TOKO
Distributor	Sterling Components	Coilcraft Europe	B.E.C. or Roxburgh
Airspaced	SM-L1.5 (5.3 - 43 nH)	Micro Spring (1.6 - 12 nH)	33CS (6.8 - 22 nH)
		Mini Spring (2.5 - 43 nH)	36CS (22 - 50 nH)
		Midi Spring (22 - 120 nH)	
		Maxi Spring (90-538 nH)	
SMT inductors	SM-1206 (6.8 - 150 nH)	0805HQ (2.5-51 nH)	LL1608FS (1.2 - 270 nH)
	SM-NE29 (10 - 1000 nH)	1008HQ (3-100 nH)	LL2012FH (1.5 -680 nH)
	SM-NE45 (100 - 2200 nH)	1206CS (10-1200 nH)	FSLM2520 (100 - 2200 nH)
		1812CS (1.2-33 uH)	

To sum up

The MAX260x series of oscillators are ideal for applications requiring a fixed frequency oscillator or narrow band VCO. Development time is kept to a minimum and its 6-pin SMD package takes up a tiny area of the PCB compared to a discrete oscillator design. An external inductor is the only component necessary to define the oscillator centre frequency. The inductor characteristics are not especially critical and can be obtained from many suppliers.

(010016-1)

Web links

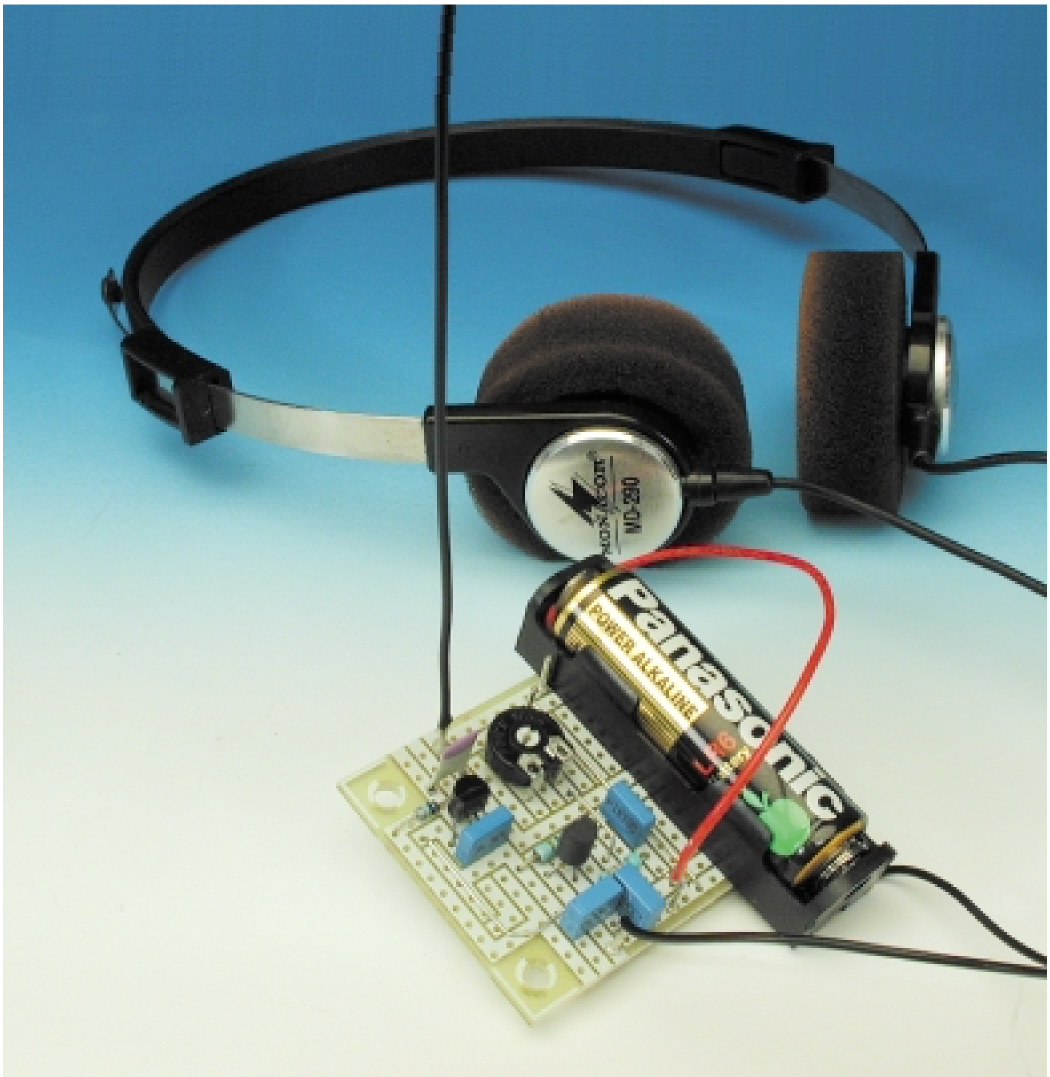
MAXIM:	www.maxim-ic.com
Coilcraft:	www.coilcraft.com
Toko:	www.tokoam.com
Neosid:	www.neosid.com
BEC:	www.bonex.co.uk.
Sterling components:	www.sterling-comp.co.uk.

1.5-Volts Medium Wave (MW) Radio

and now for something smaller.

From an idea by W. Zeiller

In this day and age when we're surrounded by hi-tech we should leave it all behind and return to a very basic circuit. So stop thinking for a moment about faster and bigger — it's the small things in life which are best!



Since we're continuously exposed to hi-tech gadgets, it's easy to forget that we can still build interesting circuits with only a few components. There is nothing revolutionary about this circuit, but this shouldn't deter the true electronics hobbyist. Apart from the educational aspect, it is satisfying to build a well-working circuit that uses only a few simple components.

The circuit for a radio receiver is still a good example of this. A medium wave radio in particular becomes a very simple circuit when we go back to basics. It is also one of the most satisfying circuits to construct: you solder a few components onto a PCB, connect a battery and you're greeted with music from the headphones. A very memorable experience!

Direct Conversion

What would be the simplest and most compact design for a medium wave radio? A superheterodyne receiver, as was used in the 'AM Receiver' in the February 2001 issue, is not suitable since it makes the circuit too complex. Rather, we should think about direct conversion techniques, as found in the good old crystal receiver.

The principle of operation of a direct conversion receiver is that the received signal (after selection and amplification) is demodulated directly, without the use of IF oscillators and the mixing of signals. In a nutshell, a direct conversion receiver is the purest type of radio, giving a superior sound quality; the total absence of whistles and squeaks is also a contributory factor.

In its simplest form, a direct conversion AM receiver consists of an RF input circuit, a diode detector and high-impedance headphones. Such a miniature radio can easily fit inside a matchbox and has the added advantage that it doesn't need a power supply. A shortcoming of such a receiver is that its sensitivity is a bit low, because no amplification takes place. You'll need a good aerial and ground connection for worthwhile reception to occur.

A big improvement in its performance can be obtained by amplifying the received RF signal before it is fed to the detector stage. This increases the sensitivity to a level where reasonable reception is still obtained when using a whip or ferrite rod aerial. The selectivity of the receiver can be improved by designing the RF amplifier with a high-impedance input, reducing the load on the RF input stage.

