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March 2005
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Valves or Transistors?

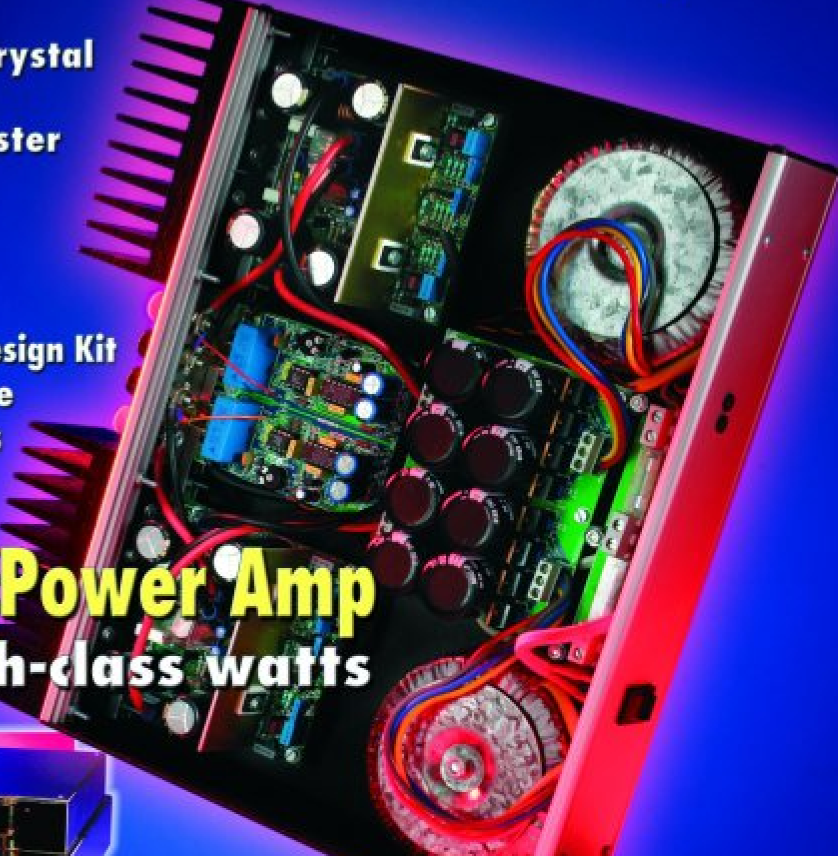
colouring vs. reality
nostalgia vs. progress
advocate vs. analyst

Build this:

- Programmable Crystal Oscillator
- Handy Opamp Tester
- Play Solitaire

Background:

- Altium FPGA Live Design Kit
- Introducing ARMee
- Anti-Noise Systems



High-End Power Amp

2 x 40 high-class watts

EL156
Valve Monoblock

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MEETING

Try to picture for yourself the bi-weekly general editorial meeting of this wonderful magazine, with its inherently international character. Everyone is there: France, England, Germany and the Netherlands, all seated around a large rectangular table structure. And in no particular order, of course. The lab is also present in full strength. Everyone at the table has at least a masters in engineering. And then there's me, with a degree in 17th-century Dutch Literature from the University of Nijmegen (1978), chairing the meeting. I hardly need say that the risk of a Babylonian confusion of tongues is all too real. And I can assure you that there are quite a few different 'right answers' to electronic questions. Of course, I try to avoid all of this by holding the floor myself as much as possible. After all, I'm the boss. But you can't keep the tension up all the time, so I like to relax the atmosphere by raising the discussion to a metaphysical level. For example, I might ask, 'Is there really any difference between transistors and valves?', and then pour myself a cup of coffee – since nobody else does that for me, even though I'm the International Editor. And of course I know perfectly well that I'm not supposed to ask such questions, since the editorial staff is officially neutral. Out of the corner of my eye I see one of the designers involuntarily cross his arms: body language for 'I'm taking a stand'. On the other side, 'ze outprint' of a text generated by one of our worthy employees is being vigorously shoved back and forth: body language for 'we don't agree on this'. Every once in a while I have to nudge things along a bit: 'Well, just say what you think!' (and hey presto see pages 25–27 in this very issue).
Sometimes I manage to finish a whole cup of coffee before making another meaningful contribution to the discussion. 'What matters here isn't technology or sound, but music. I still have the old vinyl LPs from Skip James and Jimmy Reed that I played on my very first turntable, when I had to put a heavy coin on the arm to get any sort of sound. And my son uses an MP3 player to listen to his music.'
I can feel a shudder pass through the ranks of the audiophiles at the mention of the word 'MP3'. It reminds me of the official introduction of the CompactDisc (CD – *Ed.*), when a few fellow journalists asked the Philips press spokesman how Philips had the nerve to be responsible for eliminating everything below 10 Hertz and above 20,000 Hertz from the musical experience of audio enthusiasts. And they got an answer!
In the resulting discussion, someone is bound to come up with the idea of seeing where things stand with the development of techniques to combat noise with sound.
As I always say, a magazine like this puts itself together.

Mat Heffels, International Editor

elektor electronics — leading the way

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leading the way

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Noise has a serious impact on the environment as well as on human beings. Besides classical sound reduction techniques such as insulation, researchers are trying to attack the root of the problem using electronic means. They have already had some success.

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Altium FPGA LiveDesign Kit

Modern circuits make ever more extensive use of FPGAs. Special software is required when designing with these devices. One of the companies that produces such software is Altium, who offer a cheap evaluation kit. Once the evaluation period has expired, the kit is still very useful as a development platform for your own designs!

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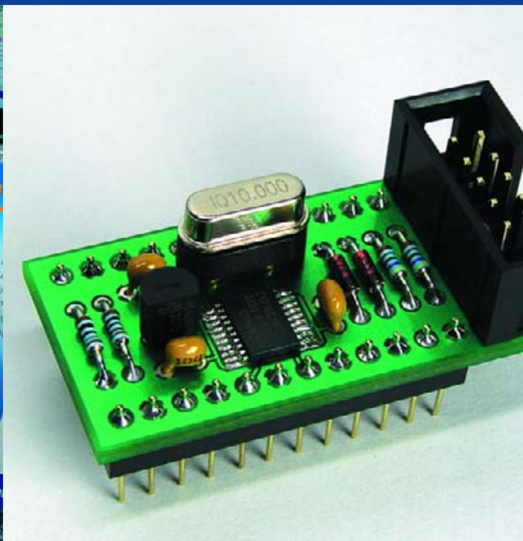


Serially Programmable Crystal Oscillator

The right crystal controlled frequency at the right time: With this DIP outline oscillator there's no need to wait for a special crystal to be made up. You can program its frequency precisely using the RS232 port of your PC. Low phase noise is a feature that makes this PLL oscillator suitable for demanding RF applications also.

Construction Projects

web technology allowing...
our completely restyled website should be online by mid-February 2005.
New features include Forum, Marketplace, News Service, Online Shop, secure payment options, article downloads…. check it out!

Elektor on CD-ROM Volume 2004 now available

Swiss Army Knife update

Jim Spence kindly informed that a software update is available for his Swiss Army Knife project (Elektor Electronics September 2004).
There is a new version of TCB (TB-2-K), which enhances the existing TCB as follows:

New comment symbol, allows source code to be commented without being stored.
Better error handling, indication line number where error occurred.
Hex - handling of hex numbers improved, numbers can now be either decimal or hex without changing the operating mode. Example, a=&FE.
LOADB - massive time saving feature, LOADB will now load a TCB program with or without line numbers.
I2C - Full support for I^{2}C devices using software, no special processor needed, new I^{2}C commands: CPOKE, CPEEK, CPUT, CGET, CREAD & CWRITE.
24LC256 - EEPROM support (LOAD & SAVE), enables programs to be saved and loaded thus eliminating the need for a battery backed RAM.
RUN - extended to allow running programs in RAM space, when called from ROM space

There is also a version that will work on the Elektor Electronics 'Flash Microcontroller Starter Kit' — the same as above but with automatic RAM detection switched off.

The updates may be obtained from our website, see September 2004 items.

TV Commercials Killer

Dear Editor — your article on the 'TV Commercials Killer' (July/August 2004) was a very practical application. The problem with it, is that in my country, South Africa, the logo is usually at the bottom section of the screen. Is there any way of making it work for me? Maybe some hints on software modification. Thank you for such a great magazine.
Charl Roux (South Africa)

The same question has been asked by a number of other readers. As stated at the end of the article, the project was not tested or post-engineered by our in-house design staff. The author's contact details may be found at the end of the parts list on page 27. Hopefully Mr. Schulze is able to help.

Clarity Class-T amplifier

Dear Jan — as you know we are selling kits of the Clarity Class-T Amplifier project (June, September, October 2004; *Ed.*).
Unfortunately we ran into supply problems with the STW38NB20 component, ST having advised us that it is no longer manufactured and available to the retail trade. Can you suggest an alternative?
Peter Geist
(Geist Electronic, Germany)

Ton Giesberts, our audio designer replies: a possible alternative is IRF's type IRF41N15D, however, it is housed in a TO220AB case! A quick comparison shows:

Parameter	IRF	ST	Unit
I_D	41	38	A
$R_{DS(ON)}$	0.045	0.065	Ω (max.)
P_{tot}	200	180	W
C_{iss}	2520	2800	pF (typ.)
T_j	175	150	°C
$R_{th(jc)}$	0.75	0.69	°C/W
Q_G	72	70	nC
U_{DS}	150	200	V

The lower value for U_{DS} is not a problem, the supply voltage being smaller that 120 V_{pp}. Tripath also recommend this transistor for use with their TA3020. The replacement has not been tested yet in the Elektor labs.

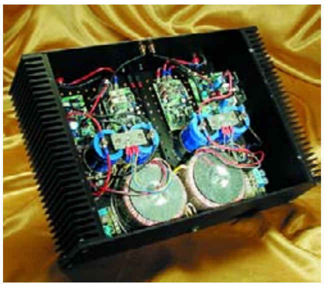
Parallel PIC Programmer

Dear Editor — I would like to inform you about a PIC programmer that's controlled over the parallel printer port and is compatible with Win2000 and XP. It is found at www.oshonsoft.com. The signal wires used can be selected by the user.
H. Sprenger (Germany)

Thanks for the tip!

Crescendo ME and electrolytics in storage

Dear Editor — it is my intention to connect an Elektor Crescendo Millennium Edition ('ME') amplifier into a loudspeaker system with an impedance of about 2.5 ohms. Can you please



advise if that is possible without alterations to the amplifier electronics?
Also, I have available a quantity of Philips 10 mF/63 V electrolytic capacitors that have been in storage for about 10 years (I think!). Can I use these components; it is possible to regenerate them, or should I bin them? Is there a way to extract the manufacturing date from the print on these devices?
C. Krampe-Zadler (Germany)

In theory, the lower impedance is just about acceptable if you assume that the amplifier does not supply more than 30 V at 2.5 Ω (i.e., 12 amps maximum). It is therefore recommended to employ a mains transformer with a slightly lower secondary voltage.
Regarding your second question, having been in storage for such a long time, the electrolytics are probably in need of being regenerated. This is best done by applying the maximum device voltage (63 volts) via a 1-megohm resistor for some time, observing all relevant safety precautions. In all probability, these devices will be usable. The manufacturing date can be gleaned from the date code (DIN 41314 / IEC 60062), as follows:

Year Code	Month Code
1995 = F	January = 1
1996 = H	February = 2
1997 = J	March = 3
1998 = K	April = 4
1999 = L	May = 5
2000 = M	June = 6
2001 = N	July = 7
2002 = P	August = 8
2003 = R	September = 9
2004 = S	October = 10
2005 = T	November = 11
2006 = U	December = 12
2007 = V	
2008 = W	

For example, a capacitor marked 'L6' was manufactured in June 1999.

OP-77 Replacement

Dear Jan — I'm sorry the project I need to ask assistance with is a bit aged ('Compact AF Power Amplifier', *EE* May 1997). The problem is that the guy at my local electronics shop tells me the OP-77 opamp is no longer available, suggesting the LM301AP as an alternative. Can you confirm this, please, or are there better alternatives? I also wonder if I can use the value 330 Ω for R8 and R9, instead of the specified 340 Ω which I am unable to obtain at such small quantities.
E. Oeynhausen (Germany)

The OP-77 may be replaced by any low-offset opamp, which positively excludes the LM301P! Also, be sure to select a type that's suitable for operation at a supply voltage of ±18 V. An excellent replacement for the venerable OP-77 is the OP177GP from Analog Devices, which is no coincidence as it is actually an improved OP-77. Another suggestion is the OPA177GP from Texas Instruments (listed by, among others, Farnell).

If at all possible you should stick to the specified 340 ohms for R8 and R9. With due regard to tolerance issues, the required value may be obtained by parallelling resistors from the E24 series, or in really acute cases, the E12 series (do check the end result with an accurate ohmmeter); for example, 360 Ω//6kΩ2 or 390 Ω//2kΩ7.

Cleaning the Mains

Dear Editor — I am looking for a circuit diagram of a good mains filter. Did you ever publish such a design and if so, when?
Jan Verbeeck (Netherlands)

It may not be obvious from the article title, but an excellent mains filter for home construction is included in 'Noise Suppression Filters' published in the November 2004 issue of *Elektor Electronics*. This article was the concluding part of our ClariTy 2×300-watt audio amplifier. Mains filters must be built, installed and used in compliance with relevant safety guidelines.

USB/I²C Interface (1)

Dear Jan — Having read your article in the December 2004 issue, I can not find conclusive evidence that the circuit, apart from its normal function, is also suitable for reading data on the I²C bus, without omitting the P82B715PN component.

My uncertainty is probably caused by the circuit symbol used for the component in the various drawings, and the fact that the examples only mention driving the outputs.
Manuel Friedmann (Germany)

Even if the circuit diagram does not show it explicitly, the P82B715PN is a bidirectional component (see www.semiconductors.philips.com/pip/P82B715PN.html). Consequently, nothing changes when compared to the circuit without the P82B715PN. Incidentally, controlling devices over the I²C bus always requires a bidirectional interface.
Also see 'Corrections & Updates', February 2005.

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– When replying to Mailbox correspondence, please quote Issue number.
– Please send your MailBox correspondence to:
editor@elektor-electronics.co.uk or Elektor Electronics, The Editor, P.O. Box 190, Tunbridge Wells TN5 7WY, England.

HIGH-END POWER AMP

High-class watts

Ton Giesberts

The High-End Preamp preamplifier published in the April & May 2004 issues naturally calls for a matching final amplifier in an identical housing. Due to the restricted size of the enclosure, the output power of this final amplifier is somewhat on the modest side, but the quality is outstanding – and it is more than adequate for filling an average living room with an impressive sound image.

The Audio Link amplifier, published a year ago, was well received by audio enthusiasts due to its very convenient user interface and high-quality, no-nonsense signal processing A matching final amplifier must naturally have the same level of quality, and here we have made an extra effort to achieve this result. Our desire to use an identical enclosure meant that there wasn't all that much space available, and the maximum output power of a final amplifier largely depends on its size. The dimensions of the heat sinks and transformers are particularly restricted by the size of the enclosure. We managed to achieve a bit less than 50 watts into 4 ohms, using 80-VA toroidal transformers.

This amplifier is thus primarily suitable for people who like to listen to music at a 'civilised' volume and are very demanding with regard to reproduction quality.

As the output power is distinctly less than what many of our previous (discrete) final amplifiers can provide, we added an accurate overdrive indicator to this design. It compares the input and output signals of the final amplifier and lights up an LED if the difference is too large. This enables the listener to keep a good eye on the output drive level.

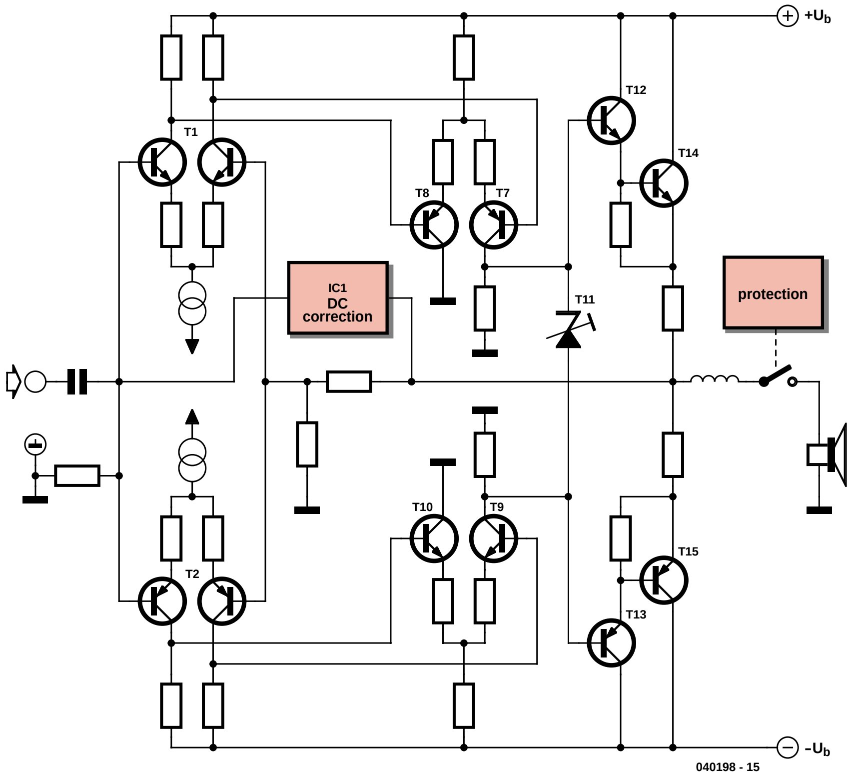


Figure 1. Block diagram of the High-End Power Amplifier, which is based on previous Elektor Electronics designs.

Output stage

Readers who have paid attention the schematic diagrams of our final amplifiers during recent years will see a lot of familiar features in the present design. There aren't any real surprises, and the sound quality is determined more by the choice of transistors and the circuit board layout than by the circuit design.

As you can see from the block diagram in **Figure 1**, each channel of the amplifier essentially consists of two differential amplifiers and an emitter follower, but with a fully symmetrical design. There is also an integrator that eliminates any DC offset voltage at the output.

The input stage differential amplifier (**Figure 2**) consists of two complementary dual transistors (T1a/T1b and T2a/T2b) from Toshiba. It is followed by a second set of differential amplifiers (T7/T8 and T9/T10), which provide most of the gain. The combination of T7 and T9 also forms a push-pull amplifier that can supply more than enough current to drive the emitter-follower output stage. The operating points of these differential amplifiers are determined entirely by the operating points of T1 and T2. The currents through the differential transistors in the second stage of the amplifier are determined by R20–R22 or R23–R25 and the voltage across R7/R8 or R10/R11, respectively. The voltages across the collector resistors of the input pairs are in turn determined by constant-current sources T3 and T4.