Clever chip

The design of the above circuit is enormously simplified by the availability of a very useful and inexpensive IC. This is the small MK484, which has a TO92 case like a BC547. In the past Ferranti made this chip, with a part number of ZN414. In fact, this integrated marvel contains a complete AM receiver, requiring only a tuned circuit and an optional audio amplifier to make a complete radio.

Figure 1 represents the internal block diagram of the MK484. As can be seen, this IC contains a high-impedance input stage (shown here as an emitter follower), a (three-stage) RF amplifier, an AM detector and an Automatic Gain Control circuit (AGC), which adapts the amplification according to the level of the input signal. The input stage has an input impedance of 4 MΩ, which makes it possible to connect the MK484 directly to the tuned circuit without having to worry that it causes too much loading. The coil of the tuned circuit therefore doesn't need a tap, which simplifies the construction.

What isn't shown in **Figure 1**, but what is a very useful property of the MK484, is that it, like its illustrious predecessor, has been designed to work from a supply of 1.5 V (1.1 to 1.8 V), with a current consumption of only 0.3 mA. This makes it possible to power the receiver from a single penlight (AA) battery; even a button cell has enough capacity to power the MK484 for several hundred hours!

Using a minimum of components

The circuit diagram in **Figure 2** is of a typical receiver with an MK484 at its hart. It can be seen that only a few external components are required: a tuned circuit (L1/C1), a bias resistor (R1) and a load resistor (R2), which also sets the operation of

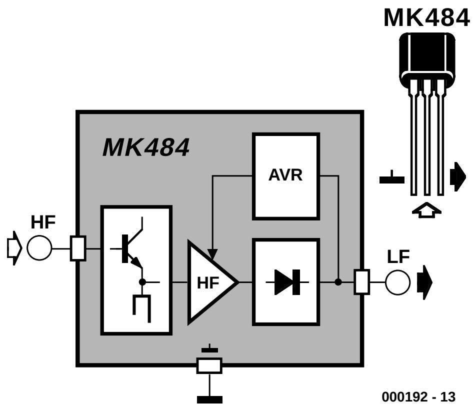


Figure 1. The MK484 contains nearly all the parts required to make a simple AM receiver.

the AGC. If you can get hold of a high-impedance (200 to 1000 Ω) magnetic earphone you could mount this in place of R2, which makes the amplifier built round T1 superfluous — the number of components required is then really at an absolute minimum. But since these magnetic earphones are hard to come by, T1 is used to drive a pair of headphones with an impedance of 32 Ω. The number of components used remains fairly small, even with this extra stage.

For reception of stations broadcasting in the medium wave band, the variable capacitor (C1) needs to have a value of 200 pF and inductor L1 should be 470 µH. The coil is made by winding 80 turns of 0.2 mm enamelled copper wire (ECW) onto a 5 cm long ferrite rod, which should have a diameter of about 10 mm.

When using 32 Ω headphones we'd recommend that the two speakers are connected

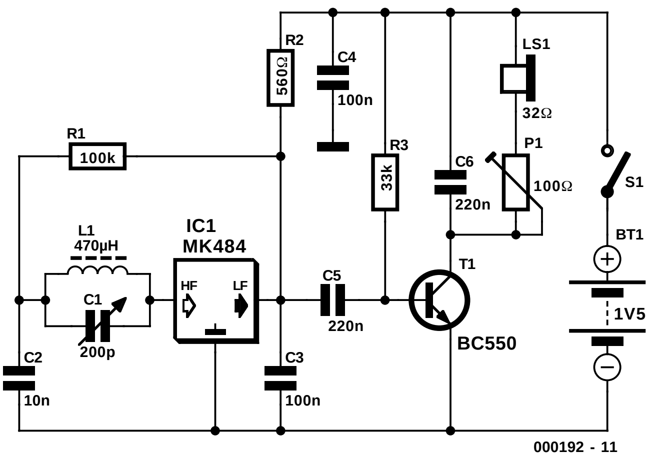


Figure 2. Only a small number of external components is needed. T1 is the headphone amplifier.

in series. This increases the impedance to 64 Ω, which halves the current drain of the audio amplifier built round T1. There is no need to worry that the sound becomes too quiet; you're likely to reduce the volume further with P1.

Coil-less

A competent hobbyist would surely be able to construct the circuit of **Figure 2** such that it fits inside a matchbox, using a button cell for the supply. This is something we've managed to do in the past with a similar design.

Under certain conditions the circuit can be reduced even further. If there is only one strong local medium wave transmitter it becomes possible to leave out the tuned circuit all together. Due to the high gain of the MK484, such a station can be received clearly even without the tuned circuit. The narrow bandwidth of the IC (150 to 3000 kHz) suppresses most interference which could be caused by local shortwave transmitters, if there are any.

This leads us to the circuit shown in **Figure 3**. Since we've lost our aerial with the removal of the ferrite rod, we have to connect a short whip aerial via C1. Some experimentation with the length of the aerial and the value of C1 will be needed in order to get the best reception.

Stamp

With the removal of the tuned circuit we've got rid of the two largest components. So while the circuit of **Figure 2** fits inside a matchbox, that of **Figure 3** could be reduced to the size of a stamp — more so if a fixed resistor is used in place of P1. It will be a challenge for the resourceful constructor to make this receiver as small as possible.

For those of us who don't like such a level of miniaturisation, we've designed a circuit layout for the coil-less receiver shown in **Figure 3**, using an 'UPBS-1' experimenters' board (UPBS = Universal Prototyping Board Size-1). These universal boards are available via our Readers Services and are ideal for this type of project. The recommended layout is shown in **Figure 4**.

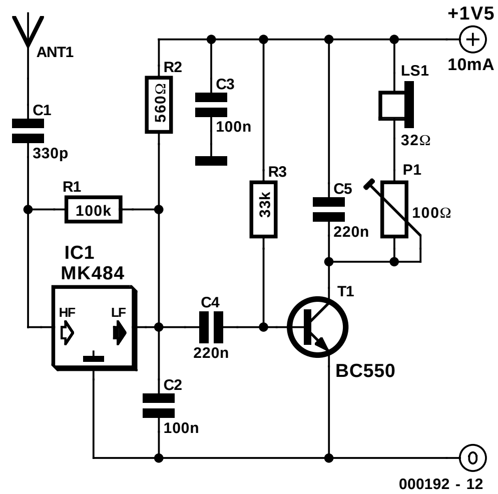


Figure 3. There is not much left without the tuned circuit, but it is still functional.

COMPONENTS LIST

Resistors:
R1 = 100kΩ
R2 = 560Ω
R3 = 33kΩ
P1 = 100Ω preset H

Capacitors:
C1 = 330pF (see text)
C2,C3 = 100nF
C4,C5 = 220nF

Halfgeleiders:
T1 = BC550
IC1 = MK484 (F-501), Conrad Electronics # 178535-88

Miscellaneous:
Headphones 32Ω (or more)
Experimenters' board type UPBS-1 (see Readers Services page)

We'd like to mention that the PCB layout can still be used if you do decide to use an L-C tuned circuit. This should be connected between the junction of C1/R1 and earth, with a DC-blocking capacitor of 10 nF connected in series — in the same way as shown in **Figure 2**.

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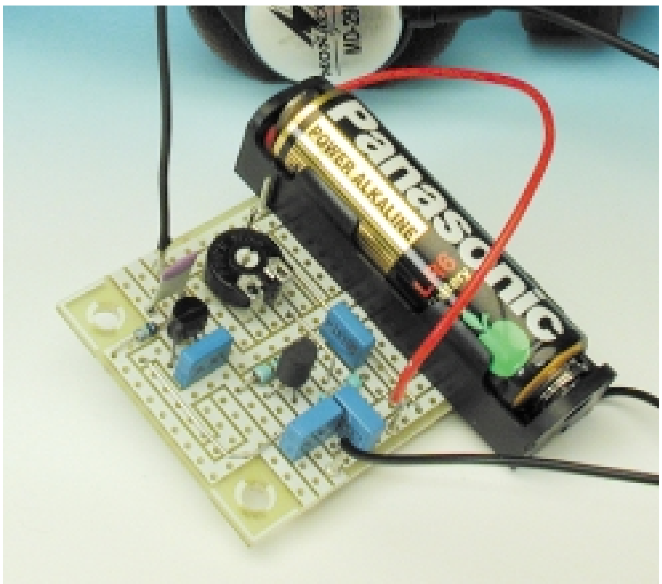

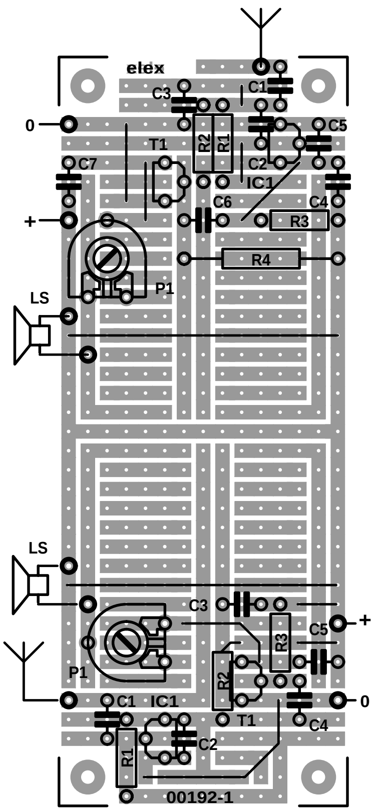

Figure 4. The circuit can be built in a matter of minutes on an UPBS-1 experimenters' board.

Experiments with Piezo Ceramics (1)

physics and electronic components

By B. Kainka

What do capacitors, electret microphones and passive infrared sensors have to do with one another? It is often interesting to go back to first principles and carry out a few experiments on the underlying phenomena. Electronics and physics belong together: and here we look at the physics behind some electronic components.

First, a simple experiment: take a ceramic capacitor and apply a high voltage. Now heat it briefly to 80 or 100 degrees Celsius, and let it cool slowly, keeping the voltage applied. The capacitor is now a completely new component with astonishing properties.

The voltage used must be on the limit of what the capacitor can stand: this will, depending on the capacitor, typically be anywhere from around 50 to over 500 V. Great care is required when using high voltages. **Figure 1** shows an isolated high-voltage supply, which allows a voltage in the desired range to be generated. Here a kind of adjustable transformer has been constructed from two transformers. A model railway transformer (without its rectifier) can also be used. A resistor limits the current to a safe value. The contacts must still nevertheless not be touched: and the voltage across the charged capacitor is not safe and can lead to painful skin burns.

A soldering iron can be used to heat the capacitor. It is sufficient to hold the heating element close under the capacitor. A few moments are

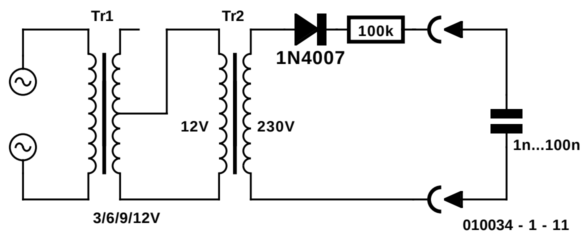

Figure 1. High voltage generator.

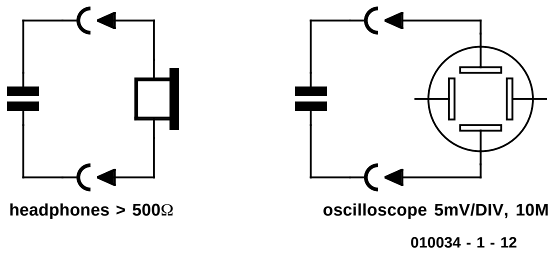

Figure 2. Testing the charge with headphones or an oscilloscope.

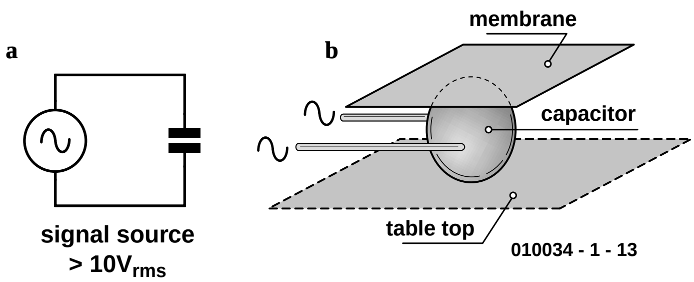

Figure 3. a. Use of the capacitor as an audio transducer.
b. Membrane arrangement.

enough, and then the capacitor can be left to cool.

A Phenomenon

After this special treatment the capacitor can be first shorted to discharge it. Then, high-impedance headphones can be used to check that it really is discharged fully. But now the interesting part: as soon as the capacitor is warmed in the fingers, it charges up a little. Connecting it to the most sensitive high-impedance headphones available, you can hear a quiet crackle. And then, when the capacitor is cooled, it charges again and you can hear it crackle again. This can be repeated as often as desired: each change in temperature leads to a small change in voltage.

Instead of headphones, an oscilloscope can be used to investigate this (**Figure 2**). With the test probes connected, a charge pulse of a few millivolts can be observed. On cooling, a pulse opposite in polarity to that seen on warming is observed.

Is this some innovative thermoelectric device or the long sought-after perpetual motion machine? Sadly not, because each change in temperature requires some energy. And a very small part of that energy is converted into electrical energy.

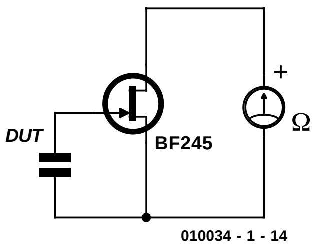

Figure 4. Static measurements with a FET.

There's more

With the aid of the oscilloscope we can make a further interesting observation. The prepared capacitor charges itself and then discharges itself again when placed under mechanical force. The force must be applied very suddenly for any effect to be visible. It is clear, for example, if the handle of a screwdriver is used to hit the capacitor. Have we made our own microphone? In principle the answer is yes, although its quality is not exactly up to studio standards.