As can clearly be seen, all of the operating points in the amplifier are directly dependent on these two current sources, so we gave particular attention to their design. The reference voltages for the current sources are provided by flat LEDs, which in turn are operated at reasonably constant currents provided by a pair of simple JFET current sources. As a result, the current sources built around T3 and T4 are practically immune to power-supply ripple and supply voltage fluctuations. Unfortunately, JFETs have a rather large tolerance range, so you should measure the voltages at the points shown on the schematic diagram and compare them with the indicated values. A deviation of more than 20% is not acceptable.

The power output stages are wired as emitter followers using a standard design with two complementary sets of transistors in Darlington configuration (T12/T14 and T13/T15). Resistors R26 and R27 are placed at the inputs of the emitter followers to linearise the input impedance of the current amplification stage and reduce the effects of various parasitic capacitances. Although this reduces the open-loop gain, it also considerably increases the bandwidth. This allows less negative feedback to be used, which from an audiophile perspective can be regarded as better for reproduction quality.

T11 is used to set the quiescent current. Here we use the same type of transistor as the NPN driver (T12). This makes assembly easier because these transistors have a plastic package that can be fitted to the heat sink without using an insulator, which is also true of T13. The quiescent current bias transistor is located on the circuit board between the two output-stage transistors (T14 and T15), which do have to be fitted with mica insulating washers, in order to achieve maximum thermal coupling. Driver transistors T12 and T13 are located on either side of this set of three transistors. The quiescent current of the output transistors is set to 100 mA, which is sufficient.

Integrator IC1 compensates for the input bias current, as well as any off-

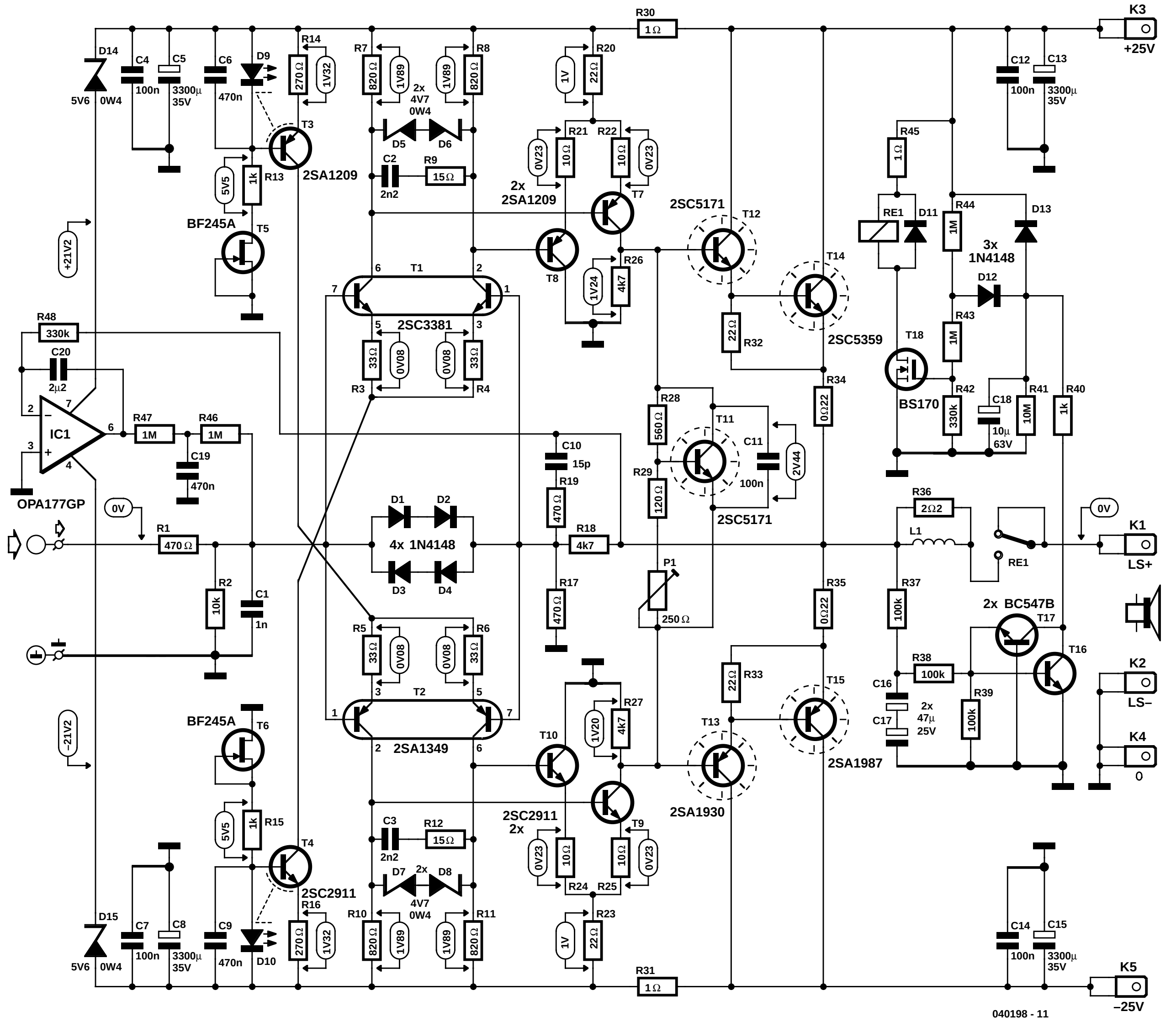


Figure 2. Careful component selection enables this design to achieve significantly better results than previous models.

sets that may arise due to inequalities in the various DC settings, by injecting a current into the input of the amplifier. The bias current is primarily due to differences in the amplification factors of the complementary dual transistors and the currents from the two current sources, as well as imbalances in the transistors of the following stages (differences in amplification factors and base–emitter voltages). A bias current of approximately 8 µA can be compensated using the indicated component values. R48 and C20 determine the integration time constant, and C19 decouples any effects from IC1 (except for bias current correction).

R1 and C1 at the input of the amplifier form a low-pass filter that prevents the amplifier from being overdriven if excessively steep signal edges are present at the input. To keep the resulting offset voltage small, input resistor R2 has a lower value than usual (10 kΩ in this case). The input sensitivity of this final amplifier has also been made relatively low to reduce the likelihood of overdriving. Its gain is thus set to approximately 10.5 using R17 and R18 (but don't forget the effect of R1).

For IC1 to do its job properly, the input must be AC coupled. With the value chosen for the input resistor, the capacitance necessary to yield a sufficiently low roll-off frequency can only be provided by a capacitor with rather large dimensions, so we decided to put it on the printed circuit board for the overdrive indicator circuit. That allows the circuit board for the final amplifier to remain relatively small and increases the distance between the mains transformers and the amplifier circuit board. An additional benefit is that the overdrive indicator can directly access the input signal after the input capacitor, which eliminates the need for an extra tap.

The frequency compensation for the amplifier, which largely determines its open-loop characteristics, is provided by C2/R9 and C3/R12. HF decoupling is provided by the C10/R19 network.

The amplifier output is switched by a 16-A relay. The type specified in the components list has an industry-standard pinout and can be replaced by other equivalent types.

To protect the loudspeakers against output offset voltage transients that may occur when the mains voltage is switched on, a switch-on delay is provided for the output relay. The delay time is approximately 6–7 seconds. The state of the relay depends on the supply voltage for the final amplifier, rather than the transformer voltage. If the supply voltage drops below 17 V, MOSFET T18 de-energises the relay.

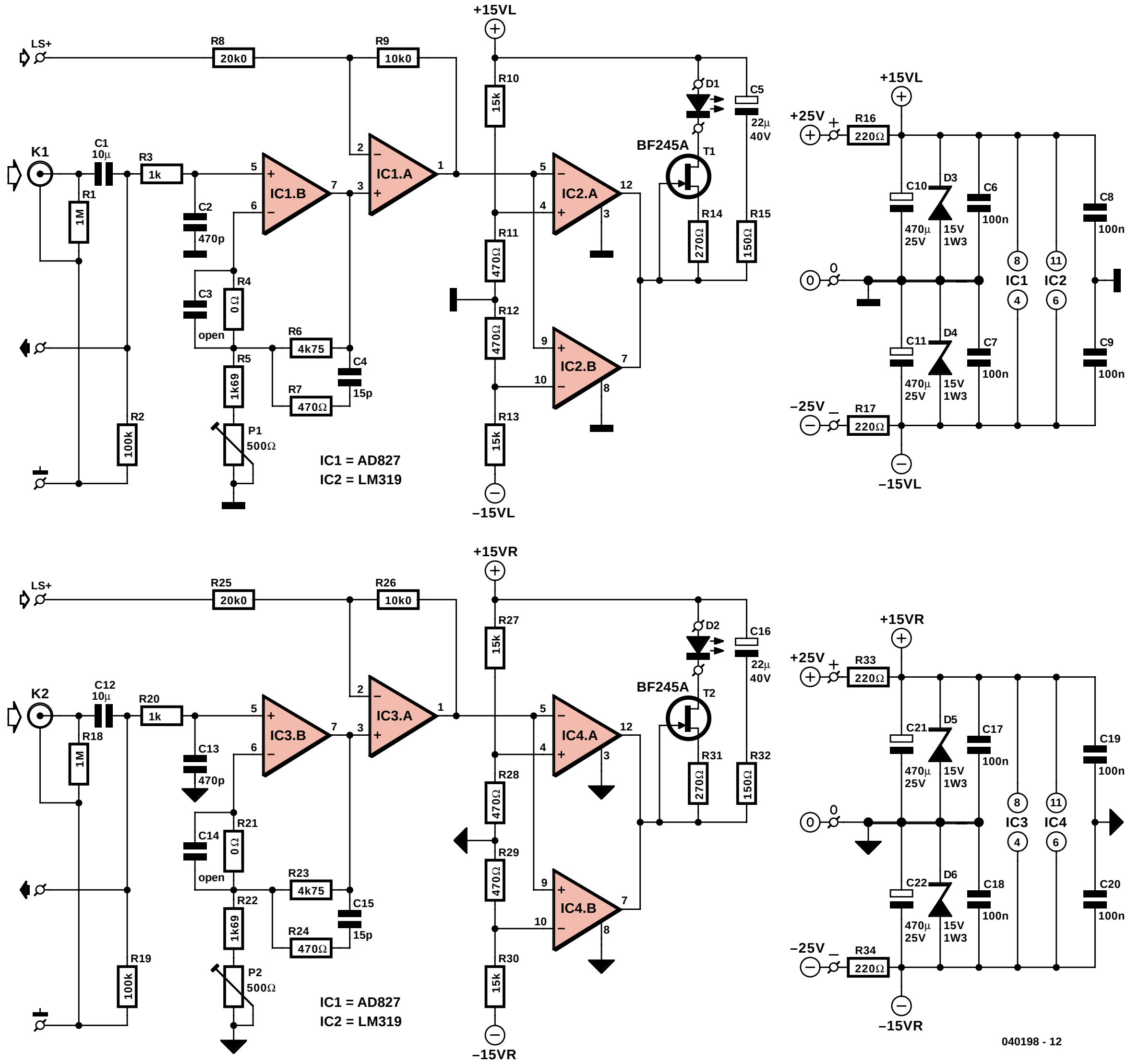


Figure 3. The stereo overdrive indicator constantly compares the input and output signals of the final amplifier and shows whether the difference is too large. The input capacitors for the amplifier boards are also located here.

The MOSFET starts to conduct when its gate voltage is approximately 2.5 V. The threshold value can vary from one FET to the next, so the delay time can also vary. The gate is connected directly to the supply voltage via voltage divider R42–R44. Diode D12 prevents C18 from holding the MOSFET in the 'on' state when the supply voltage drops, and D13 ensures that C18 discharges if the supply voltage drops below half its normal value. This avoids excessive reduction in the delay time if the amplifier is switched on too quickly after being switched off.

The relay coil is connected to the supply voltage via R45, so a relay with a lower coil voltage can be used by changing the value of R45. Be sure to use a resistor with an adequate power rating.

The DC protection circuit is also coupled to the switch-on delay circuit. It de-energises the relay by quickly discharging C18 in the switch-on delay circuit. If a positive DC voltage is present, T16 will conduct and discharge capacitor C18 via R40 if the voltage at the amplifier output is greater than +1.5 V. If an excessively large negative voltage is present at the output, T17 will conduct due to the negative voltage on its emitter. In this case, the discharge current from C18 must flow through R37 and R38. The resistance values of voltage divider R42–R44 also determines the values of R37–R39. The latter values must be kept relatively low, since otherwise the threshold for detecting a negative DC voltage on the amplifier output would be too great. It is approximately –3.5 V with 100-kΩ resistors. This means that the circuit is somewhat less sensitive to negative DC voltages than to positive DC voltages, but that does not have any practical consequences.

Overdrive indicator

As the output power of this amplifier is rather modest by contemporary standards, it's quite possible for it to be overdriven during loud music passages, which will result in clipping. For this reason, a reliable LED indicator circuit has been added to the design to indicate overdrive conditions. The indicator circuit (**Figure 3**) constantly compares the output signal with the input signal and generates a warning if they do not match. Apart from having greater amplitude, the output signal from the amplifier should have the same phase and frequency characteristics as the input signal.

The signals are compared by attenuating the output signal and amplifying the input signal. The input signal amplifier built around IC1b (or IC3b for the other channel, which is not further described here) simulates the transfer

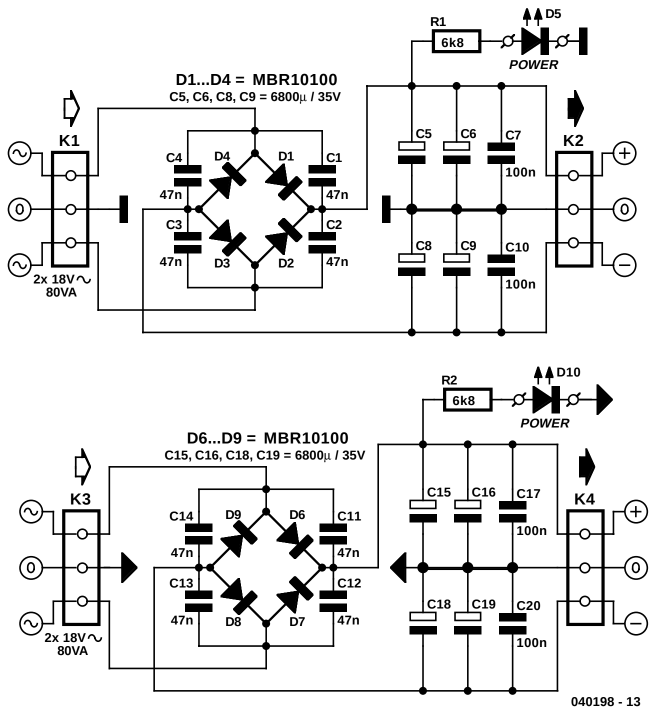


Figure 4. The power supply uses Schottky diodes for low voltage losses and has a buffer capacity of 2 × 13,000 µF per channel.

characteristics of the final amplifier as well as possible using simple means. For instance, RC network R3/C2 has approximately the same time constant as the low-pass filter at the input of the final amplifier (R1//R2 and C1). R3 and C4 are reserved for unusual input configurations in other applications and are not used here. The network formed by C4 and R7 does not have much effect here, but it is included for the sake of completeness.

IC1 is a fast dual opamp (type AD827). Here IC1a is configured as a difference amplifier that compares the two signals at its inputs. R8 and R9 attenuate the loudspeaker signal by a factor of two (10.5 ÷ 2). This means that IC1b

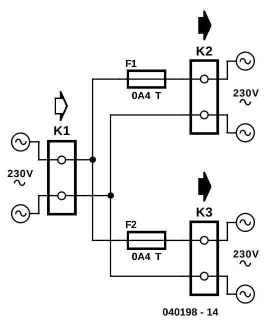


Figure 5. An auxiliary circuit for the mains fuses and some terminal blocks for connecting the transformers and mains switch.

only has to provide a gain of 3.5 (10.5 ÷ 2 ÷ 1.5). The gain of IC1a can be adjusted using P1 to compensate for variations in component values. In theory, P1 should be set to 210 Ω.

The output signal from IC1a is fed to the input of IC2, a dual comparator that is used here as a window comparator. The threshold levels are +0.5 V and –0.5 V, which means that the output voltage must differ from the expected value by more than 1 V (10.5 times the input voltage) to generate an warning signal. It hardly needs saying that a large difference is most likely a sign of clipping, which occurs when the amplifier is overdriven. Reducing the threshold levels would excessively increase the sensitivity of the circuit to noise and other forms of HF interference, so the LED would tend to light up before the amplifier was actually being overdriven.

The combined open-collector outputs of the comparators charge capacitor C5. Here R15 provides current limiting. The voltage on C5 is used to cause LED D1 to shine quite visibly even if overdrive only occurs during short peaks. After the comparator outputs have returned to the high-impedance state, the LED will continue to be supplied with current via constant-current source T1. This causes the energy stored in the capacitor to be used as effectively as possible, so the LED will shine with constant brightness as long as possible. The current through the JFET is set to approximately 2 mA by R14. Using a low LED current allows a fairly small capacitor to be used for C5, which saves space. However, this requires using a low-current LED that shines brightly with a current of 2 mA (an example is given in the components list).

The input ground is kept strictly separate from the remainder of the circuit on the circuit board, in order to avoid ground loops. The input signal ground is connected to a star point on the amplifier circuit board. The input signal and the ground connection of the input socket on the indicator board go from here to the amplifier board. Resistor R1 ensures that the input terminal of C1 is always connected to ground. R2 is necessary to provide the bias current for the input of IC1b. As R2 is connected in parallel with the input, we decided to tie it to the input ground of the amplifier board. Drawing the bias current for IC1b from that source does not have any negative effects. The advantage of placing the input capacitor on the indicator board is that it allows the signals to be compared from DC to more than 20 kHz. This avoids any measurement errors due to the lower roll-off frequency.

The supply voltage for the indicator circuit is taken from the main power supply and stabilised using Zener diodes, since the value is not critical. Allowance has been made for a fairly high load on the supply and associated voltage slumps. For this reason, the Zeners are operated at relatively high currents, which means that the associated series resistors (R16, R17, R33 and R34) must be 1-watt types. To save space, we selected especially small types (Vishay BCcomponents PR01 series).