It might well be supposed that the effect is reversible, i.e. that the capacitor also acts as a loudspeaker. So, let us apply an alternating voltage (**Figure 3a**). We can use a tone generator operating at around 1000 Hz or the output of an amplifier. The signal voltage should be as high as possible. With 10 V_{rms} to 50 V_{rms} we can get satisfactory results. To make the sound audible, we need also a membrane to act as a diaphragm. For this we can use a piece of card or tin. The capacitor under test can be clamped between the membrane and a firm surface (**Figure 3b**).

Static forces

If the capacitor is connected to the input of an extremely high-impedance amplifier, we can also detect static forces. A constant pressure produces a voltage which remains present as long as the capacitor is not discharged. This can be shown very simply using an ohmmeter and a JFET (**Figure 4**). The FET alters its resistance as its gate voltage changes. The small changes can be measured very easily using a digital multimeter. Temperature changes can also be registered in this way.

When these possibilities have been exhausted, various types of ceramic capacitor can be tried: tubular capacitors, plate capacitors or multi-layer capacitors. The latter are available, for example, as surface-mount devices. The individual layers are so thin that they cannot stand very high voltages. If the maximum voltage is exceeded, then breakdown occurs and the capacitor is no longer usable. In other cases, only the outer case will be damaged. Capacitors mistreated in this way must of course not be used afterwards: it is better to experiment with old components from the junk box. It is in any case interesting to find out which types of capacitor make the best loudspeak-

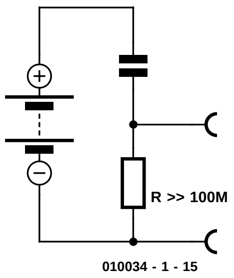

Figure 5. Principle of the condenser microphone.

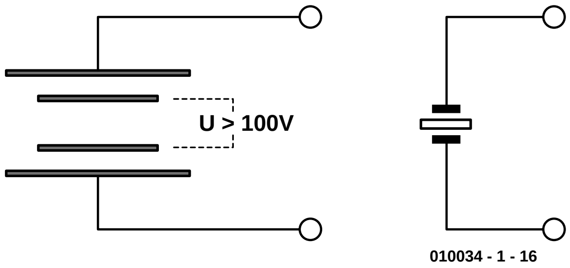

Figure 6. The internal charge of a polarised capacitor.

ers, microphones and pressure sensors.

Some physics

The capacitance, *C*, of a capacitor is equal to its charge, *Q*, divided by the voltage, *U*, across it.

$C = Q / U$

For example: a capacitor is charged for ten seconds with a constant current of 1 mA. The total electric charge is thus 10 mAs = 10 mC or 0.01 Coulomb. Fully charged, the voltage across the capacitor is measured at 10 V. The capacitance can now be determined:

$C = 0.01C / 10 V = 0.001 C/V = 0.001 F = 1000 \mu F$

Our ceramic capacitors naturally have rather smaller capacitances, such as 100 nF. That means that with a given charging current they reach the final voltage so much quicker, and that the stored charge is correspondingly smaller.

As is well known, a capacitor (or "condenser" as it used to be called in the old days) can be constructed from two metal plates separated by a vacuum or layer of air. Its capacitance is given by:

$$C = \varepsilon_0 \cdot \frac{A}{d}$$

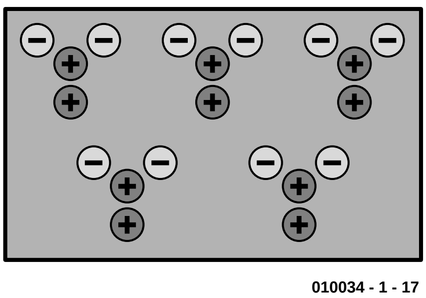

Figure 7. Model of a polarised dielectric.

where the **permittivity** of free space (or **electric constant**) $\varepsilon_0 = 8.8542 \cdot 10^{-12}$ F/m. The surface area *A* is measured in m^2, and the separation *d* in m. For a plate capacitor with an area $A = 100\ cm^2 = 0.01\ m^2$ and a separation d = 1 mm = 0.001 m this formula gives a capacitance of 88.5 pF.

The capacitance becomes smaller as the plate separation grows, and becomes larger as the plates are placed closer together. If the plate separation is increased on a charged and isolated capacitor, then the voltage across it increases. This may seem amazing, but it follows directly from rearranging the capacitance formula:

$U = Q / C$

The charge *Q* cannot change, because the connections are isolated. Hence the voltage must rise when the capacitance drops. This is the principle of operation of the condenser microphone in its basic form.

Capacitor = microphone

One of the two capacitor plates is the membrane. Mechanical movements are converted into electrical oscillations. For the microphone to work, the capacitor must be charged from a voltage source, typically of a few hundred volts. The microphone itself has a capacitance of only a few picofarads (**Figure 5**). The signal voltage must only be lightly loaded, and so a high-impedance connection is required. A high-impedance amplifier is best placed next to the microphone itself, and not at the end of a long cable.

Of course, the function of the condenser microphone is reversible. Electrical forces can thus be converted into movements of the membrane. This is the principle used in electrostatic loudspeakers.

When we compare our ceramic capacitor microphone with the condenser microphone we see only one difference: the lack of a voltage source. Or perhaps not: has something in our special treatment created some kind of battery? We must look at the whole thing as a capacitor within a capacitor, where the inner capacitor is permanently charged (**Figure 6**).

This process of charging the inner capacitor is called 'polarisation'. The charged material is now an 'electret'. There are various plastics and ceramics with which this can be done. Some materials, like quartz, already have this permanent electric field property without special treatment. A polarised capacitor, or electret system, can be represented with the circuit symbol for a quartz crystal.

Dielectric constants

Ceramic capacitors are frequently made from barium-titanate or similar ceramics. These materials have a particularly high dielectric constant. In comparison to an air-dielectric capacitor the capacitance can be as much as 10,000 times higher, i.e. ε_r = 10,000.

$$C = \varepsilon_0 \cdot \varepsilon_r \cdot \frac{A}{d}$$

Table 1 shows ε_r for certain important materials. Air, glass and epoxy resin all have relatively modest dielectric constants. But with water things are rather different, and its dielectric constant is conspicuously

Table 1

Dielectric constants for various substances

Substance	ε_r
Vacuum	1
Air	1.00059
Paper	2 - 2.5
Glass	2 - 12
Mica	4 - 8
Epoxy resin	3.6
Water	81
Ceramic substances	up to 10,000

high. What does water have that the others do not? Water is a polarized substance, meaning that the positive and negative charges in its molecules are not distributed in a uniform way. Each water molecule is a tiny charged capacitor. If they are aligned in an applied electric field, then together they create their own stronger field. Exactly the same thing happens, on a larger scale, in certain ceramic substances. When the internal charges are arranged at random, then the substance is not charged. This can be changed by the process of polarisation described above. The model of a polarised dielectric is shown in **Figure 7**.

Materials that can 'store' an electric field in this way are called **ferroelectric**. The idea is to indicate a comparison with the magnetic properties of iron. Iron contains magnetic dipoles whose orientation can be affected by an external magnetic field. If the magnetisation is strong enough, the internal field can be partly retained. **Remanence** (or residual magnetism) is a typical characteristic of iron and steel. Particularly good magnets can be made by magnetising the material at high temperature and then cooling it: exactly the same happens with ferroelectric materials in an electric field. The parallel goes further: each material has its own characteristic temperature beyond which the stored field is lost: this is called the **Curie point**. A piece of iron made to glow red will certainly lose its magnetisation.

The Curie point of the commonly used barium-titanate ceramics is around 55 °C. It must therefore be possible to return our polarised capacitor to its original condition. So, warm it briefly with a soldering iron (or even in hot coffee), and it soon will no longer work as a loudspeaker. If necessary, it can be used as if it were new.

Ferroelectric ceramics can in principle be polarised without the use of heat. For some materials it suffices to apply a high enough voltage to create the electret. The effect is not particularly distinctive with commonly available ceramic capacitors: hence our use of heat and applied voltage for best results.

The processes going on in the dielectric are relatively complicated. Sometimes the special properties are only noticeable in a limited temperature range. Ceramic capacitors with high values generally also have high temperature coefficients. Looking in more detail, we see that the capacitance has a maximum value at a certain temperature, which is not necessarily room temperature. Put simply, the capacitance changes with changes in temperature. The internal charge, however, remains constant. Hence the voltage across the capacitor must change, and exactly this effect was observed at the start of this article. Any temperature change changes the external charge on the capacitor.

In the case of an air-dielectric capacitor the plate separation must be halved to change the voltage across the capacitor by 50%. For a polarised ceramic capacitor, a change of rather less than 1 % in dielectric thickness has a similar effect. In such a hard material, this naturally has to do with the large forces that are at work. Inside the material charged crystal lattices shift against one another, leading to large changes in the electric field. At this level the effect operates differently. When a voltage is applied the material distorts as a result of the electrical forces. The possible movements are not great, but the forces are considerable. A ceramic loudspeaker does not have a great travel, but a relatively large diaphragm can be moved.

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In the second and final part of this article we focus on technical applications of piezo ceramics.

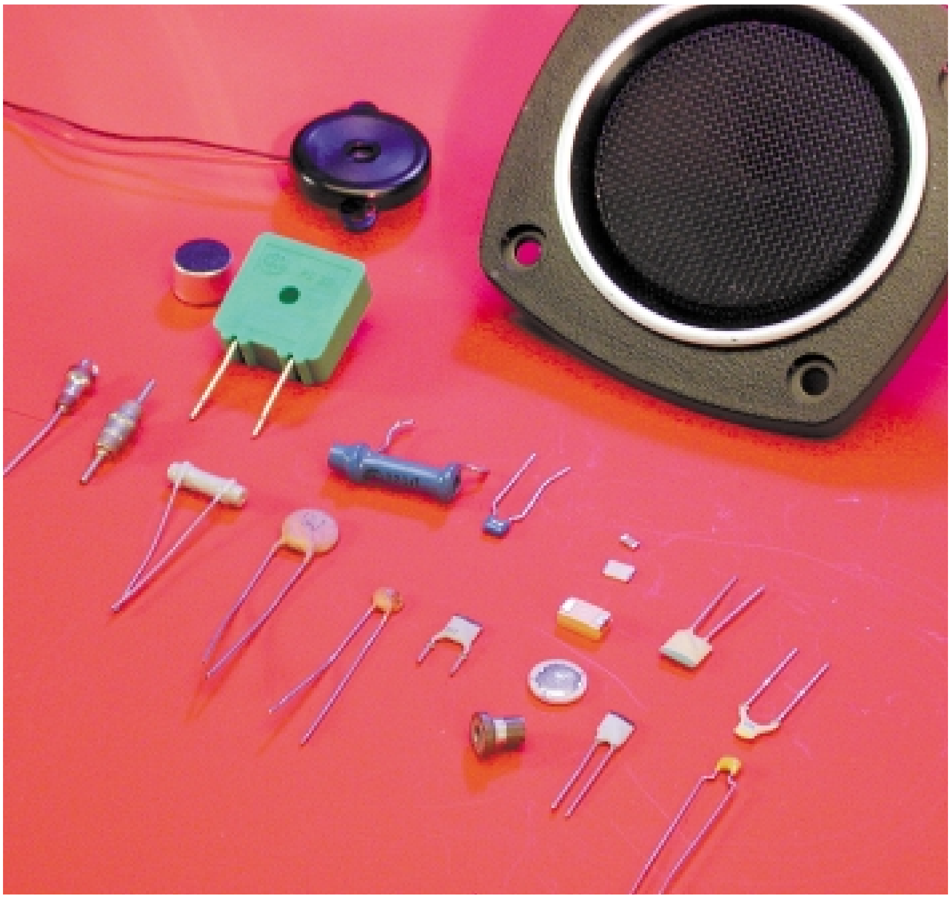

Figure 8. Some electronic components made from ceramic substances. The front row shows a selection of RF capacitors.

Model Remote Control

using an IR remote controller

If you thought that an infra-red (IR) controller was only good for zapping TV channels then read on. This novel circuit uses a micro controller to decode RC5 format IR signals and put them to a better use...

This compact circuit is designed to allow a model car, boat or robot etc. to be controlled from a handheld remote controller that would normally be used to control a TV, audio amplifier or satellite receiver. The only requirement is that the IR remote controller must send control messages using the RC5 format.

At the heart of the circuit is a COP8782 micro controller from National Semiconductor, this interprets the infra-red signals and sends out proportional signals to control a model servo for steering. It also has a MOSFET bridge circuit to drive a motor for forward, stop and reverse functions.

Infra-red receiver with µC support

The infra-red control signal is received by IC3. This device includes a high sensitivity photo diode, amplifier, filter and demodulator for picking up the 36 kHz modulated signals. The device specified can be either a SFH5110-36 from Siemens or a TSOP1736 from Temic. **Figure 1** shows the internal block diagram of this three pin device.

The complete receiver circuit diagram is shown in **Figure 2.** The demodulated signal from IC3 is connected to the G0 input of the micro controller IC1. The micro controller performs error detection and interprets the received RC5 format message. R8 and C4 form a low pass filter to de-couple any interference on the supply to IC3. IC2 produces The 5 V supply to the circuit and requires an input voltage of 9.6 V (equivalent to 8 nicads). IC2 can be replaced by a low-drop regulator equivalent and this would allow operation from a 7-cell nicad pack.

The microcontroller used here should be no stranger to regular readers of *Elektor Electronics*, it has been used many times in the past, just to re-cap its main features:

– 4096 x 8 OTP EPROM
– 128 Bytes RAM
– 1 µs cycle time at 10 MHz
– 16-bit-Timer with the operational modes:
  - Timer with auto-reload
  - Timer as External Event Counter
  - Timer with Capture function
– 16 I/O connections, 14 of these can be configured as outputs or inputs.
– Selectable output pin configuration: Tri-State, push-pull or pull-up
– Microwire interface
– Interrupt sources:
  - External Interrupt, selectable on rising or falling edge
  - Timer Interrupt
  - Software Interrupt

For the (steering) output signal to the servo the internal timer is used in its *Timer with Auto-Reload* mode. Once this mode has been programmed during reset initialisation of the chip, the timer will output a defined pulse repetition on pin G3 without any further software intervention. The output signal required to control the majority of servo types should be a pulse of between 0.9 and 2.1 ms with a repetition rate of around 20 ms. To position the servo mid way in its travel a pulse width of 1.5 ms is necessary whereas 1.00 ms will move it to its minimum position and 2.0 ms to its maximum position. This is the so called **pulse**

width modulation (PWM) method of proportional control.