Power supply

A separate printed circuit board has been designed for the power supply (**Figure 4**). It just fits into the enclosure between the transformers. The power supply design is quite conventional, with a standard 80-VA toroidal transformer, rectifier, and electrolytic smoothing capacitors for each channel. To make it easier to connect the transformers, we also designed a separate circuit board with three terminal blocks and two fuses (**Figure 5**) to allow the primary leads of the two transformers for a stereo amplifier to be properly connected and fitted. The secondary leads

Measured performance

Input sensitivity		1.5 V_{eff} (30 W, 8 Ω, 1 % THD+N) 1.4 V_{eff} (26 W, 8 Ω, 0,1 % THD+N)		
Input impedance		9.3 kΩ		
Sine-wave power	 (0.1 % THD+N) (1 % THD+N)	**8** Ω 26 W 30 W	**4** Ω 42 W 47 W	**2** Ω 58 W 65 W
Power bandwidth	(referred to 1 W/8 Ω)	1.5 Hz to 265 kHz		
Slew rate		18 V/µs		
Rise time		2 µs		
Signal to noise ratio	(referred to 1 W/8 Ω)	111 dBA 101 dB (22 Hz to 22 kHz linear)		
Harmonic distortion plus noise (B = 80 kHz)	 1 kHz 1 kHz 20 kHz	**8** Ω 0.0014% (1 W) 0.0024 % (15 W) 0.0037 % (15 W)	**4** Ω 0.0019 % (1 W) 0.0035 % (30 W) 0.0075 % (30 W)	**2** Ω 0.0029 % (1 W) 0.0053 % (40 W) 0.01 % (40 W)
Intermodulation distortion (50 Hz : 7 kHz = 4 : 1)		0.0030 % (1 W) 0.0077 % (10 W)	0.0037 % (1 W) 0.012 % (20 W)	0.007 % (1 W) 0.018 % (30 W)
Dynamic IM distortion (3.15-kHz square wave + 15-kHz sine wave)		0.0013 % (1 W) 0.0010 % (15 W)	0.0013 % (1 W) 0.0013 % (30 W)	0.0016 % (1 W) 0.002 % (40 W)
Damping factor	1 kHz/8 Ω 20 kHz/8 Ω	> 450 > 230		
Open-loop parameters (15 W/8 Ω/1 kHz)	Gain Bandwidth Output impedance THD+N 1 kHz THD+N 20 kHz	2183 55 kHz 0.5 Ω 0.48 % 0.53 %		
Overdrive indication (at P_{max})	10 Hz 1 kHz 20 kHz	1.5 % THD+N 1.5 % THD+N 0.3 % THD+N		
DC protection		+1.5 V/–3.5 V		

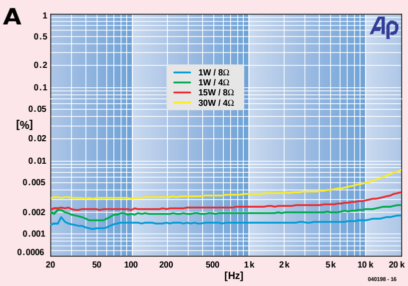
Besides these numerical results, we also recorded several plots that give a bit more insight into the quality and character of the amplifier.

Plot A shows several measurements of harmonic distortion and noise with a bandwidth of 80 kHz. Two measurements were made at a power of 1 W with 8-Ω and 4-Ω loads, one measurement was made at 50% maximum power level with an 8-Ω load, and the final measurement was made with the same output voltage and a 4-Ω load. The four curves are very close together, which suggests that the amplifier is practically insensitive to the load impedance.

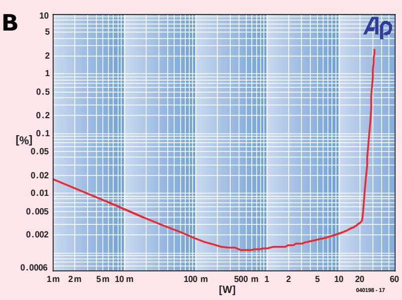
Plot B shows distortion plus noise as a function of output amplitude with an 8-Ω load. At power levels up to 0.5 W the measured value is determined by the very low noise level. Above this level the distortion increases gradually until approximately 21 W. At this point the amplifier departs from linear operation and enters a soft-clipping region. The soft-clipping region extends over roughly the final ten watts of the amplifier's output power (or the final 3.5 volts of its maximum output voltage). This measurement was made using a 1-kHz signal and a bandwidth of 20 kHz to make the effect of the harmonic components more visible.

Plot C shows a Fourier analysis of a 1-kHz signal at 1 W with an 8-Ω load. Here the harmonic distortion is dominated by the second harmonic, which lies at a level of approximately –104 dB. This makes it negligible. The same holds true for the effect of the magnetic field of the mains transformer, which is responsible for the 50-Hz component in the plot. This level corresponds to a power of only 10 pW.

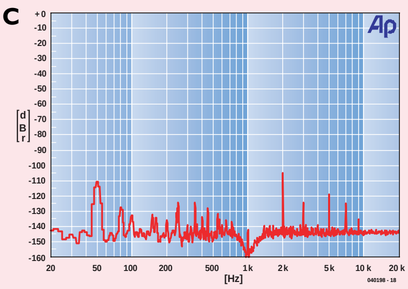
A



B



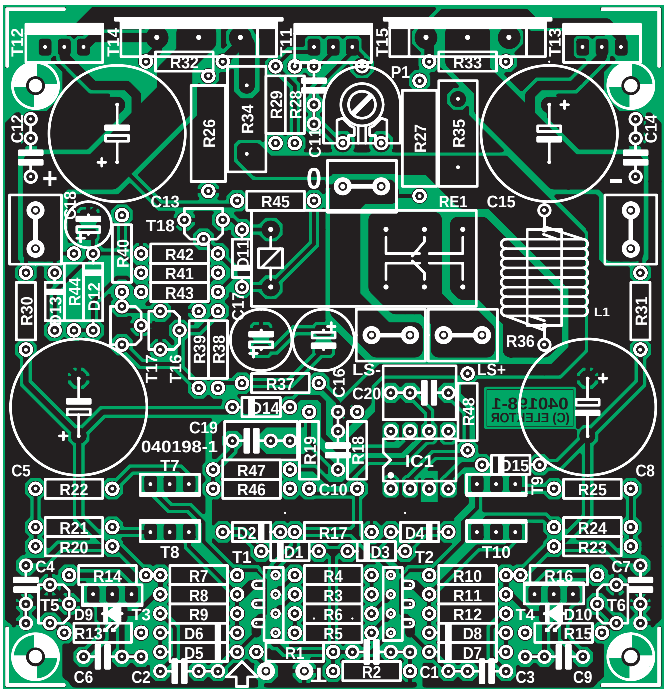
C

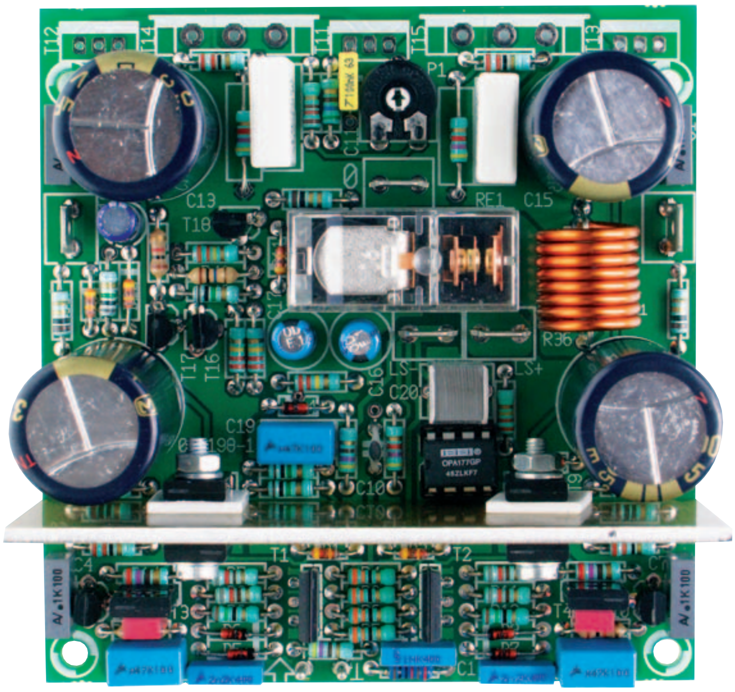


of the transformers are connected to the printed circuit board holding the smoothing capacitors and rectifier diodes. The electrolytic smoothing capacitors are quite ordinary types (6800 µF, 35 V) with radial leads and a diameter of 25 mm (there's no room in the enclosure for anything larger). The type used here has a quite low profile (only 30 mm), so there's more than enough room in height.
Due to the relatively low supply voltage, we used Schottky rectifiers made by ON Semiconductor. They are rated at 100 V / 20 A peak. The forward voltage drop per diode is only 0.5 V at 3.5 A.
Three-way circuit-board terminal blocks are used for all external leads to simplify fitting the wiring and any necessary maintenance. There are also two LEDs on the circuit board (one per channel) to serve as supply voltage indicators. If the mains switch does not have a pilot lamp (on the front panel), they should be used instead.

Construction

The compact, double-sided printed circuit board for the amplifier is shown in **Figure 6.** We don't often use double-sided boards for *Elektor Electronics* final amplifiers, but here this provides definite advantages and yields shorter signal paths between the components.
Fitting the components to the board should not pose any particular difficulties (no SMDs this time), but you do have to work accurately. All of the components are very close together, which means you have to fit them tidily, use components with the proper dimensions, and take care when fitting them (correct polarity and value).
There are a few details that deserve special mention. LEDs D9 and D10 must be fitted with their flat sides in contact with T3 and T4 for good thermal coupling, in order to make the current sources temperature-independent.
Transistors T7–T10 are screwed to a common heat sink. Use a piece of aluminium for this, with dimensions of 85 × 38 mm (the same as the circuit board) and a thickness of 1.5–2 mm. The height is such that the top panel of the enclosure can just be fitted in place. Use *ceramic* insulators (Fischer type AOS 220) for all four transistors, and screw them in place in facing pairs.
Use the specified low-inductance resistors for R34 and R35, since otherwise there's a very good chance that the amplifier will be unstable.
Coil L1 can be made by winding 1.5-mm



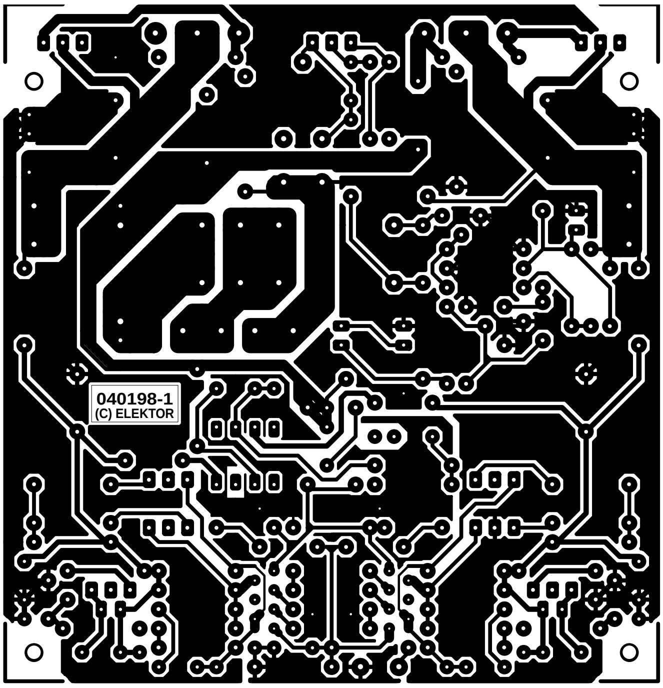


COMPONENTS LIST

Amplifier board (040198-1)

Resistors:
R1,R17,R19 = 470Ω
R2 = 10kΩ
R3-R6 = 33Ω
R7,R8,R10,R11 = 820Ω
R9,R12 = 15Ω
R13,R15,R40 = 1kΩ
R14,R16 = 270Ω
R18,R26,R27 = 4kΩ7
R20,R23,R32,R33 = 22Ω
R21,R22,R24,R25 = 10Ω
R28 = 560Ω
R29 = 120Ω
R30,R31,R45 = 1Ω
R34,R35 = 0Ω22 MPC71
R36 = 2Ω2, 1W (e.g. Farnell # 306-0408)
R37–R39 = 100kΩ
R41 = 10MΩ
R42,R48 = 330Ω
R43,R44,R46,R47 = 1MΩ
P1 = 250Ω preset

Capacitors:
C1 = 1nF
C2,C3 = 2nF2
C4,C7,C11,C12,C14 = 100nF
C5,C8,C13,C15 = 3300µF 35V, radial, lead pitch 7.5mm, diam. 18mm max. (e.g. Farnell # 303-6467)



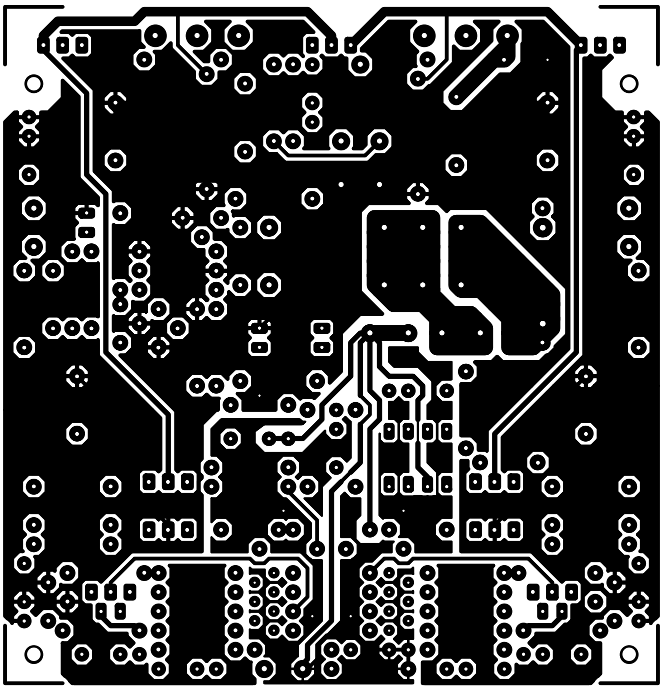


Figure 6. The mono amplifier circuit board is double-sided and through-hole plated to keep the signal paths between the components short and avoid wire bridges.

diameter enamelled copper wire on a 10-mm drill bit (the smooth part, of course). Leave fairly generous leads and scrape them nicely clean with a small knife. Then insert R36 in the coil, bend over its leads and fit the assembly to the circuit board. Then solder the coil and resistor in place. Cut off the free ends of the coil leads after they are soldered, but not too close to the circuit board.

Don't solder the driver transistors, output transistors or quiescent-current transistor to the circuit board just yet. The leads of these transistors must first bent to the right spacing, so they can rest properly on the heat sink without any mechanical strain after being fitted. That is described in detail under 'Final assembly'.

Next comes the indicator circuit board (**Figure 7**). It doesn't involve much out of the ordinary. R16, R17, R33 and R34 must be fitted a bit above the board and bent slightly away from the decoupling capacitors, since they must dissipate a fair amount of heat. If you wish, you can use a multimeter to adjust presets P1 and P2 to 210 Ω before soldering them to the board (refer to 'Alignment' at the end of the article). For C1 and C2, be sure to buy a type that has the right dimensions to fit on the circuit board.

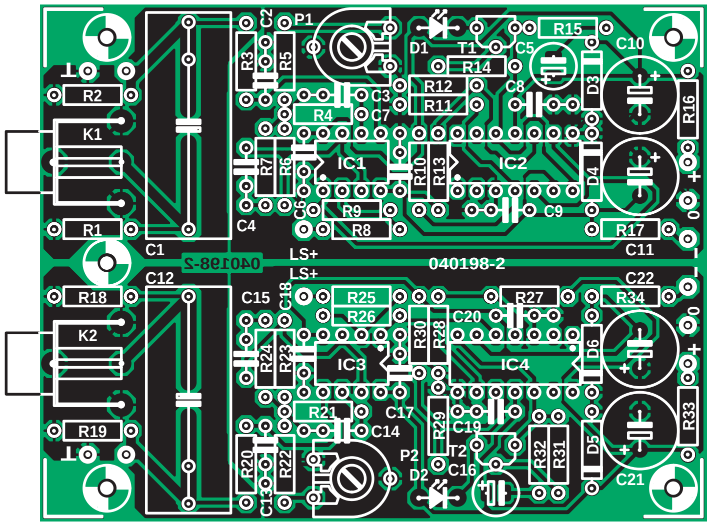
The power supply board shown in **Figure 8** can be quickly put together, but here again you must be sure to use capacitors with the right dimensions and the proper type of Schottky diode. **Don't forget the wire bridges under and between D2/D3 and D7/D8.**

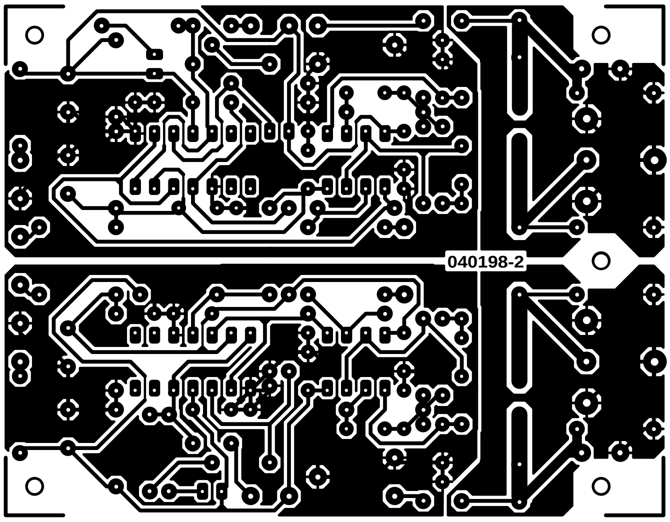
C6,C9,C19 = 470nF
C10 = 15pF
C16,C17 = 47µF 25V, radial
C18 = 10µF 63V, radial
C20 = 2µF2 MKT, lead pitch 5mm or 7.5mm

Inductors:
L1 = 8 turns 1.5-mm diam. (SWG30) enamelled copper wire, inside diam. 10mm

Semiconductors:
D1-D4,D11-D13 = 1N4148
D5-D8 = Zener diode, 4V7, 0W5
D9,D10 = LED, flat, red
D14,D15 = Zener diode, 5.6V, 0.5W
T1 = 2SC3381BL (Toshiba, Huijzer, Segor-electronics)
T2 = 2SA1349BL (Toshiba, Huijzer, Segor-electronics)
T3,T7,T8 = 2SA1209 (Sanyo, Farnell # 410-3841)
T4,T9,T10 = 2SC2911 (Sanyo, Farnell # 410-3853)
T5,T6 = BF245A
T11,T12 = 2SC5171 (Toshiba, Huijzer, Segor-electronics)
T13 = 2SA1930 (Toshiba, Huijzer, Segor-electronics)
T14 = 2SC5359 (Toshiba, Huijzer, Segor-electronics)
T15 = 2SA1987 (Toshiba, Huijzer, Segor-electronics)
T16,T17 = BC547B
T18 = BS170
IC1 = OPA177GP (Texas Instruments/Burr-Brown, e.g. Farnell # 205-023)

Miscellaneous:
K1-K5 = PCB-mount spade (Faston) terminal, vertical, 2 pins
RE1 = PCB relay 24V, 16A, 1100Ω (e.g. Omron G2R-1-24 VDC)
Heat sink, 1.25 K/W e.g. (Fischer SK411, height 50mm)
Ceramic insulators for T7-T10 (Fischer AOS220)
Mica insulating washers for T14 & T15 (e.g. Conrad Electronics # 189049)
Enclosure: e.g. UC-202H/SW (Monacor/Monarch)





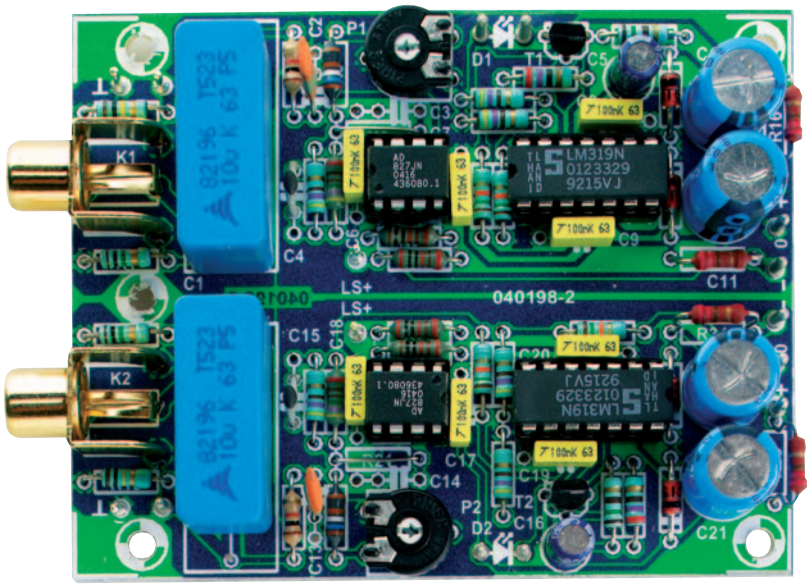


Figure 7. The indicator circuit board also holds the input capacitors (C1 and C12) for the amplifier boards.

COMPONENTS LIST
Indicator board (040198-2)

Resistors:
R1,R18 = 1MΩ
R2,R19 = 100kΩ
R3,R20 = 1kΩ
R4,R21 = 0Ω (wire bridge)
R5,R22 = 1kΩ69
R6,R23 = 4kΩ75
R7,R11,R12,R24,R28,R29 = 470Ω
R8,R25 = 20kΩ0
R9,R26 = 10kΩ0
R10,R13,R27,R30 = 15kΩ
R14,R31 = 270Ω
R15,R32 = 150Ω
R16,R17,R33,R34 = 220Ω, 1W PR01 (Vishay BCcomponents, Farnell # 337-778)
P1,P2 = 500Ω preset

Capacitors:
C1,C12 = 10µF 63V MKT, lead pitch 22.5/27.5mm, W × L = 11 × 31mm max.
(e.g. Farnell # 400-2015)
C2,C13 = 470pF
C3,C14 = not fitted
C4,C15 = 15pF
C5,C16 = 22µF 40V, radial
C6–C9,C17–C20 = 100nF
C10,C11,C21,C22 = 470µF 25V, radial

Semiconductors:
D1,D2 = LED, red, low-current, 3mm, 20mcd at 2mA (Kingbright L-934LSRD, Farnell # 637-075)
D3-D6 = Zener diode, 15V 1.3W
T1,T2 = BF245A
IC1,IC3 = AD827JN (Analog Devices, Farnell # 246-165)
IC2,IC4 = LM319N

Miscellaneous:
K1,K2 = cinch socket, PCB mount (e.g. Monacor T-709G)

COMPONENTS LIST
power supply board (040198-3)

Resistors:
R1,R2 = 6kΩ8

Capacitors:
C1-C4,C11-C14 = 47nF ceramic, lead pitch 5mm
C5,C6,C8,C9,C15,C16,C18,C19 = 6800µF 35 V, diam. 25mm, lead pitch 10mm (Farnell # 652-090)
C7,C10,C17,C20 = 100nF MKT, lead pitch 5mm or 7.5 mm

Semiconductors:
D1-D4,D6-D9 = Schottky rectifier, 10A, 100V, TO-220AC package (e.g. ON Semiconductor MBR10100, Farnell # 878-443)
D5,D10 = LED, red, low-current

Miscellaneous:
K1-K4 = 3-way PCB terminal block, lead pitch 5mm

Finally there is the small circuit board with the mains fuses and terminal blocks for distributing the mains voltage (**Figure 9**). Of course, this board isn't essential, but it makes the wiring quite a bit easier (and safer!).

Final assembly

As we already mentioned, the amplifier uses the same type of enclosure as the 'High-End preamp' preamp. That's why we went to so much trouble to keep everything as compact as possible. For the heat sinks, we looked for types that were not too long and still had sufficiently low thermal resistance. We finally settled on the Fisher SK411, which with its modest dimensions of 116.5 × 50 × 50 mm leaves enough space between the heat sinks to fit the loudspeaker terminals (using relatively small types) and the input connectors. To maximise the available space, we milled a 1.7-mm rebate around the edge of the heat sinks, so their bases fit nicely between the two fastening rails. This also causes the edges of the heat sinks to be flush with the side faces of the enclosure. Each heat sink is secured by four M3 screws fitted through the heat sink and the rails. The head of an M3 screw will fit between the cooling fins of this type of heat sink. In principle, the output transistors could be attached the same way. This would make assembly a bit easier, but it would also make the circuit board unnecessarily large.

A small 1.5-mm aluminium plate can be fitted between the two heat sinks (measure carefully) to hold the speaker terminals and input connectors. The original rear panel of the enclosure is not used.

There's actually not any room left for the mains cable (and certainly not for an IEC mains inlet). In our prototype, we used a sturdy mains cable and strain relief routed through the bottom plate next to the circuit board with the fuses. In this case, the enclosure must rest on high feet that are high enough to prevent the mains cable from being pinched off.

Fit the mains transformers as close as possible to the front panel. The mains switch and LEDs from the overdrive indicator can then be fitted on either side of the fuse circuit board and the transformers. Attach the fuse circuit board to the base plate using 10-mm standoff sleeves with nylon screws and nuts (the terminal blocks we used are partially open at the rear). This ensures Class-II isolation (be sure to read the instructions on the 'Safety Guidelines' page). In general, always pay careful attention to the isolation of components carrying mains voltage. Maintain a distance of at least 6 mm from parts that can be touched by users.

For each power supply terminal, insert the supply leads for the amplifier boards and indicator board together in the appropriate terminal blocks. Heavy wire with a cross-sectional area of 2.5 mm^2 must be used for the amplifier boards, but thin flexible wire is adequate for the overdrive indicator board. Make the supply leads for the amplifier boards as short and direct as possible. We crimped tab connectors to the ends of the leads, but you can also solder them directly to the boards. Connect the speaker outputs on the circuit boards to the output sockets in the same manner. At the sockets, we simply fitted wire lugs to the ends of the wires, which avoids the contact resistance of an additional plug. The power supply leads for the overdrive indicator board can be simply soldered in place. Fit the overdrive indicator board in the middle between the amplifier boards. The input sockets will then be almost

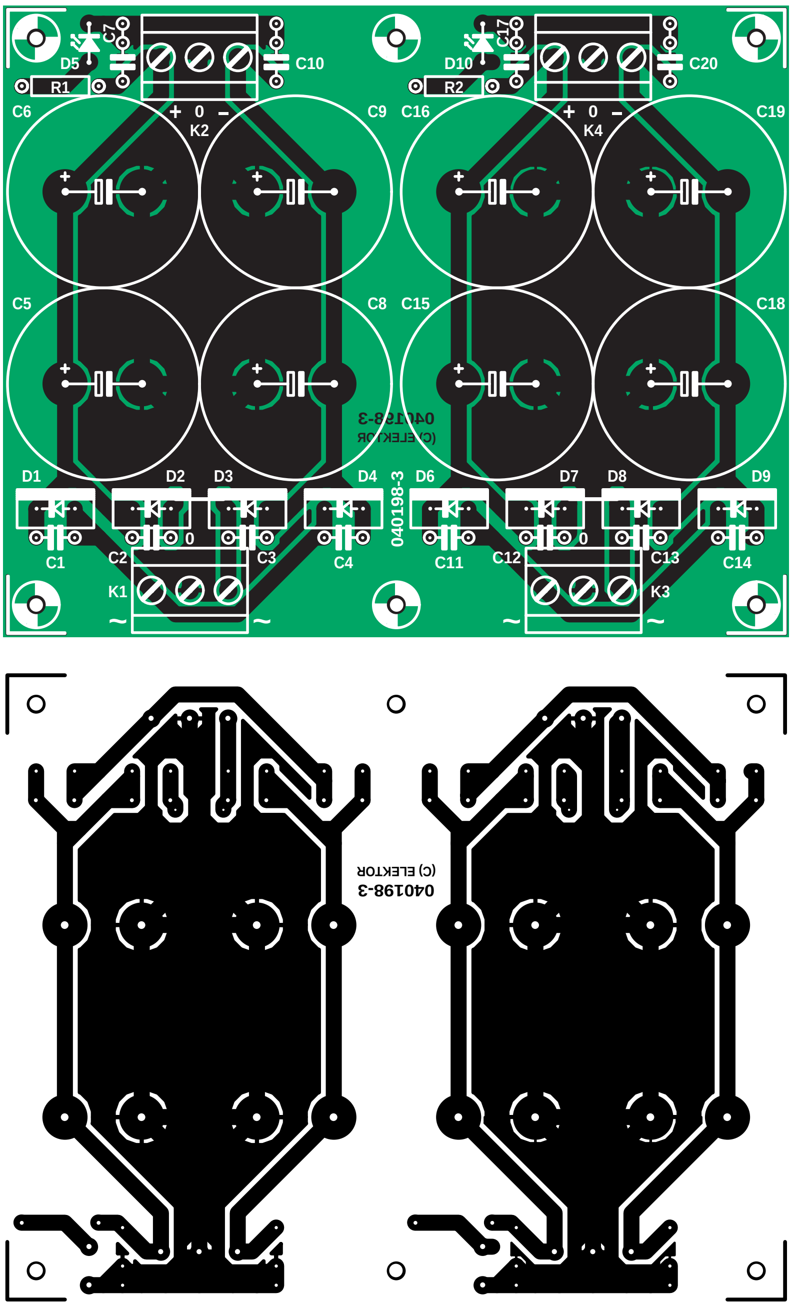
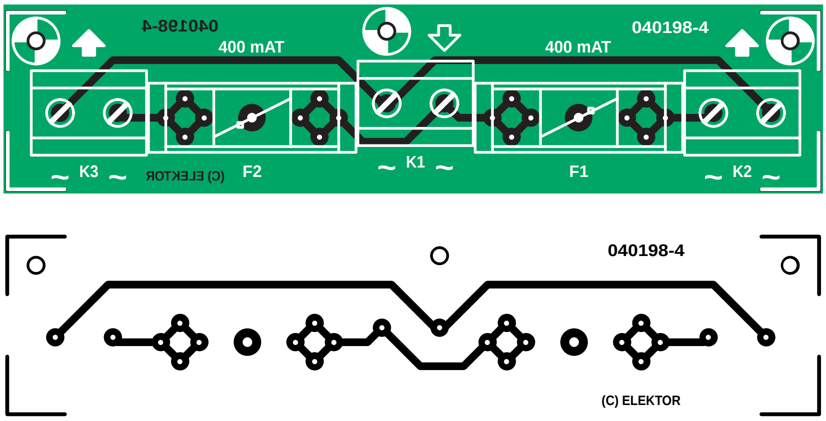


Figure 8. The power supply circuit board holds a total of eight electrolytic capacitors to power two amplifier boards.



COMPONENTS LIST

Fuse board (040198-4)

Miscellaneous:

K1–K3 = 2-way PCB terminal block, lead pitch 7.5 mm

F1,F2 = fuse, 400mAT (time lag), with PCB mount holder

Figure 9. A small auxiliary circuit board for the fuses and mains connections.

flush with the rear panel. Naturally, it will take a bit of measuring and fitting to get the holes for the connectors in the right places.

The manner in which the heat sinks are attached causes the mounting surface to be approximately 6 mm away from the edge of the circuit board, assuming that the amplifier circuit boards are attached directly to the rails. This means that the leads of the transistors must be bent twice allow them to be positioned correctly without creating permanent mechanical stresses in the transistors. For the large

PCBs:

Required for a stereo amplifier (see Readers Services page):

2 ea. order code **040198-1**
1 ea. order code **040198-2**
1 ea. order code **040198-3**
1 ea. order code **040198-4**

output transistors, the first bend will have to be somewhat closer to the plastic package than what is intended (in the wide part), since otherwise the leads will touch the rails or won't fit at all. Another possibility, which has the disadvantage that it weakens the structure, is to file away enough metal at the appropriate locations to provide adequate clearance. We intentionally rejected this option.

Once you have taken care of this detail and the transistors can easily be placed flat against the heat sink with their leads passing freely through the holes in the circuit board, you can mark the locations of the mounting holes on the back of the heat sink. Naturally, you should do this before the transistors have been soldered to the circuit board, but after the final holes for fastening the amplifier circuit boards have been made in the bottom plate. Once the output transistors and their drivers have been fitted to the heat sink and soldered to the circuit boards, the bottom plate can be removed quite easily. Be sure to not exert too much force on the leads of the transistors when the boards are suspended this way.

Power transistors T14 and T15 must be fitted to the heat sink using insulators (mica washers), while the other three transistors (T11, T12 and T13) can be screwed directly to the heat sink. Be sure to use thermal paste with all of the transistors.

After you have made all the mounting holes for the remaining circuit boards, transformers, mains switch and LEDs (in the front panel) and ventilation holes in the enclosure, you can screw everything else in place and fit the wiring.

Use well-screened audio cable to connect the inputs of the amplifier boards to the terminals near the input sockets on the overdrive indicator board. Use the two ground terminals next to the input sockets to connect the enclosure

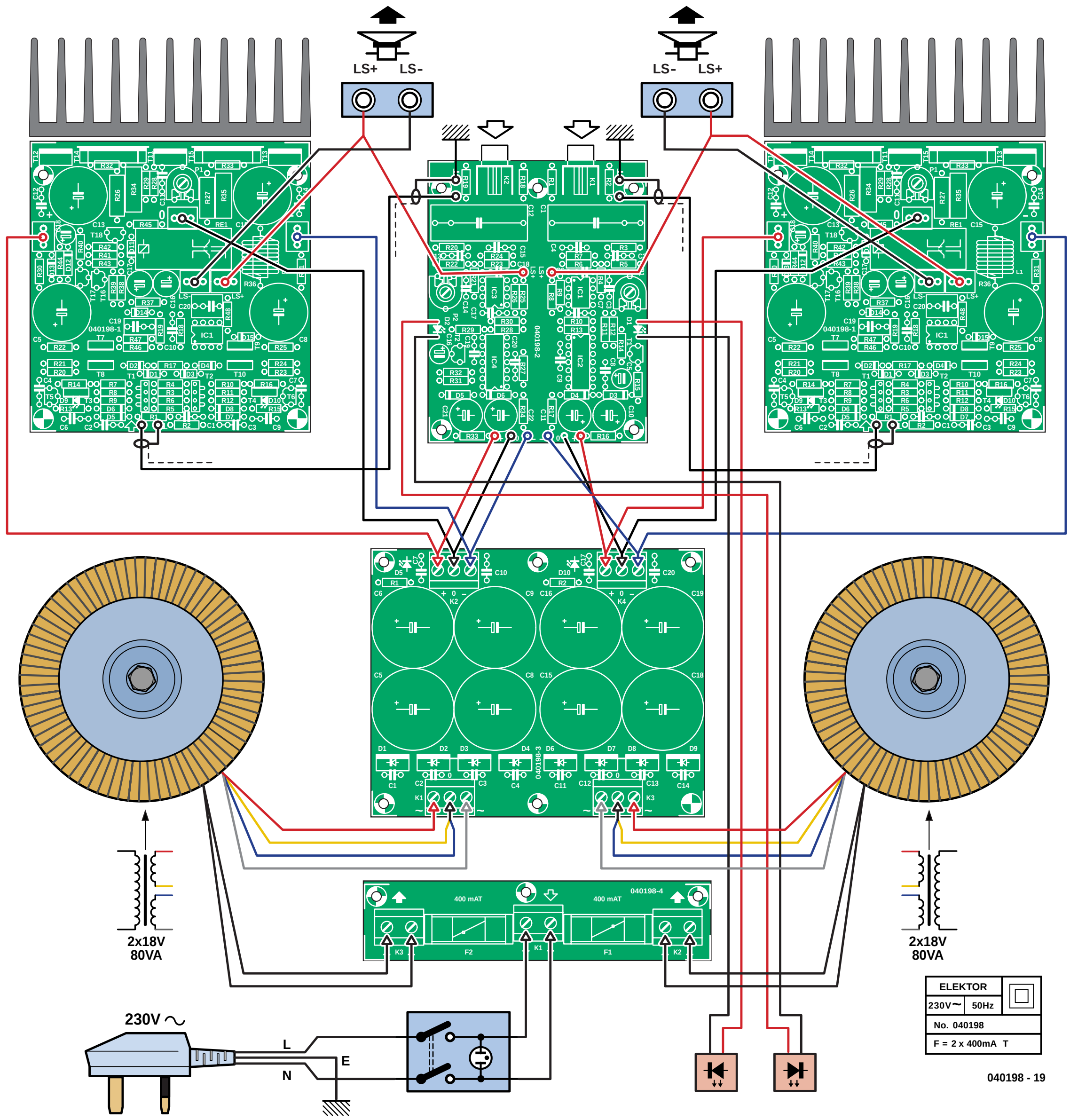


Figure 10. The wiring diagram shows the proper way to connect everything together inside the enclosure.

Transistor selection

Toshiba type 2SC3381 and 2SA134 dual transistors are used for the differential amplifiers in the input stage. These transistors may be difficult to obtain, but they are practically the only good complementary dual transistors still commercially available.