In the last few years all servos sold in the U.K. irrespective of manufacturer have identical pin configurations so there should only be a problem with the servo connector if you are using older servos. The signal names of the servo output pins are printed on the PCB so it is possible to work out the correct connection if you are using these older type servos (refer to the pin layouts used by the servo manufacturers in **Figure 3**). It is important to wire the servo correctly, you risk damaging the servo and this circuit if you mix up the wires. The layout of the servo connector on the PCB will allow a 1-to-1 connection to all modern servos.

Output pins L6 and L7 are used to control the models drive motor. These are used to switch Logic-Level Power MOSFETs connected in a bridge configuration. This type of MOSFET can be driven directly from a 5 V logic signal. The types used here have an $R_{ds(on)}$ of about 0.1 Ω and can easily handle a motor current of 4 A. If you intend switching higher currents then it will be necessary to fit a heat sink or replace the MOSFETs with types having a lower $R_{ds(on)}$. The gates of these devices are connected to ground via 100 kΩ resistors R3 and R4, these ensure that the MOSFETs remain switched off when the micro controllers output pins are briefly tri-stated during initialisation of the micro controller. Diodes D3 to D6 are used to protect the MOSFETs from reverse voltage spikes generated by the motor.

A reset signal is produced by the network formed by R1, C1 and D1. LED D2 flashes each time a RC5 coded message is successfully received. This gives a good indication that the remote control link is working correctly.

IR signals in RC5 format

The RC5 code was developed by Philips and is currently the most widely used standard for European entertainment equipment. All RC5 compatible remote controllers must conform to the RC5 message protocol standard. Each message is composed of 14 bits and is made up of the following fields:

– 2 Start bits to allow the AGC in the receiver to adjust its level.
– 1 control bit indicating the start of a new message.
– 5 bits indicating the system address. (MSB - LSB).
– 6 bits containing the message command. (MSB - LSB).

Features

Controlled by an RC5 infra-red signal	
Connections:	Model servo
	Model drive motor
Motor control:	Forward/Stop/Reverse
Operating range:	approx. 10 m

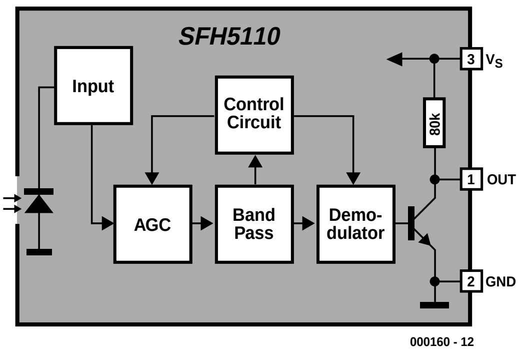

Figure 1. Block diagram of the IR receiver chip.

To reduce the intermodulation effects produced when the IR signal is mixed with other sources of IR interference (e.g. incandescent lamps) the message is sent using biphase encoding. With this method the data signal changes state mid-bit, a falling edge indicates that the data bit is a zero, a rising edge indicates a logical one. The data signal will also change state at the end of a data bit if the next bit has the same value as the present bit (see **Figure 4**).

The time taken to send one complete message is 113.778 ms which is made up of 24.889 ms for the message information followed by a gap of 88.889 ms. The carrier frequency is 36 kHz. A single control bit (toggle bit) in the message is used to tell the difference between pressing a key again and holding a key down, it changes its value (or toggles) only when a key is first pressed. This feature often creates problems for the re-programmable type of remote controller that 'learns' the control sequences from an existing controller.

Five bits following the toggle bit form the address field and are used to identify the type of equipment. This ensures that the remote controller for the stereo does not control the TV. The TV simply ignores messages that are

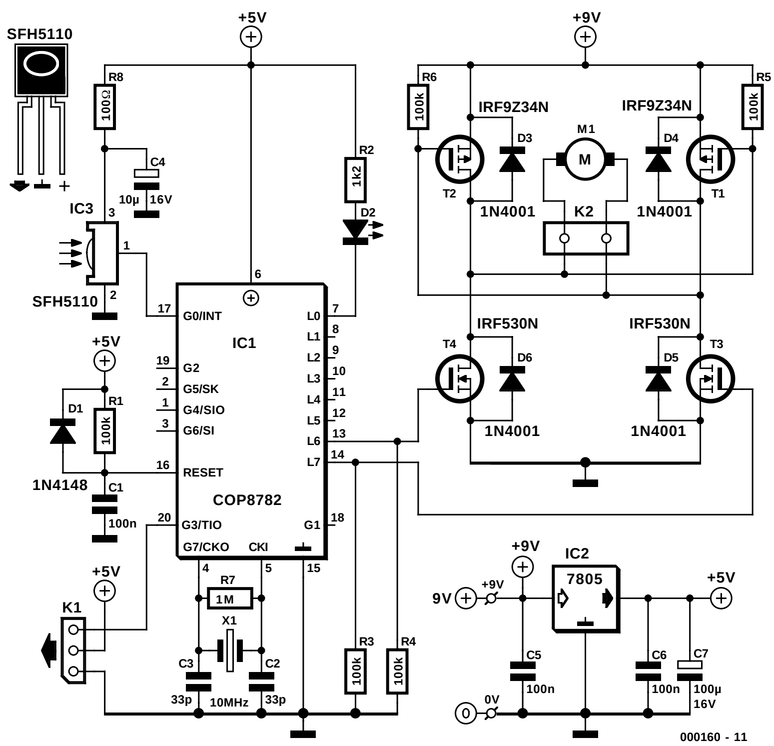

Figure 2. The complete circuit with IR receiver, Microcontroller and motor driving bridge.

addressed to the stereo and vice versa. This is similar to the operation of the address bus of a processor.

The next six bits are used for sending commands to the equipment. For example to increase sound volume of the equipment (Master Volume+) command 16 would be sent. This command field would be the same irrespective of the type of equipment being controlled. This allows equipment to be controlled by a remote controller device from a different manufacturer. An article in the March and April 2001 editions of *Elektor Electronics* provides detailed information of the codes used for this and other IR communication standards.

Input G4 (pin 1) of the controller chip is used to alter the way in which the controller handles the RC5 messages. If the G4 input is hard-wired to ground this controller circuit will only respond to commands sent out by a remote controller for a stereo amplifier (address 17). Leaving G4 open–circuited (it has an internal pull-up resistor) ensures that the controller responds to any RC5 compatible remote controller, it simply ignores the address field in the message.

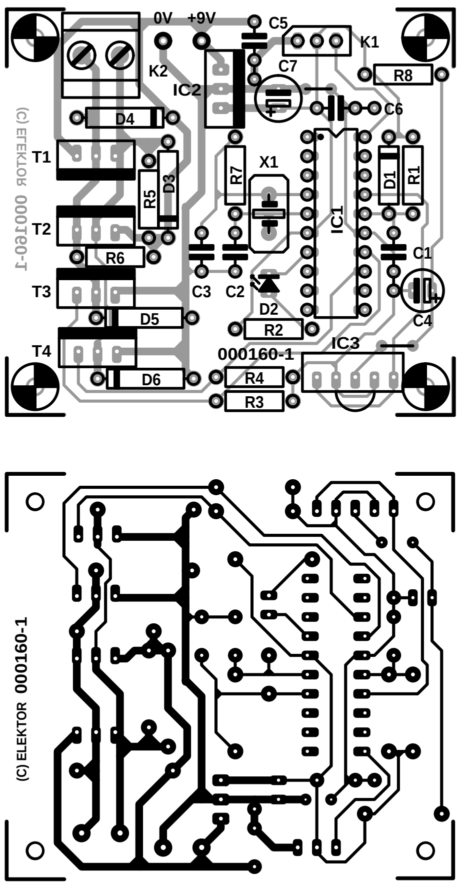

Figure 5. Layout and component placement of the double-sided PCB.