We selected Sanyo type 2SC2911 and 2SA1209 transistors for the differential amplifiers in the voltage gain stage. These transistors come in a TO-126 package and have outstanding linearity combined with low output capacitance (3 pf and 4 pF, respectively), and they can handle up to 160 V. The maximum dissipation is 10 W (4 W at 100 °C), and the maximum rated collector current is 200 mA for 1 second or 140 mA continuous. The maximum allowable emitter–base voltage is only 5 V, so a pair of 4.7-V Zener diodes are connected back to back between the outputs of each of the input differential amplifiers (D5–D8). The dual transistors have similar restrictions, so two back-to-back diodes are placed between the inputs of each of the differential amplifiers (D1–D4).

Matching the respective pairs of transistors (NPN-NPN and NPN-PNP) will certainly help improve the quality of the amplifier, so you should pay attention to the h_{FE} parameter classification (GR/BL, R/S/T/, and R/O) when purchasing these components.

For the output transistors and drivers we chose some old friends, namely the transistors used in the Titan 2000 amplifier (Elektor Electronics, February–May 1999). These output transistors have a relatively constant h_{FE} up to 5 A, so they do themselves proud in this amplifier. They can handle a maximum continuous current of 15 A, which in practice means that the amplifier can be used with load impedances as low as 1.5 Ω .

For integrator IC1 we selected an OPA177, which has a very low offset voltage (100 µV between –40 °C and +85 °C). This opamp has a maximum bias current of ±6 nA, which makes a small contribution to the residual offset at the output of the amplifier (±2 mV maximum, equal to 6 nA × 330 kΩ).

We chose a MOSFET for T18 in the protection circuit so that voltage divider R43/R44 could have a high impedance. This allows the capacitance of C18 to be kept relatively small, which has two benefits: the dimensions of C18 can be kept modest, and the DC protection circuit does not have to draw so much current from C18.

to the grounds of the two channels. This avoids creating any ground loops. The measurement inputs for the amplifier output signals are located in the middle of the overdrive indicator board. The easiest way to connect them is to use thin, flexible wire to connect them to the output sockets.
For the sake of clarity, the complete wiring diagram of the final amplifier is shown in Figure 10.
One of the cover panels for the enclosure is solid, while the other one has punched perforations. To keep the assembly sturdy, we decided to use the solid panel for the bottom panel and the 'open' panel for the top panel, which provides continuous ventilation for the enclosure. However, we recommend making additional ventilation holes in the bottom panel near the amplifier circuit boards to provide additional cooling for T7–T10.

Alignment

After everything has been assembled and the wiring has been completed, you come to the moment when you switch on the amplifier for the first time. As a precautionary measure, connect 33 Ω, 5 W resistors in series with the power supply lines before switching on the amplifier, just in case something has been fitted incorrectly. Also first turn P1 fully to the left (anticlockwise) to set the quiescent current to a minimum. After this, you can switch on the power.
To set the quiescent current, you can measure the voltage between the emitter of T14 or T15 and R36 (which should be easily accessible with a measurement probe). Adjust the voltage to 22 mV. If you can do this for both output stages and the settings remain stable, you can switch off the amplifier and remove the resistors from the supply leads.

The easiest way to adjust the overdrive indicator is to use an oscilloscope, but you can certainly manage without one. After the quiescent current has been set, drive the amplifier to just below its maximum output amplitude using a 1-kHz sine-wave signal. This is best done using an 8-Ω load resistor with a power rating of at least 30 W, but it can also be done without a load. Measure the signals at the outputs of IC1a and IC3a, and adjust them to a minimum. If you do not have access to a scope, you can use a multimeter. The simplest method is to set the presets to 210 Ω before soldering them to the circuit board.

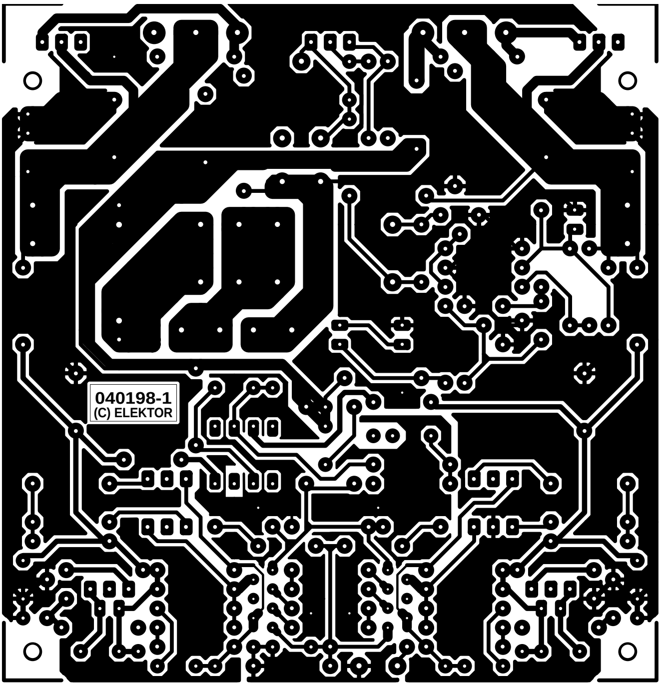
Let there be sound!

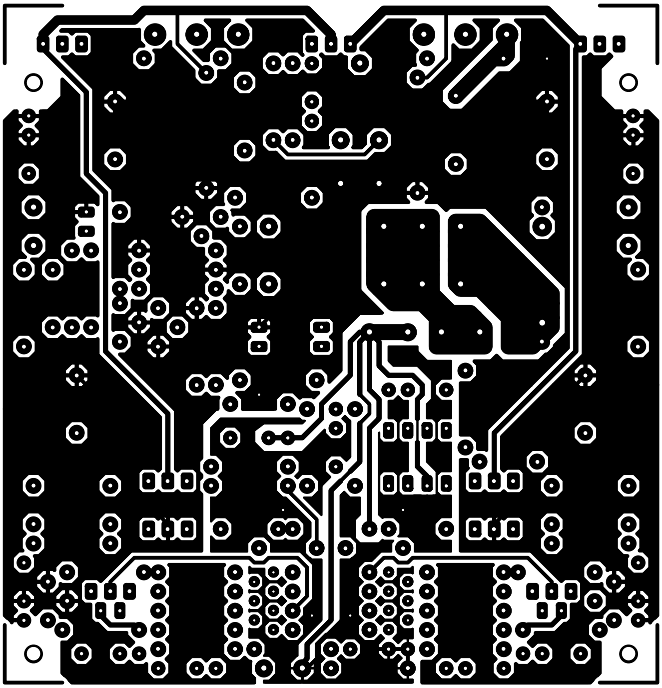
Once all this has been done, the amplifier can do what it's supposed to do: amplify audio signals in a very refined manner. The initial sound impressions from this amplifier are clearly reminiscent of its predecessors with the same basic design, such as the *Elektor Electronics* Power Amp, HEXFET Power Amp and IGBT Power Amp. The choice of components and the compact, double-sided circuit board layout appear to bear fruit here in the form of an especially transparent, unstrained reproduction with a very spacious, well-detailed stereo image. This amplifier demands a pair of outstanding loudspeakers, preferably relatively large types (since they usually have better efficiency than compact speakers). After all, the output power is rather modest, but that shouldn't present a problem for anyone outside the ranks of hard-rock, grunge and heavy metal fans. The accurate indicators are quite helpful in setting the maximum volume, and this amplifier shows once again how little power is actually necessary for outstanding stereo reproduction at a respectable sound level.

(040198-1)

Warning

This design incorporates power supply transformers connected to the mains. That can lead to potentially lethal situations if you do not pay careful attention to the safety regulations for such circuits. Please be sure to carefully follow the instructions on the Electrical Safety guidelines page, which appears from time to time in the magazine. If the type of mains transformer shown on the schematic diagram is used, this circuit can be assembled in an enclosure as a Class-II device. Attach the type label shown on the wiring diagram to the assembled enclosure.





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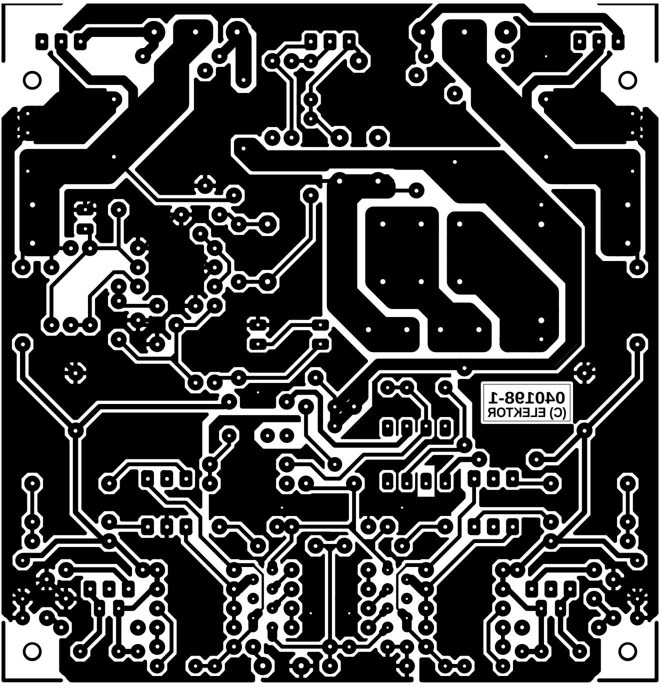

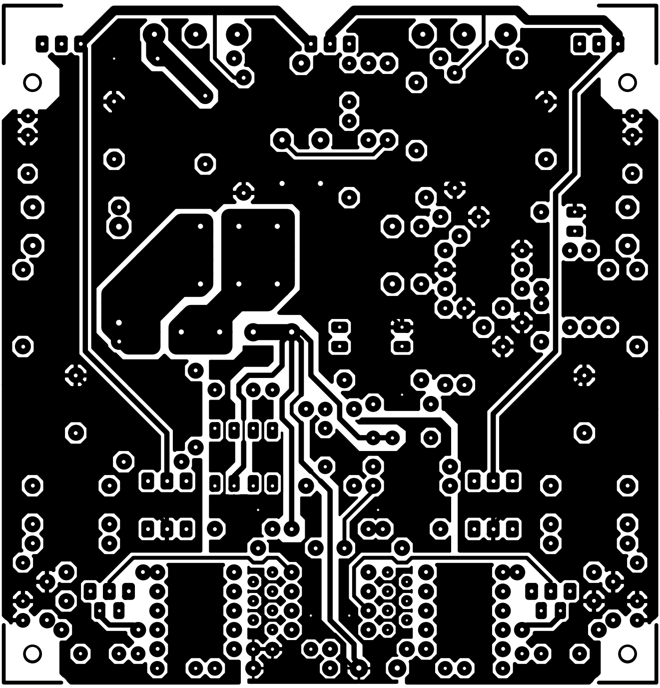

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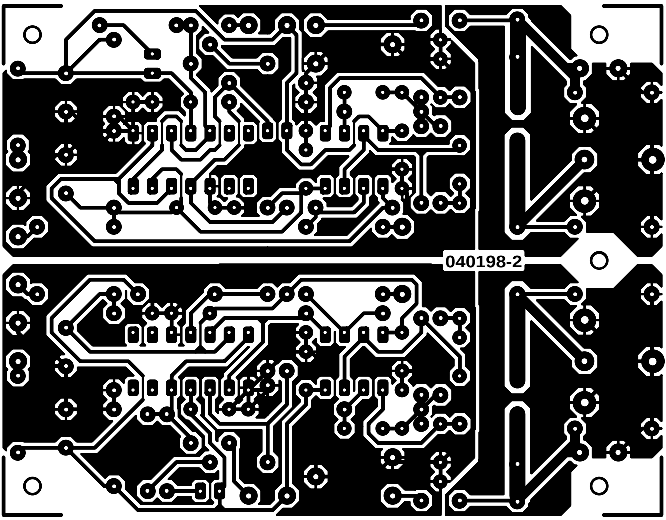

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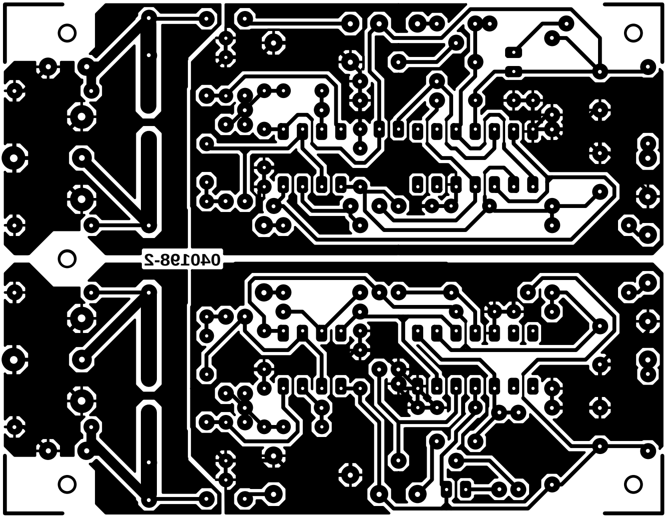

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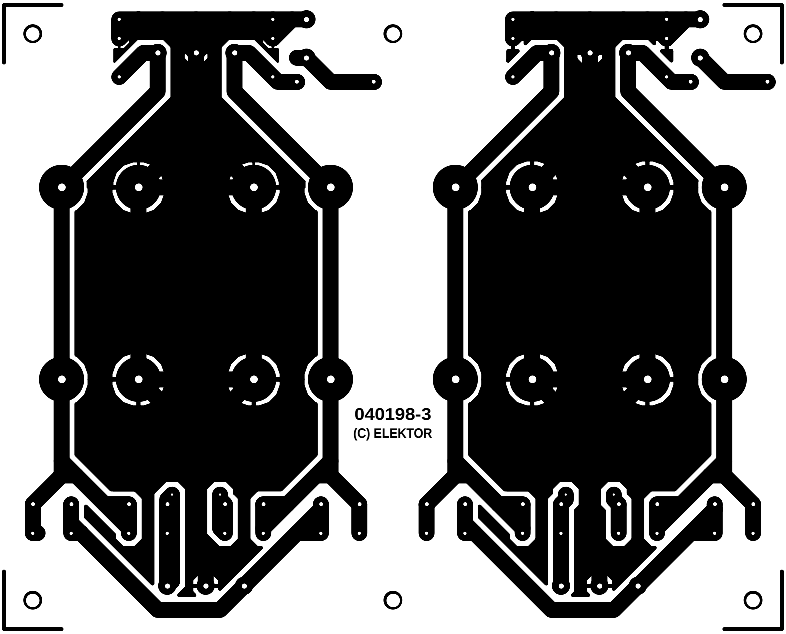

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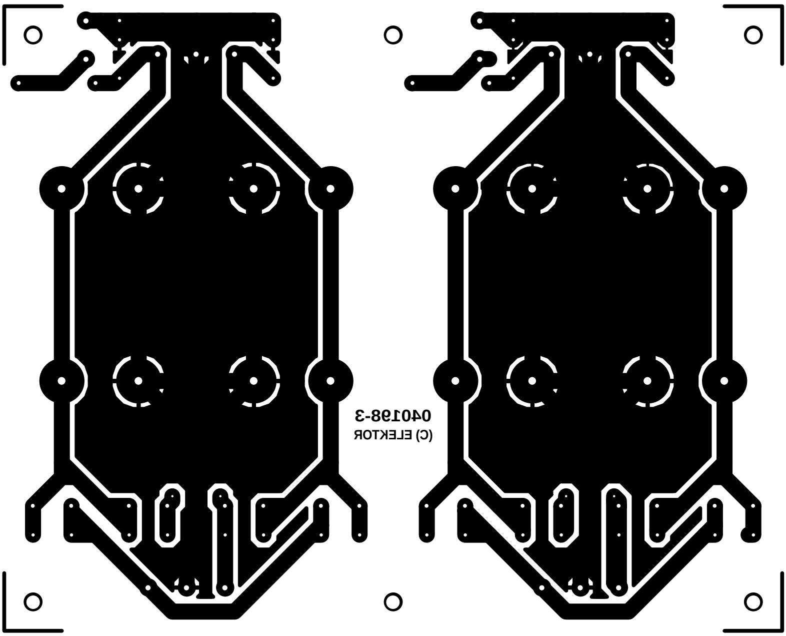

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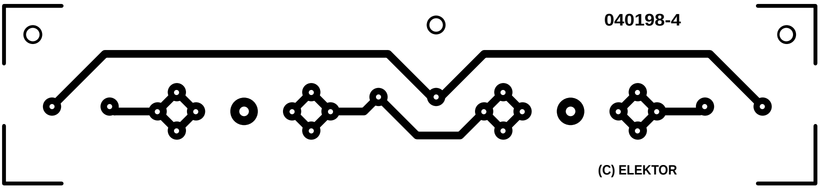

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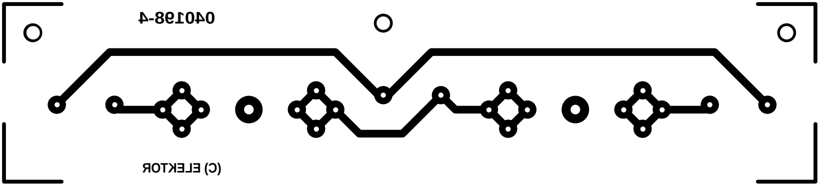

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Opinions on Valves

Valves versus transistors

We have two audio amplifier design articles in this issue: one with valves and the other with transistors. They are just as different as people's opinions on the issue of valves versus transistors.

Does it make sense to go back to the amplifier technology of our grandparents? Is there really such a thing as the 'legendary valve sound'? Or is this only a question of our emotions refusing to bow to physical measurements? We attempt to resolve this issue by viewing it from three different perspectives. One belongs to a designer in our own Elektor Electronics lab, a second one belongs to the world of professional studio engineering, and a third one combines the physical and musical aspects.

Figure 1. The Schoeps M222 valve microphone is designed for fixed or portable use in professional environments. The 'Harmonics' rotary switch allows the user to adjust the valve settings in order to obtain the desired sound.

The audio designer: no emotions

For many years now, Ton Giesberts has been a stalwart of the small but tight-knit team that mans the *Elektor Electronics* laboratory. During this time he has become a specialist in audio circuitry. He has been responsible for most of the major audio designs we have published in *Elektor Electronics*, such as the All Solid-State Preamplifier (1989), the Medium Power AF Amplifier (1990), the Output Amplifier for Ribbon Loudspeakers (1992), the Battery-Operated Preamplifier (1997), the Titan 2000 (1999) and the Crescendo ME (2001) to mention but a few. This very issue features his latest creation, which 'naturally' uses transistors: the splendid, compact 'High-end Power Amplifier'.
Given the never-ending dispute between valve and transistor enthusiasts over what sounds better, and in light of the fact that we feature both types of amplifiers in this issue, we naturally would like to hear Ton's thoughts on this subject. We should start by saying that Ton is an absolute stickler for technical quality, so DIY designs with poor performance figures haven't a chance of leaving the audio lab with his approval.