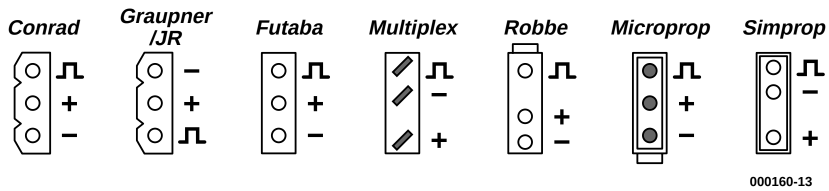

Figure 3. Pin assignments for different makes of servo. *(view from underside)*

Construction and use

The double-sided PCB is shown in **Figure 5**. It is small enough to fit easily inside a model car or robot. Fitting the components should not pose too much of a problem providing that you are careful and observe the correct polarity of the important devices. After a careful visual check of your soldering handiwork you can apply power to the circuit (initially with no output load connected to the circuit) and then fit the board into the model.

When mounting the infra-red receiver in the model make sure that you position it so that it can pick-up the transmitted IR signals from any direction, mounting it on the top of the model will probably be best, preferably on a reflective surface to optimise the received IR signal.

Unlike conventional radio model remote control equipment it is only possible to issue one command at a time, pressing two keys at once will have no effect. This control method is a little awkward at first and takes some getting used to but after a little practice it should be possible to gain smooth control.

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Model control functions with the IR remote controller.

VOL- key	Servo left
VOL+ key	Servo right
key 3	Motor reverse
key 2	Motor forward
key 1	Motor stop

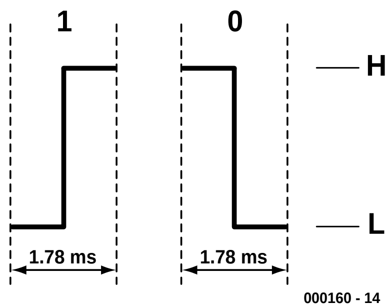


Figure 4.biphase encoding.

COMPONENTS LIST

Resistors:
R1,R3-R6 = 100kΩ
R2 = 1kΩ2
R7 = 1MΩ
R8 = 100Ω

Capacitors:
C1,C5,C6 = 100nF
C2,C3 = 33pF
C4 = 10µF 16V radial
C7 = 100µF 16V radial

Semiconductors:
D1 = 1N4148
D2 = LED, red, high efficiency
D3-D6 = 1N4001
T3,T4 = IRF530N (Conrad Electronics # 158747,Farnell #637-439)
T1,T2 = IRF9Z34N (Conrad Electronics # 159166, Farnell # 934-677)
IC1 = COP8782, programmed, order code **000160-41**
IC2 = 7805
IC3 = TSOP1736 (Conrad Electronics # 171069), SFH5110-36 (Siemens), PIC26043SM (Farnell # 139-877), IS1U60 (Sharp) or TFMS5360, see text

Miscellaneous:
X1 = 10MHz quartz crystal
K1 = 3-way pinheader
PC1,PC2 = solder pins
K2 = 2-way PCB terminal block, lead pitch 5mm
PCB, order code **000160-1**
Disk, contains source and hex files, order code **000160-11**

CPU Overclocking

go faster at no extra cost

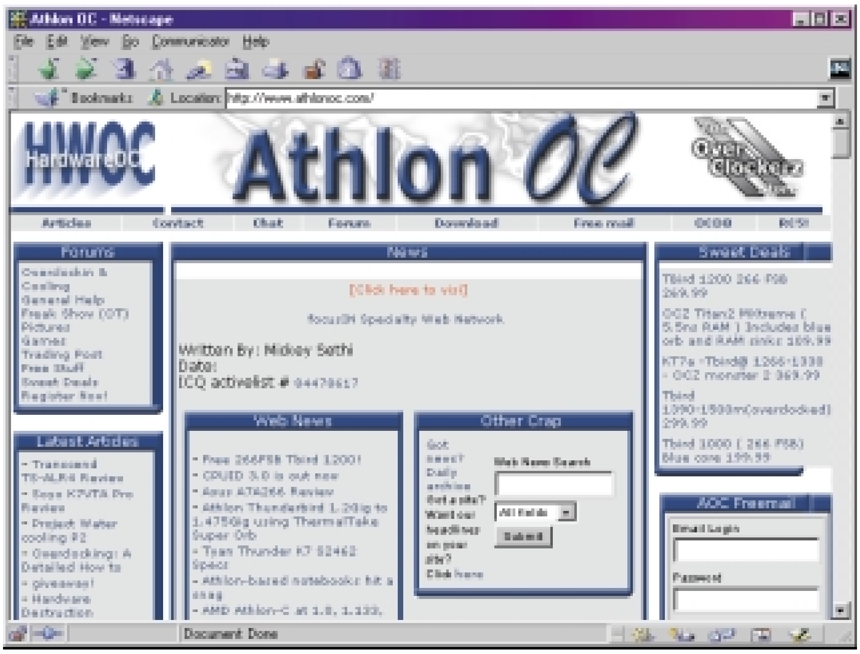
By Harry Baggen

With clock rates of 500 MHz and more, it would appear that modern CPUs have enough processing power for all imaginable applications. However, computer users always want everything to go even faster. With a few simple measures, you can get certain types of processors to run at higher clock rates without spending a single extra penny.

Overclocking is a sort of game, in which the objective is to get a processor that has been specified for a particular clock rate to run at a higher clock rate. This is frequently successful if the supply voltage for the core is modified and extra cooling is provided. After all, most processors are 'spec'ed' by their manufacturers such that there is still a reasonable amount of performance margin (at least 10% on average).

Manufacturers of CPUs are not happy when users overclock their products, so they use various measures to try to protect their processors against overclocking. Intel processors are fairly well protected, but the protection of AMD processors is fortunately somewhat simpler. In particular, the recent Athlon and Duron processors in the Socket-A version can easily be 'boosted'. This is because there are several jumpers on the ceramic package that determine the configuration. Anyone who wants to overclock his or her processor can quite simply cut through some of these jumpers or reconnect some of them (using a sharp pencil) in order to configure the processor for a different multiplication ratio or core voltage. An additional factor is that the new AMD CPUs have proven to be true speed demons, which can easily run 100 or 200 MHz faster than their normal clock rates.

Motherboard manufacturers have anticipated this trend, and the latest motherboards often offer various handy settings in the BIOS. These allow the values of factors such as the core voltage, frontside bus frequency and multiplier to be easily set without even opening up the enclosure. Most of the modern generations of motherboards with a VIA KT133 or KT133A chipset have such setting options in the BIOS, which makes overclocking easy.