Ton has a clear-cut opinion of valve amplifiers: "*They're simply no good! They colour the sound, and they have poor damping, which degrades the sound. A transistor amplifier does a much better job of keeping the speakers under control, and it adds almost nothing to the sound.*" According to him, that's not actually the fault of the valves, but instead comes from the fact that most valve amplifiers are poorly designed. What's more, the output transformer in a valve output stage forms an additional obstacle to efforts to obtain realistic reproduction. He bases his opinion on the many measurements he has made on valve amplifiers over the years. Ton doesn't have anything against valves as such. He thinks that excellent designs can be made using them, but his experience is limited to transistor designs, so he's reluctant to tackle designing a valve amplifier. "*That's the wrong challenge for me; I definitely belong to the silicon generation.*"
According to Ton, there are some other serious drawbacks to valve amplifiers besides their sound quality. "*In our age of environmental awareness, how can you justify marketing an amplifier with such poor efficiency? And what's more, valves have a relatively short life, and I wonder if they don't contain poisonous substances. Maybe they should be treated as hazardous waste!*"
As you can imagine, Ton would never build or buy a valve amplifier. He thinks it's perfectly possible to design valve amplifiers and transistor amplifiers with equally good characteristics, but it's a lot harder to do so using valves.
"*If someone chooses a valve amplifier for its 'warm' or 'natural' sound, that's fine with me, as long as he or she is happy with it. But I want a system that adds as little as possible to the original sound; the output signal must be the best possible copy of the input signal.*"

The valve microphone: physics versus emotions

By Bernhard Vollmer

The author has been actively involved in quality assurance at Schoeps, a German manufacturer of studio microphones, for nearly two decades. As a supplier to Schoeps, his company Elvo makes the model M222 valve microphone (www.m222.de).

What would music be without emotion? Should our experience, ears and sensitivity take precedence over the technical aspects? Improvements are only conceivable if well-founded understanding provides the basis for further development. Even so, comparative listening tests are always necessary along the way and for judging the end result. But what is the proper way to make comparisons? Human senses are not absolute; anyone who spares himself the trouble of making comparative tests and only listens to a single product will always fall victim to his expectations ('valve sound is warm because valves are warm'). What makes things worse is that people usually don't realise this, so they base their work on preconceived notions.

Generally speaking, there are two questions that always have to be answered:

1. Is there a difference in sound, and if so, what is its nature?
2. Which result is better?

Prolonged listening sessions are necessary to answer these questions. Whether you find something warm or cold also depends on the temperature of your previous surroundings. Ito verify the assumption that 'A' sounds better than 'B', your only option is to directly compare them one after the other without any delay in between, possibly by switching back and forth between the two while the sound is playing. In this way, you can recognise whether there is any difference, how large it is, and what its nature is. Any pause between the two signals poses a large risk of false assessment.

Another important consideration in such 'A'-'B' comparisons is that the levels of the signals being compared must be absolutely identical. People cannot hear a difference of only 0.5 dB in sound level as a difference in loudness. However, the signal with the higher level will unconsciously be judged to sound better.
A comparative test with switchable valve signal paths is pointless if there are significant measurable differences in frequency responses, or if you do not carefully match the signal levels. Measurable differences should be eliminated as much as possible before you put your ears into play. Otherwise you will only hear the prevailing differences, which you could have already demonstrated by measurement.
Many comparative tests of valve and transistor amplifiers have already been conducted, and such tests lead to comparable results if they are done properly. To avoid going into unnecessary detail, these results can be summarised as follows.
At low signal levels, valves are noisier, but the differences in sound are small. At medium signal levels, the differences are scarcely audible. At relatively high signal levels, valves increasingly create harmonic signals, especially at low frequencies. As a result, the sound is no longer neutral as with transistors, but the result can sound more pleasant, particularly with highly impulsive sound events (such as speech and guitar music) As the harmonics of low frequencies are still low frequencies, the resulting sound is often described as being warmer or fuller. If the designers haven't made any gross errors, the differences between valve and transistor amplifiers are much smaller than many people might think
The Schoeps M222 valve microphone is designed for valve enthusiasts and can be used as a fixed or portable microphone in professional environments. The valve settings can be modified by the user to obtain the desired sound.

The valve sound: myth or reality?

By Klaus Rohwer, PhD

The author is a physicist and amateur jazz harmonica player (http://www.klausrohwer.de/).

That famous valve sound, does it really exist? Yes, it does — but it doesn't have to come from a valve! Here we have to make a distinction between devices for *producing* music and devices for *reproducing* music. The first category includes music amplifiers and effect generators, while the second category includes hi-fi and high-end sound systems for the living room. The first type of equipment is optimised for high efficiency (volume) and designed to enable musicians to obtain the sound they want to produce. The second type of equipment should be optimised to reproduce the input signal in an amplified form with the least possible adulteration or distortion. However, many musicians and audio enthusiasts evidently believe that something must glow somewhere in order to generate a 'warmer' sound. I consider this to be purely an *idée fixe*.

Firstly, as regards music production, what we mean here by 'valve sound' is primarily demanded by guitar players and blues harmonica players. The sound of a jazz guitar is just as inconceivable without valve sound as the sound of Chicago blues. It consists of four components:
– soft clipping, instead of the hard clipping characteristic of transistor amplifiers;
– a compression effect, due to a power supply with high internal impedance;
– a 'fuller' sound, due to additional even harmonics produced by a characteristic curve that is asymmetric (even in the small-signal range);
– increased coloration by loudspeaker characteristics, due to a high-impedance output.

Soft clipping results from the gradual bend in the gain characteristic (the ratio of output voltage to input voltage), in contrast to the sudden bend in the characteristic curve of a normal transistor amplifier. Symmetric clipping, whether hard or soft, always produces only odd harmonics in the output signal. However, with soft clipping the amplitude of the higher-order harmonics drops much faster than with hard clipping, with the result that the sound is less 'hard' (**Figure 2**). Similar considerations apply to high-impedance drive (current drive) for the loudspeakers. Signal compression can also be obtained using transistors, although it is more difficult than with valves. In the present age of digital signal processors (DSPs), it can be achieved quite easily, and there are several commercially available effect generators and even complete amplifiers (such as those made by Line 6; see the weblink) that can emulate certain types of amplifiers with extraordinary fidelity, as the author has been assured by several users.

As regards music reproduction, if you want to depart from a straight-line characteristic, which is actually the ideal for any hi-fi or high-end amplifier, you don't necessarily have to use valves, since the same results can be obtained using transistors, as already mentioned.

(040461-1)

Weblinks:

www.m222.de (description of the Schoeps M222 microphone)

www.schoepsclassics.de (older-model Schoeps valve amplifiers)

www.line6.com/products.html (effect processors and software for valve amplifier emulation)

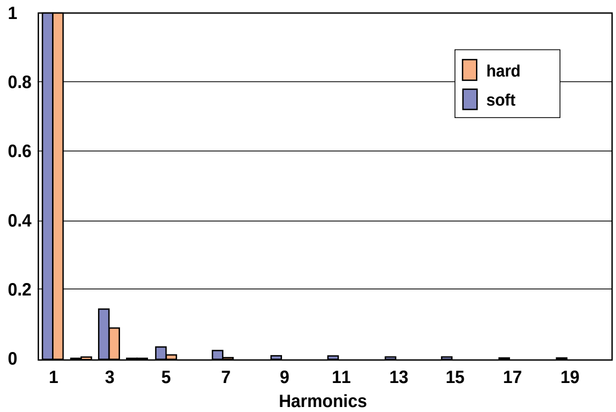


Figure 2. With soft clipping, the amplitude of higher-order odd harmonics in the output signal decreases much more quickly than with hard clipping, so the sound is significantly less harsh.

EL156 Audio Power

Gerhard Haas

Thanks to its robustness, the legendary EL156 audio power pentode has found its way into many professional amplifier units. Its attraction derives not just from its appealing shape, but also from its impressive audio characteristics. We therefore bring you this classical circuit, updated using high-quality modern components.

AMPLIFIER Return of a legend

The EL156 was manufactured in the legendary Telefunken valve factory in Ulm, near the river Danube in Germany. The EL156 made amplifiers with an output power of up to 130 W possible, using just two valves in the output stage and one driver valve. Genuine EL156s are no longer available new at realistic prices, and hardly any are available second-hand. The original devices used a metal valve base which is still available, but a new design using original valves and metal valve bases would anyway be impractical, given the lack of availability at reasonable prices.

Made in China

Fortunately this valve is still being produced in China, using the original Telefunken machines. A normal octal base is now used, with pinout the same as that of the EL84, 6L6, KT88 and similar valves. The devices are still not exactly cheap, but the price is not too unreasonable and the valves are generally supplied with bases included. Comparison with original Telefunken valves shows that the devices are a successful mechanical and electrical copy and are suitable for use in a hi-fi amplifier.

Before we proceed to describe the design, we should first look at a few special features of these valves. In the text box we compare the basic specifications of the EL156 with those of the well-known and widely-used EL34. This information will to a large extent determine the design of the amplifier. In order to obtain sufficient output power, the anode voltage must be at least twice as high as the screen grid voltage. The driver circuit must be designed to cope comfortably with the comparatively low-impedance load presented by the grid leak resistors. The popular ECC83 (12AX7) is ruled out, since it operates at only around 1 mA. The ECC82 (12AU7) audio double triode can be operated with an anode current of 10 mA, and so would appear to be suitable; however, its open-loop gain is only 17, which is not enough to give adequate sensitivity, even before allowing any margin for negative feedback. The ECC81 (12AT7), however, which has an open-loop gain of 60 and which can be operated with anode currents of up to 10 mA, can be used to build a suitably low-impedance circuit.

Two EL156s can be used to produce an output power of 130 W with only 6 % distortion. To improve reliability and increase the life of the valves, however, we have limited the maximum power. A genuine hi-fi output at 100 W with low distortion is better than 130 W at 6 %, especially when there is a large component at the unpleasant-sounding third harmonic.

The whole circuit is built on four printed circuit boards, forming a monoblock. **Figure 1** shows the power supply and the amplifier together. The power supply capacitors are cascaded to filter the high anode voltage in order to obtain the required voltage stability. To supply the relatively high currents required by the screen grids of the EL156s two separate high voltage supplies are produced using bridge rectifiers from two isolated transformer windings ('hi' and 'lo'). Immediately after the rectifiers these supplies are connected in series and individually filtered. Choke Dr1, with a value of 2.3 H, is rated for a current of 0.3 A and filters the anode supply, while Dr2, with a value of 4 H and rated for 0.18 A, filters the screen grid voltage. The driver valve is also powered from the screen grid supply. The screen grid voltage must be well filtered since any hum present on it will be amplified through to the output: the screen grid has some control effect. The values suggested give good filtering and hence low hum. Radial 100 μF/500 V electrolytic capacitors are recommended to make the power supply compact; a working voltage of 500 V ensures adequate margin to give reliable operation even in the event of mains overvoltage. Note the discharge resistors in parallel with the electrolytics. The negative grid bias voltage is provided by a diode and electrolytic capacitor: this voltage is further filtered on the amplifier board.

It is not possible to build an ultra-linear amplifier using the EL156 with a high anode voltage. The same goes for the EL34. The output transformer is therefore connected in such a way that the impedance of the grid connection to the output valve is much lower than in conventional valve circuits, and considerably lower than the maximum permissible value of 100 kΩ. This relaxes the requirements on the tolerance of the valves, and select-on-test of the valves is not required.

Coupling capacitors C9 to C11 have relatively large values. This is needed to ensure that sufficiently low frequencies can be handled in the low-impedance circuit. The input and the phase inversion stages (V1 and V2 respectively) have relatively low anode and cathode resistors. The supply voltage for the input and phase-inversion stages is regulated by Zener diodes D1 to D4. The operating point of V1 is therefore independent of supply voltage fluctuations caused by the output driver stage. R1 and C2 block high frequencies. Capacitor C4, connected in parallel with negative feedback resistor R11, suppresses high frequency oscillations. C3, between anode and grid of V1a, performs the same task. R4 and R6 are effectively in parallel to AC signals, and, in combination with negative feedback resistor R11, set the overall gain. The amplifier is designed with only a moderate amount of negative feedback: this improves the resulting sound.

An E-1220 transformer (Tr1) with a 1:2 turns ratio is fitted at the input to the amplifier. This gives adequate input sensitivity as well as providing isolation. Differential or quasi-differential audio connections are in theory less susceptible to interference and prevent earth loops. Also, the 1:2 ratio gives an extra 6 dB of sensitivity without added noise, leaving a little more margin in hand for negative feedback. The printed circuit board also allows for a

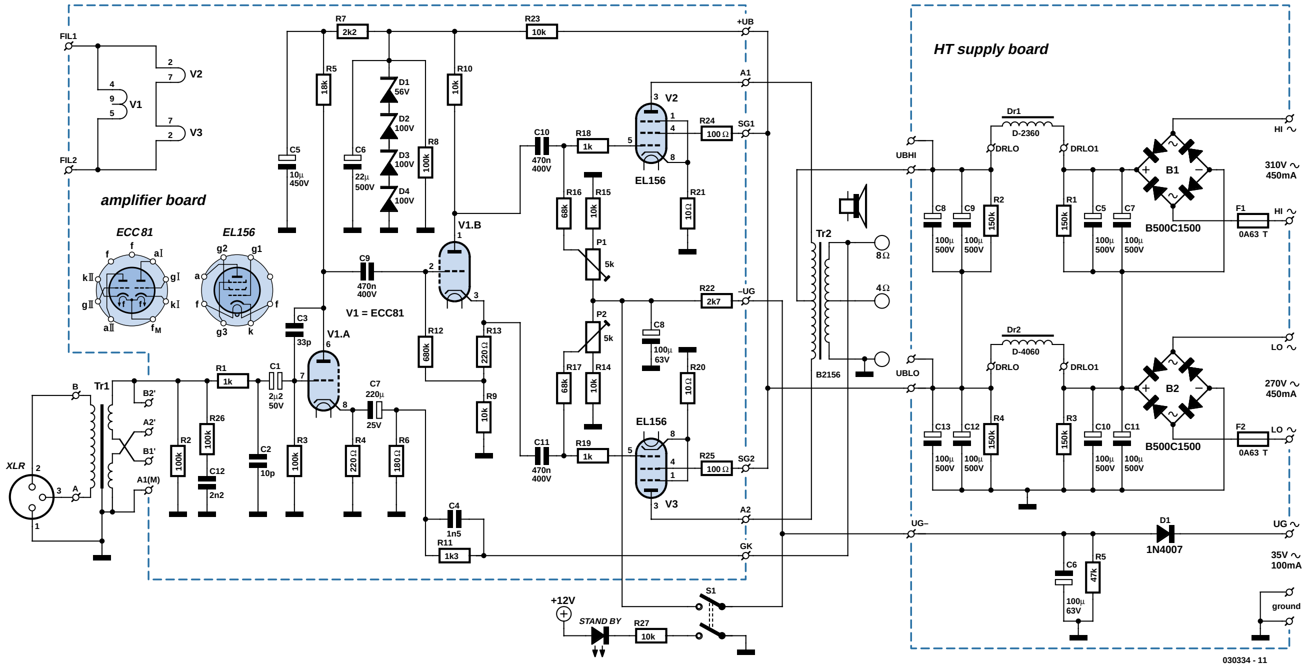


Figure 1. The heart of the valve power amplifier with its output transformer and high-voltage power supply.

1:1 connection, in which case twice the input signal level will be required for full drive. The desired ratio can be selected using wire links. The combination of C12 and R26 compensates for the response of the transformer at higher frequencies.

Stand by me

Valve power output stages are often designed with a stand-by function. This prolongs the life of the output stage valves, normally by switching off the anode supply, while the heater and other voltages remain. On leaving stand-by the amplifier is immediately ready for action.

In view of the high anode voltage, an ordinary switch or relay is not suitable. We take a different approach, shorting out R22 using the stand-by switch, so that the negative grid bias voltage on the output stage valves is raised. Only a very small quiescent current now flows. According to the valve data books this is if anything preferable to switching off the anode supply as prolonged operation with the heater on without applying an anode voltage gradually reduces the emissivity of the cathode. An LED connected to the second pole of the double-pole switch indicates when stand-by mode is active. The LED can be powered from the heater supply.

DC heater supply

To minimise hum a regulated low drop-out DC heater supply is provided, using the familiar 723 voltage regulator and a MOSFET (**Figure 2**). The supply has been designed to minimise losses, and to this end the heater filaments of the two EL156s are connected in series. The ECC81 can be arranged so that it operates from a 12.6 V heater supply. At double the voltage (using 12.6 V rather than 6.3 V) only half the current flows, which means that losses in the bridge rectifier are considerably reduced. With the given component values power losses in T1 will be kept low.

A further trick is used to reduce the voltage drop due to the current limit circuit. The 723 includes a silicon transistor for current limiting, whose base-emitter junction senses the voltage across the current limit sense resistors R4 and R5. Normally the silicon transistor would switch off at

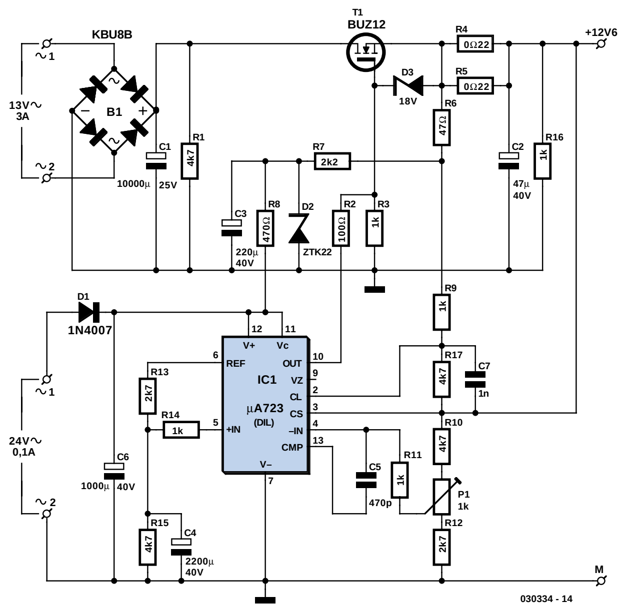


Figure 2. This low drop-out DC regulator provides the heater supply.

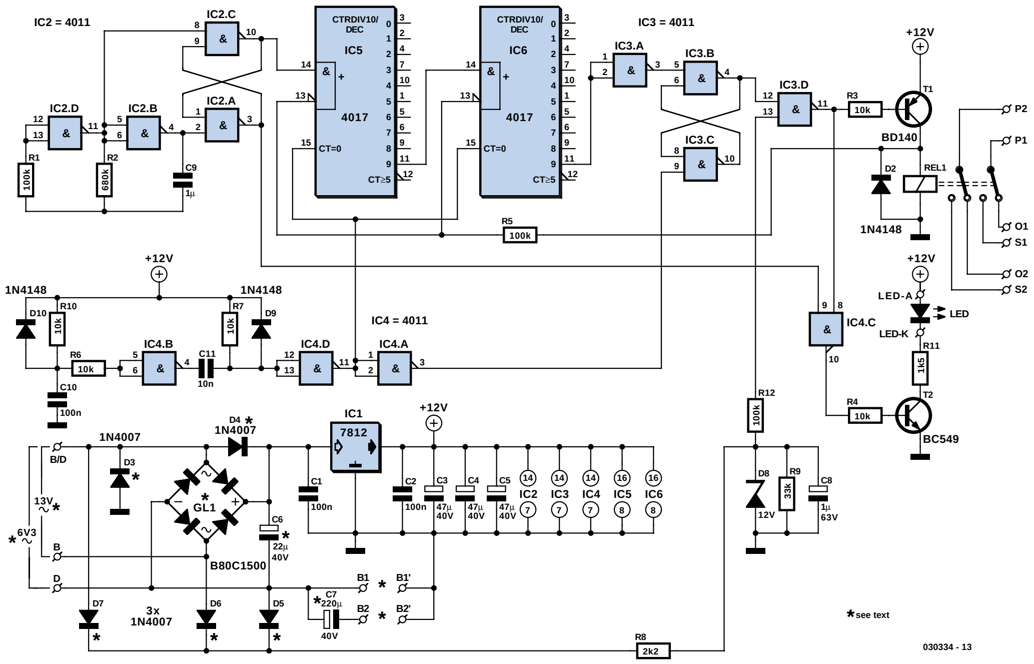


Figure 3. This clever switch-on delay circuit prevents clicks, hum and rumble.

about 0.6 V. Here, however, the base is provided with a stabilised bias voltage from temperature-compensated Zener diode D2 via R7 and R8. This results in a smaller voltage drop across R4 and R5 being needed to trigger current limiting.

The reference voltage produced by the 723 is divided down by R13 and R15. C4 is in principle necessary to filter out any noise on the reference voltage, but could be made considerably smaller. Because of the relatively high value of the capacitor, the voltage at pin 5 rises slowly, providing a 'soft start' to the heater supply. A BUZ12 FET with an R_{DSON} of just 28 mΩ is used for T1.

It is important to ensure that the voltage difference between the cathode and the heater is not too high as otherwise arcing can occur. The negative side of the heater supply **must** therefore be connected to the negative side of the high voltage supply.

Rattle and hum

When the output stage is switched on and while it is warming up the various reservoir and coupling capacitors may charge at different rates. This can give rise to hum and rumble. The circuit in **Figure 3** can suppress these sounds effectively. The normally-closed relay contact shorts out the output transformer for a set time (which the valve output stage can comfortably cope with). Only after a certain interval does the relay pull in, removing the short circuit. This approach avoids putting relay contacts in the signal path. The layout of the printed circuit board allows a relay with two changeover contacts to be fitted so that the circuit can be used for other applications, including with a stereo output stage.

In our case the circuit runs from the 13 V heater supply, from which it will draw a maximum of 200 mA. If the switch-on delay circuit is to be connected to an existing amplifier, the circuit will work equally well from a 6.3 V heater supply winding, as long as there is sufficient spare current capacity. In this case a voltage doubler circuit is used. Depending on the choice of power supply voltage a number of components must be added to or removed from the circuit, as indicated on the component mounting plan and in the parts list.

Switching on

When power is applied C8 immediately starts to charge via R8, taking the pin 13 input of IC3.D high. At the same time C10 charges via R10. As soon as the input threshold voltage of IC4.B is reached its output goes low and, via the high-pass network formed by C11 and R7, generates a brief low pulse at the input to IC4.D. This signal is inverted and then used to reset the 4017 counter to zero, and then inverted again and used to reset the flip-flop formed by IC3.B and IC3.C. The output of this flip-flop at pin 4 thus goes low (if it was not already low). The output of IC3.D is consequently high, T1 does not conduct and the relay does not pull in. The output of the amplifier is therefore short-circuited.

IC2.B and IC2.D form a 1 Hz clock generator, the frequency being determined by C9 and R2. The flip-flop formed by IC2.A and IC2.B makes this clock available to the cascaded counters IC5 and IC6. After 100 counts the flip-flop comprising IC3.B and IC3.C is set via inverter IC3.A. The output of IC3.D then goes low, T1 conducts and the relay pulls in. The short-circuit is removed and the audio signal is now passed through to the loudspeaker. At the same time this high signal is used to disable to counters via R5.

During the switch-on process the pin 8 input to IC4.C is high, and a 1 Hz signal is present on pin 9. The output on pin 10 will therefore also carry the 1 Hz signal, and so the LED flashes. Once the switch-on delay is complete pin 8 goes low, forcing the output of the NAND gate high. The LED now glows continuously.

Switching off

When the amplifier is turned off, there is no longer any voltage present on the transformer. The transformer voltage is monitored continuously by D5 and D6 (13 V operation) or by D5 and D7 (6.3 V operation). If the voltage is not present,

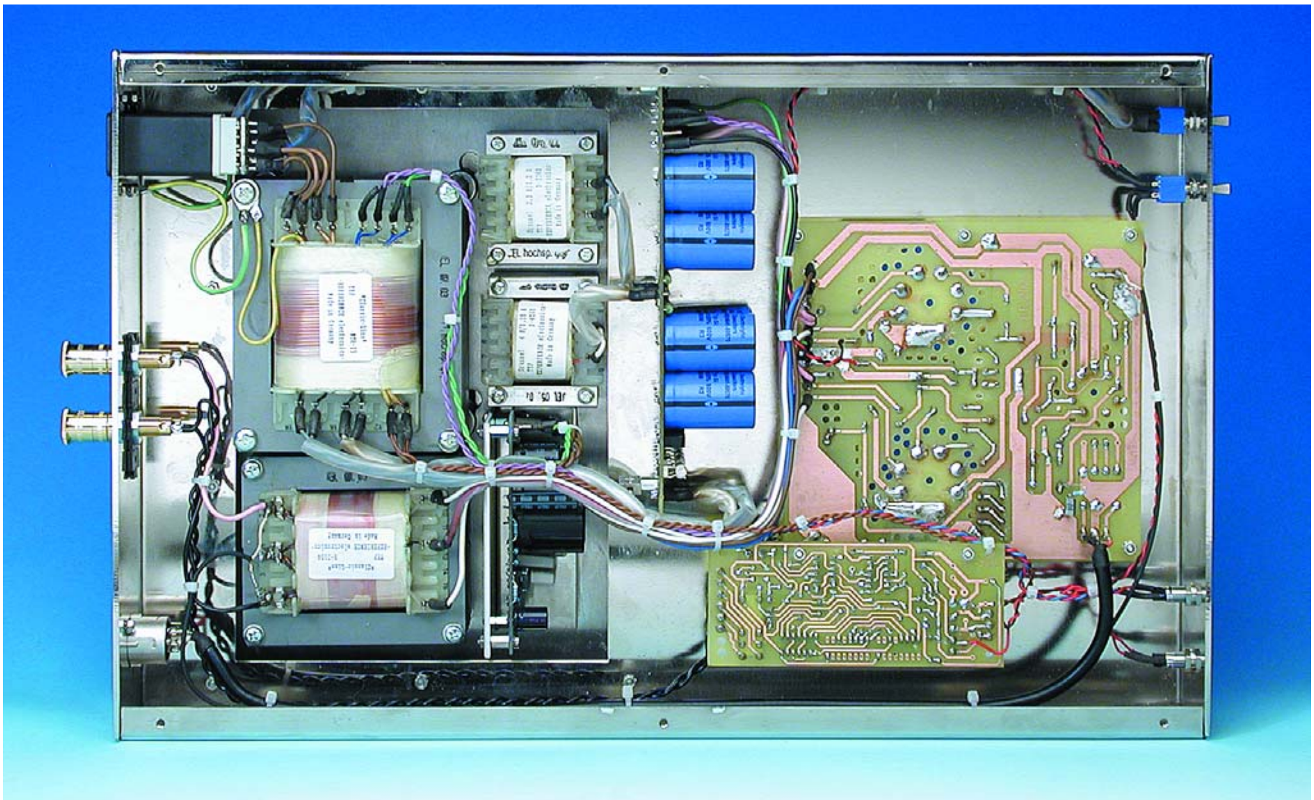


Figure 4. The amplifier seen from below.

the relay drops out immediately. Diodes D1 and D9 ensure that capacitors C10 and D11 discharge quickly. If power is applied again, the whole cycle must be repeated. This ensures that the output is once again muted, ensuring that the unwanted effects mentioned earlier are avoided.

Construction

The monoblock amplifier comprises a total of three printed circuit boards and several wound components fitted into

A valve is a valve is a valve...

Parameter	Symbol	EL34	EL156	Units
Heater voltage	U_f	6.3	6.3	V
Heater current	I_f	1.5	1.9	A
Maximum anode voltage	U_{amax}	800	800	V
Maximum cathode current	I_{kmax}	150	180	mA
Maximum anode power dissipation	P_{Vmax}	25/27,5	40	W
Maximum screen grid power dissipation	P_{vg2}	8	8/12	W
Screen grid current	I_{g2}	11/22	5/25	mA
Maximum screen grid voltage	U_{g2max}	425	450	V
in class AB operation		425	350	
Grid bias voltage	U_{g1}	–39	–24	V
AC grid driving voltage	U_{g1AC}	23	18	V
Transconductance	S	11	13	mA/V
Internal resistance	R_i	15	20	kΩ
Maximum grid leak resistor	R_{g1max}	700	100	kΩ

In some cases quantities have been rounded or guideline values shown, since not all data books exactly agree on all the figures. Values shown after an oblique are at maximum drive.

The EL34 and EL156 are genuine audio power pentodes rather than beam power tetrodes like the 6L6, KT88 or 6550. They are similar to one another, but not identical in all respects. The EL156 requires approximately 27 % more heater power, while offering a maximum cathode current about 20 % higher. Its maximum anode power dissipation is about 60 % higher than that of the EL34. The EL156 also features a higher internal resistance, higher transconductance, higher current and slightly lower grid bias voltage. Maximum drive can also be obtained using a smaller grid

Figure 5. The comforting glow of an EL156 in action.

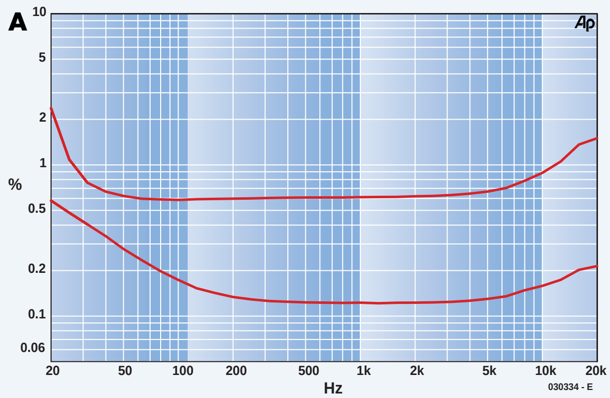
an enclosure as shown in **Figure 4**. Our prototype amplifier was housed in a seamlessly-welded nickel-plated aluminium enclosure polished to a glossy finish. The use of aluminium provides shielding from magnetic interference which mostly originates in the fields produced by the transformers. All the ground connections must be brought together at the amplifier board and bonded to the enclosure at a single point using a bolt and solder tag. If this

driving voltage. Because of these characteristics, greater gain can be achieved using the EL156 than with the EL34, and as a consequence we only need a double triode in the driver stage, despite our high output power.

For both valves a number of details must be observed in high power operation. At higher supply voltages the screen grid voltage must be fixed at a given maximum value. It is also in general necessary to provide a fixed grid bias voltage. The maximum permitted grid leak resistor is considerably lower for the EL156 than for the EL34. The maximum value of the grid leak resistor is specified in the data sheet for each valve. In theory a valve can be driven without dissipating power, but in practice a small grid current flows which must be drained away. Account must be taken of this when designing the driver circuit. The EL156 can only be driven efficiently when the anode voltage is sufficiently high: power has its price! In a triode circuit using class AB push-pull operation dissipation can reach 30 W. The screen grid voltage for the EL156 in high power class AB push-pull operation must be at least 350 V; for the EL34 at least 400 V is required.

On the test bench

Measured characteristics (all measurements taken with an 8 Ω load)				
Parameter	**Conditions**		**Value**	**Units**
Input sensitivity	90 W, 1 % THD+N		1.4	V_{eff}
Input impedance	20 Hz 1 kHz 20 kHz		4 9 1.08	kΩ
Sine wave output power	1% THD+N		90	W
Bandwidth	–3 dB, 1 W		41	kHz
Slew rate	10 µs step		5	V/µs
Signal-to-noise ratio	at 1 W, bandwith = 22 Hz to 22 kHz		88 102	dB BA
Harmonic distortion and noise over 80 kHz bandwidth	1 W	1 kHz 20 kHz	0.12 0.21	%
	50 W	1 kHz 20 kHz	0.6 1.43	
Intermodulation distortion	50 Hz : 7 kHz = 4:1	1 W 50 W	0.5 2.6	%
Dynamic intermodulation distortion	3.15 kHz square and 15 kHz sine wave	1 W 50 W	0.064 0.33	%
Damping factor	1 kHz 20 kHz	2.9 2.3	–	



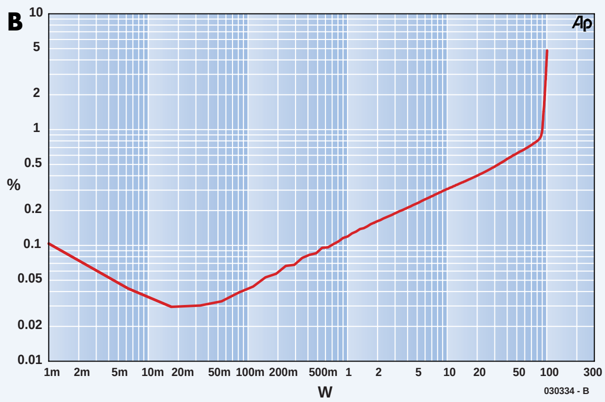


Figure A shows the total harmonic distortion plus noise (THD+N) as a function of frequency when the amplifier is driven at 1 W and at 50 W. The measurement was carried out using a bandwidth of 80 kHz. As is to be expected from a valve amplifier, the distortion increases as the core in the output transformer approaches saturation. This is not a particular disadvantage as the human ear is insensitive to low frequencies and does not find higher distortion levels unpleasant.

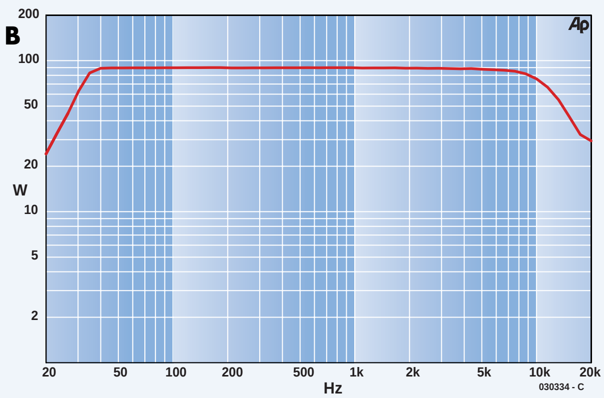
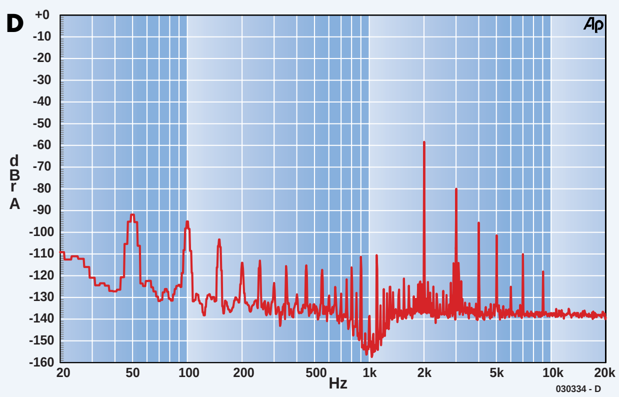
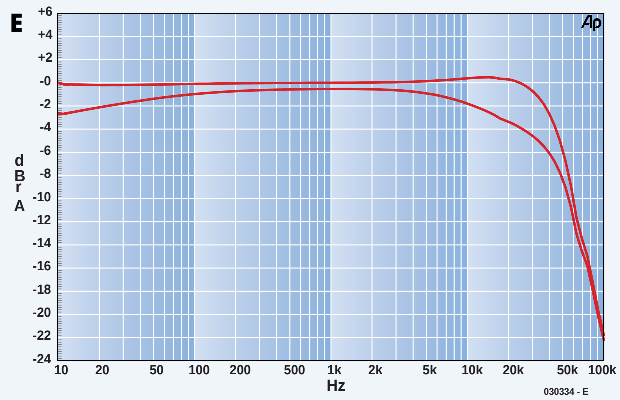


Figure B shows distortion as a function of drive level. The distortion rises from about 50 mW onwards, being dominated by harmonic components. The measurement was taken using a bandwidth from 22 Hz to 22 kHz in order to show more clearly the effect of harmonic distortion at low power levels. At 90 W the amplifier starts to clip.

Figure C shows the maximum power as a function of frequency for a fixed distortion (here 1 %). The bandwidth used for the distortion measurement was 80 kHz. The maximum power starts to fall off towards the upper and lower ends of the amplifier's frequency range. At the upper end of the frequency range the situation is not too bad, since generally less power is required here anyway. Things are different below 40 Hz, where deep tones often demand a lot of power.





A spectrum analysis of the distortion to a 1 kHz sine wave signal at an output power of 1 W is shown in **Figure D**. Almost all the distortion is accounted for by the second harmonic at –58.3 dB. The third harmonic lies at –80 dB, and all the remaining harmonics, along with mains hum, are below –90 dB. The hum component is mainly due to the magnetic field of the transformers (50 Hz component); otherwise the 100 Hz component would be expected to be much more significant.

Finally, **Figure E** shows the effect of the input circuit to the amplifier, where a 4.7 kΩ resistor and a 2.2 nF capacitor are effectively connected in parallel across the secondary side of the input transformer. If the output impedance of the preamplifier is greater than 50 Ω, the input impedance of the power amplifier has a clear effect. The upper curve shows the normal frequency response with an impedance of 20 Ω, while at 600 Ω the frequency response falls off markedly at both the upper and lower end of the frequency range.

is not done, the enclosure will act as an antenna and the amplifier will hum. The high voltages used mean that the enclosure **must** be earthed. Input and output connections must be isolated from the enclosure, or else stray earth currents may arise.