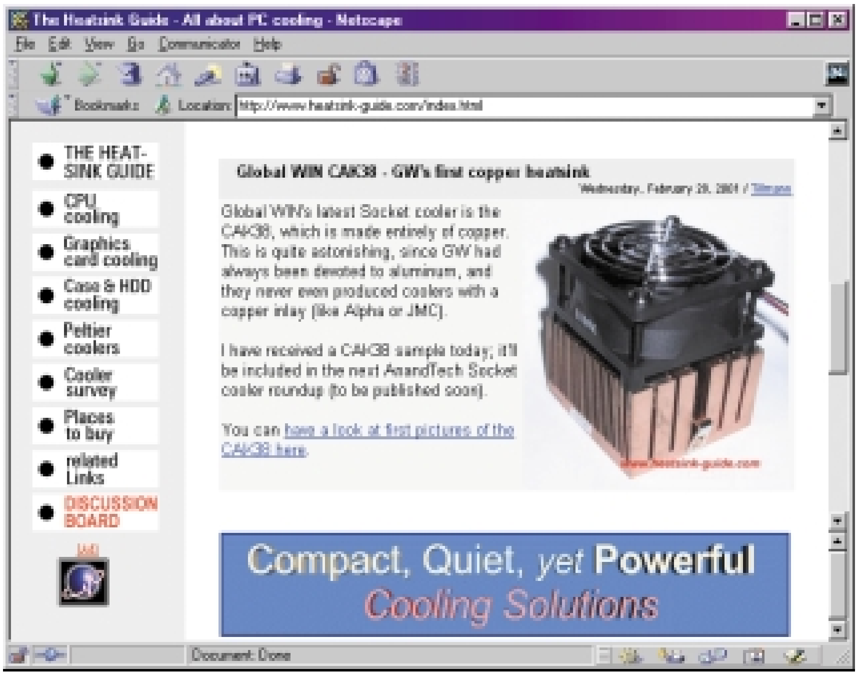


There are many sites on the Internet that are exclusively dedicated to the subject of overclocking.

The **Athlon OC** site [1] offers a series of interesting articles about overclocking and cooling Athlon and Duron CPUs. This site also has information on subjects such as water cooling and modifying the motherboard (in order to allow the core voltage to be increased). There is also a chat section, a forum section and a downloads section.

Overclockers.com [2] is also completely focussed on this subject. It has a collection of more than 700 entries related to overclocking, accessories, testing and so on. This site is of somewhat more general interest than Athlon OC, and you will also find articles about boosting graphics cards here.

Tweaktown [3] is preoccupied with a whole range of different measures



for 'souping up' a computer system. Tests of motherboards, graphics cards and the latest drivers, and of course all sorts of 'tweak' articles, can tempt even the most conservative computer owner to start tinkering with his or her system.

Overclocked Inside [4] is a German overclockers' site with a large amount of practical information. An especially good feature of this site is that you can obtain a graphic representation of the jumper arrangements that are required with an Athlon or Duron processor for a particular core voltage and frequency. Another nice feature is the test of all sorts of CPU coolers, with the possibility of listening to the fan noises for yourself.

Incidentally, large heatsinks with hefty fans are a necessary evil in the world of overclocking. Athlon and Duron processors are well known for their appetites for ampères and their need for good cooling assemblies, and overclocking simply aggravates the situation. If cooling is neglected, the silicon core will quickly die from overheating.

There are sites that have completely specialised in heatsinks, fans and thermal pastes. One of these is **The Heatsink Guide** [5], which is totally specialised in this subject. Here you will find information about cooling not only CPUs, but also graphic cards, hard disk drives and the computer enclosure itself.

2CoolTek [6] is actually a company that sells heatsinks and fans, but this site also has a very good comparison of many different types of heatsinks, including a large table that precisely reports which temperatures were measured with the tested coolers.

Would you like to try it for yourself? Cost should not present any real obstacle. A Duron will only set you back around 60 pounds, and once you have it you can experiment to your heart's content. Naturally, you must first have a motherboard that supports an Athlon or Duron processor (Socket A).

Finally, we should mention a number of overclocking FAQ's (Frequently Asked Questions) that can be found on the Net. Here you can read about the other people's experiences with overclocking. We have noted five interesting **FAQ lists** in our links summary [7].

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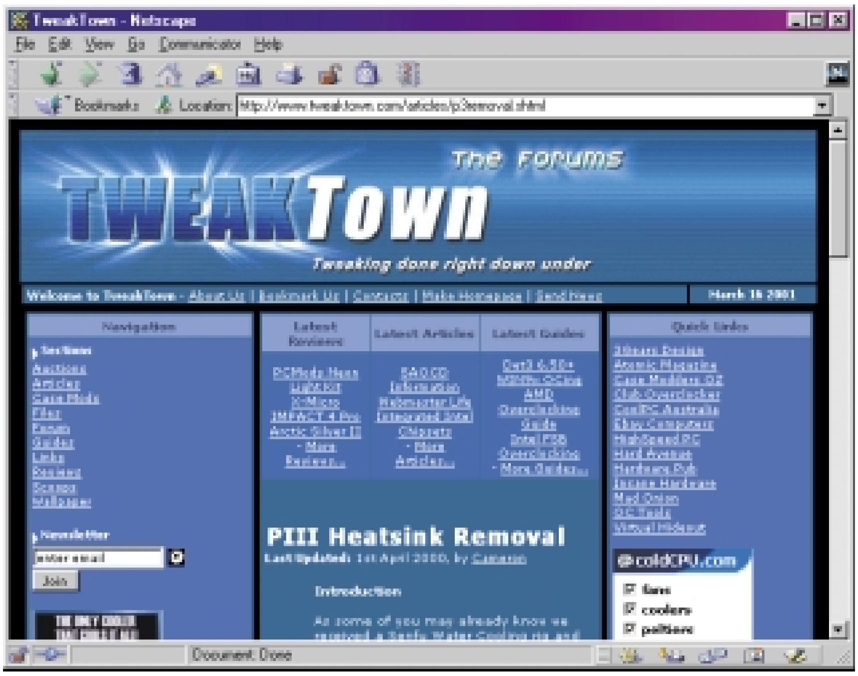

Website url's

(also available as hyperlinks at www.elektor-electronics.co.uk)

[1] Athlon OC:
http://www.athlonoc.com/

[2] Overclockers.com:
http://www.overclockers.com/

[3] Tweaktown:
http://www.tweaktown.com/

[4] Overclocked Inside:
http://www.ocinside.de/index_e.html

[5] The Heatsink Guide:
http://www.heatsink-guide.com/index.html

[6] 2Cooltek Socket heatsink test:
http://www.2cooltek.com/socket_test.html

[7] FAQ's:
Athlon overclocking FAQ:
http://www.tech-report.com/faq/athlon.x

Ninja Micro's Overclocking Forum:
http://www.ninjamicros.com/cgi-bin/Ultimate.cgi?action=intro

Hyperformance PC Overclocking FAQ:
http://www.hyperformance-pc.com/overclocking_faq.htm

Icrontic forums:
http://www.apushardware.com/vb/index.php

AMD-MB faqs:
http://www.amdmb.com/faqs.html