Cable with a cross-section of at least 0.5 mm^2 should be used between the relay contacts and the loudspeaker outputs. Thinner cables have too high a resistance, with the result that rumble can be heard faintly through the loudspeakers during the warm-up phase. The heater connections need cable with a cross-section of 1.5 mm^2, the earth connections cable with a cross-section of 0.75 mm^2, the high voltage connections cable with a cross-section of 0.5 mm^2, and finally the auxiliary supply connections cable with a cross-section of 0.25 mm^2.

Operation is relatively straightforward. First check again that all the components are mounted correctly and that the wiring is correct. Next check the auxiliary voltages, leaving out the fuse in the high voltage supply for now. When mains power is applied the negative grid bias voltage should immediately be present on the valve bases, and can be adjusted using the trimmer potentiometers. For the moment, set the voltage to its maximum negative value. Next check the heater voltage and adjust it to 12.6 V. If voltage can be adjusted over a range of two to three volts, but the value of 12.6 V cannot be reached, then resistor R10 or R12 will need to be changed. Next fit the valves. Shortly, the heater filaments should start to glow, as shown in **Figure 5**.

We can now proceed to test the circuit with the high voltage present. It is absolutely essential that a load resistor rated to at least 150 W must **always** be connected. Do not forget to switch off the unit before fitting the high voltage supply fuse!

Turn the unit back on, and connect an oscilloscope across the load resistor to act as an output monitor. Once the warm-up phase is complete the quiescent current of the output valves can be set. Measure the voltage drop across each cathode resistor, R20 and R21, using a multimeter. Alternately adjust the currents through V2 and V3 for a voltage drop of 450 mV, which corresponds to 45 mA per valve. Next connect a signal generator producing a 1 kHz sinewave to the input, and gradually increase the amplitude of the signal. Observe the output signal on the oscilloscope. It should increase in amplitude without distortion or spurious oscillation until the point where it starts to clip. If the amplifier does have a tendency to oscillate, check that the wiring is correct and that the earthing is sound. If the amplifier goes into large-amplitude oscillation as soon as it is switched on, which can be seen on the oscilloscope as a distorted squarewave with a frequency of approximately 100 Hz, and heard as a hum in the transformer and output valves, switch off immediately. The effect indicates that the output transformer has been connected with the wrong polarity, and anode 1 must be interchanged with anode 2, turning positive feedback into negative feedback. As long as the circuit is switched off immediately there should be no damage to the valves or other components.

(030334-1)

Note

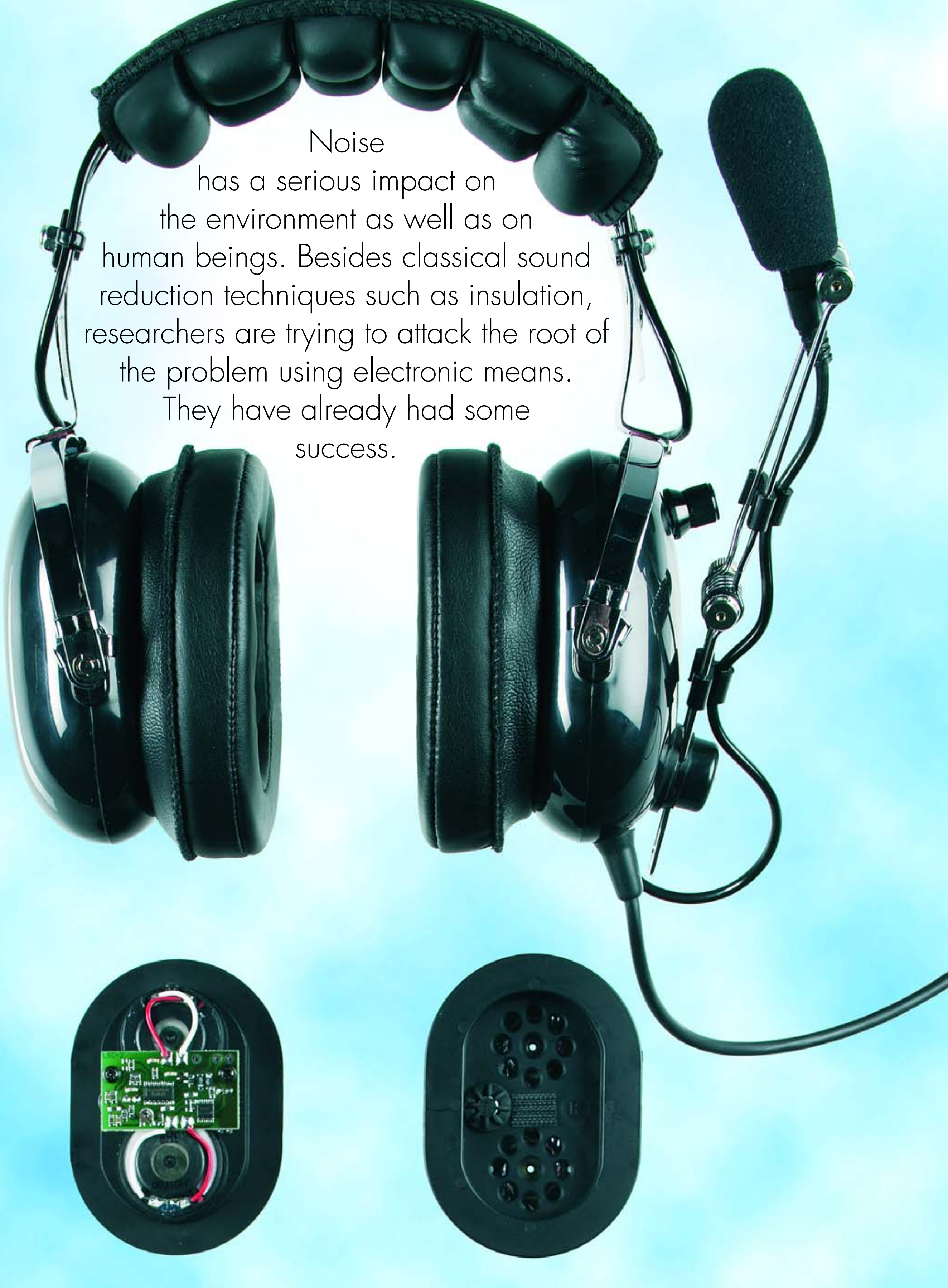
For reasons of space, printed circuit board layouts, component mounting plans and parts lists are only available in electronic form on the Internet, downloadable from

www.elektor-electronics.co.uk.

Ready-made unpopulated printed circuit boards and kits can be obtained from the author (experience.electronics@t-online.de).

ANTI-NOISE

Noise has a serious impact on the environment as well as on human beings. Besides classical sound reduction techniques such as insulation, researchers are trying to attack the root of the problem using electronic means. They have already had some success.



The sound of silence

Sometimes it happens that an underlying physical process is well understood on paper, but the job of translating the theory into the real world takes a long time. One example of this is the patent awarded to the German physicist Paul Lueg in 1933. His idea was astonishingly simple and elegant. In his original patent document (**Figure 1**) he describes how an incoming sound wave can be detected using a microphone, converted to an electrical signal, given a 180 ° phase shift, and mixed with the original signal using a loudspeaker. The result is silence (**Figure 2**). Well, at least in theory the result is silence; the frequency response of the microphone, loudspeaker and electronics along with other effects such as temperature and humidity mean that in practice things are harder than that.

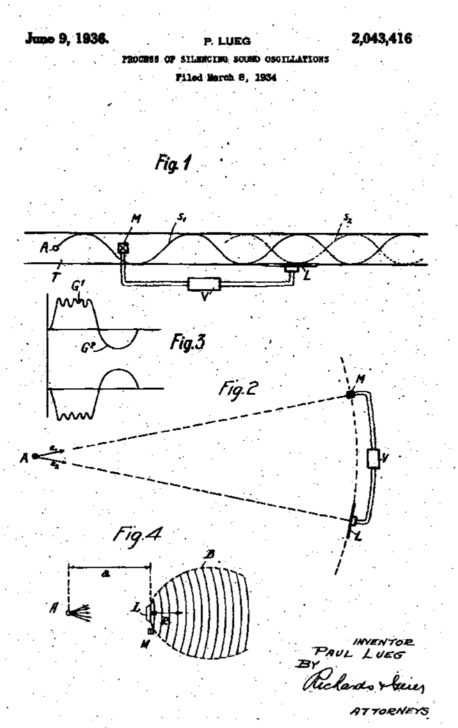


Figure 1. Facsimile of the patent document by Paul Lueg.

Anti-noise in headphones

Noise has a serious impact on the environment as well as on human beings. It causes tiredness, loss of concentration, nervousness and irritability, and can lead to permanent hearing damage. Passive ear protection reduces noise especially in the middle and upper frequency ranges, but is not adequate at lower frequencies (**Figure 3**).

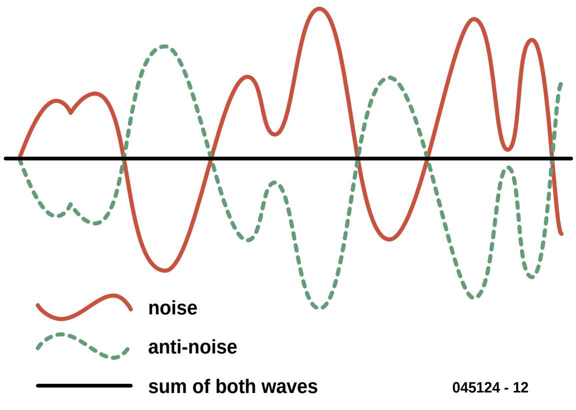


Figure 2. How anti-noise systems work (but sadly, only in theory!)

Is it possible to use Paul Lueg's patent in the battle against noise? Indeed it is: the most widespread and well-known application is in air travel, where active noise compensation is used in headphones for crew and passengers (first class only, of course!). The system is known as ANR (active noise reduction) or ANC (active noise cancellation). In the USA at least, there is considerable consumer demand for ANR headphones to allow music to be heard undistorted in noisy surroundings. The market fulfils this demand, with units available from thirty dollars from such well-known brands as Philips, Sony, Aiwa, Maxell, Bose, Shure, Koss, Altec-Lansing and Jensen. Panasonic also offers in-ear headphones with ANR. Sennheiser (www.sennheiser.com) is among the leading suppliers of headsets and headphones featuring ANR, with their NoiseGard™ system. An electret microphone capsule is fitted in each headphone unit, along with a negative-feedback circuit and a transducer system. The sound received by the microphone, which includes the wanted signal as well as the interference, is amplified and the wanted signal filtered out. The interference signal is processed, phase shifted by 180 °, and added to the wanted signal. The result is sent to the transducer system. Superimposing the anti-phase signal considerably reduces the amplitude of the interference, whereas the wanted signal, which has not passed through the compensation circuit, is unaffected.

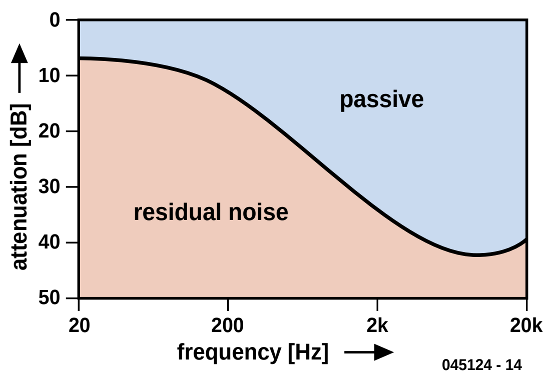


Figure 3. Attenuation in closed headphones without ANR.

NoiseGard™ headsets and headphones are available in closed and open versions. The open versions are suitable for compensating for the noise level and spectrum found in business jets or commercial aeroplanes. More than 10 dB of compensation is achieved over the range from 400 Hz to 1000 Hz (**Figure 4**). This allows the

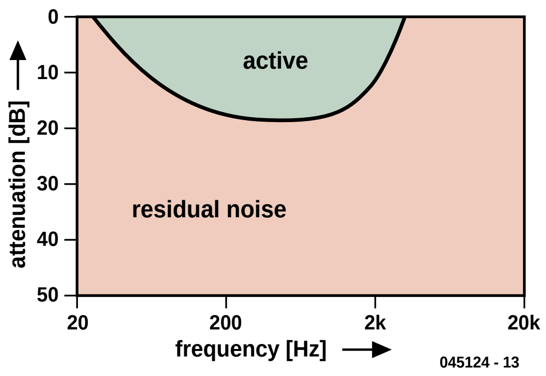


Figure 4. An active system with open headphones works particularly well in the middle frequency range.

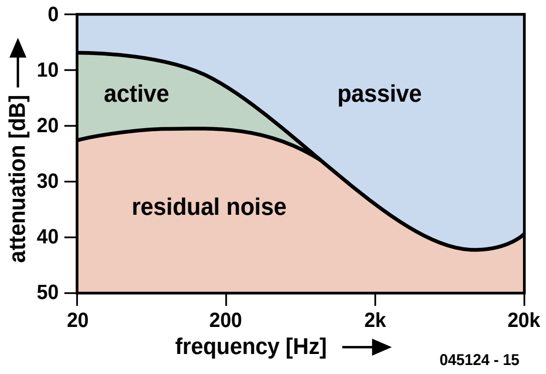


Figure 5. With closed headphones ANR also attenuates the lower frequencies.

volume level of speech radio communications to be reduced, without compromising their intelligibility. Thanks to the open form of the headphones, voices and alarm signals in the cockpit can be heard practically without distortion. Noise, on the other hand, is considerably attenuated. Lightweight open headphones offer excellent comfort to the wearer, which is particularly important for long journeys.

If passive ear protectors, such as earplugs or closed headphones, are combined with active noise compensation, noise in the frequency range from 25 Hz to 500 Hz can be attenuated by more than 25 dB (**Figure 5**). The total effect of active and passive noise compensation is around 30 dB over the entire audio spectrum. Closed systems are particularly suitable for aircraft with a high noise level in the cockpit, such as helicopters and military aircraft, as well as for most propeller aircraft. These closed systems can be retrofitted with ANR kits to add active noise compensation. The picture at the beginning of the article shows a headset together with an ANR module from an ANR kit. As can be seen from the photograph, each module consists of two headphone loudspeakers with an electret microphone capsule positioned between the two loudspeakers. On the rear side of the module is a printed circuit board carrying SMD components. On closer inspection, the two ICs reveal themselves to be a TL064C (quad FET opamp) and an MC34119 (low power audio amplifier). Superficially the circuit looks like a relatively simple analogue ANR circuit, but tests and experience show that it is highly effective. Kits of this type are available from outlets dealing in aircraft equipment. The modules and headset pictured are available from Büscher-Flugversand, 43466 Wolfhagen, Germany (www.flugversand.de, website in German only).

Active noise control

In the applications of noise compensation that we have described the spatial volume over which noise is reduced is narrowly limited. Under these conditions ANC is at its most effective. In larger spaces and in the open air it is

Figure 6. Typical low-cost ANR headphones for listening to music in noisy environments. The electronics box is connected to the signal source (personal stereo etc.) and linked to the headphones, which also include two microphones, using an 8-core screened cable.

Anti-noise online

We looked at the specialist headphones website **HeadWize** in our 'Electronics Online' feature in the October 2003 issue of Elektor Electronics. This site is a rich source of information on ANR headphones. As well as an article on the various **basic principles** of operation [1] that is well worth reading, there is also a complete guide to constructing an analogue **ANR circuit** [2].

Somewhat more complex is the DSP **application note** 'Adaptive Active Noise Control for Headphones Using the TMS320C30 DSP' by Texas Instruments [3].

A good overview of the subject, including lots of information, ideas and links can be found on the ANC pages by **Signal Systems Corp.** [4] and the ANC FAQ site by **Christopher E. Ruckman** [5]. **Edge Consulting Group** [6] offers a **news site** for acoustics, vibrations and signal processing with a page on 'Active Noise Control'.

Anyone considering adding an **ANR kit** to a passive headset will find detailed information at **ANR-Headsets.com** [7] and **Headsets Inc.** [8].

Web addresses:

[1] http://headwize.com/tech/anr_tech.htm
[2] http://headwize.com/projects/showfile.php?file=noise_prj.htm
[3] www-s.ti.com/sc/psheets/spra160/spra160.pdf
[4] www.signalsystemscorp.com/ancindex.htm
[5] http://users.erols.com/ruckman/ancfaq.htm
[6] www.ecgcorp.com/
[7] www.anr-headsets.com/
[8] www.headsetsinc.com/

much harder to achieve noise compensation. There has, however, been some progress in this direction. In aviation, noise compensation has been used in the cockpit since the mid-90's, especially in turboprop aircraft and, more recently, in a business jet. The first commercial aircraft to be fitted with such a system is the Swedish Saab 2000. The system, by Ultra Electronics Ltd. (www.ultra-quiet.com), uses 72 microphones in the cabin to continuously analyse the noise spectrum. A powerful microprocessor system processes the signals and generates a phase-shifted correction signal that is emitted though 36 loudspeakers. ANC is particularly useful for reducing propeller noise. A similar application area is reducing the noise produced by large ventilation fans in climate control equipment. Industrial motors can also be equipped with ANC. In the automotive sector ANC is still at the research stage, but it would of course be interesting to be able to exploit the existing loudspeaker arrangements.

It would make more sense to try actively to reduce the noise at the point of production rather than at the receiver. This is an area of current research, but the practical results have so far been few. **Figure 7** shows the construction of a research project at the German Aerospace Centre aimed at reducing the noise from an aircraft engine turbine. Experts calculate that a reduction of between 10 dB and 20 dB in noise level can be achieved. This corresponds to between one half and one quarter of the original volume.

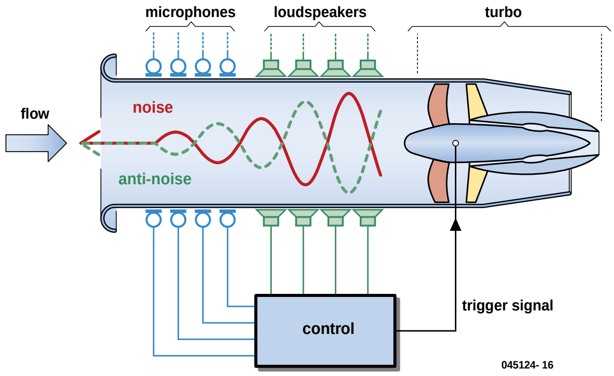


Figure 7. Active noise control at source: research at the German Aerospace Centre.

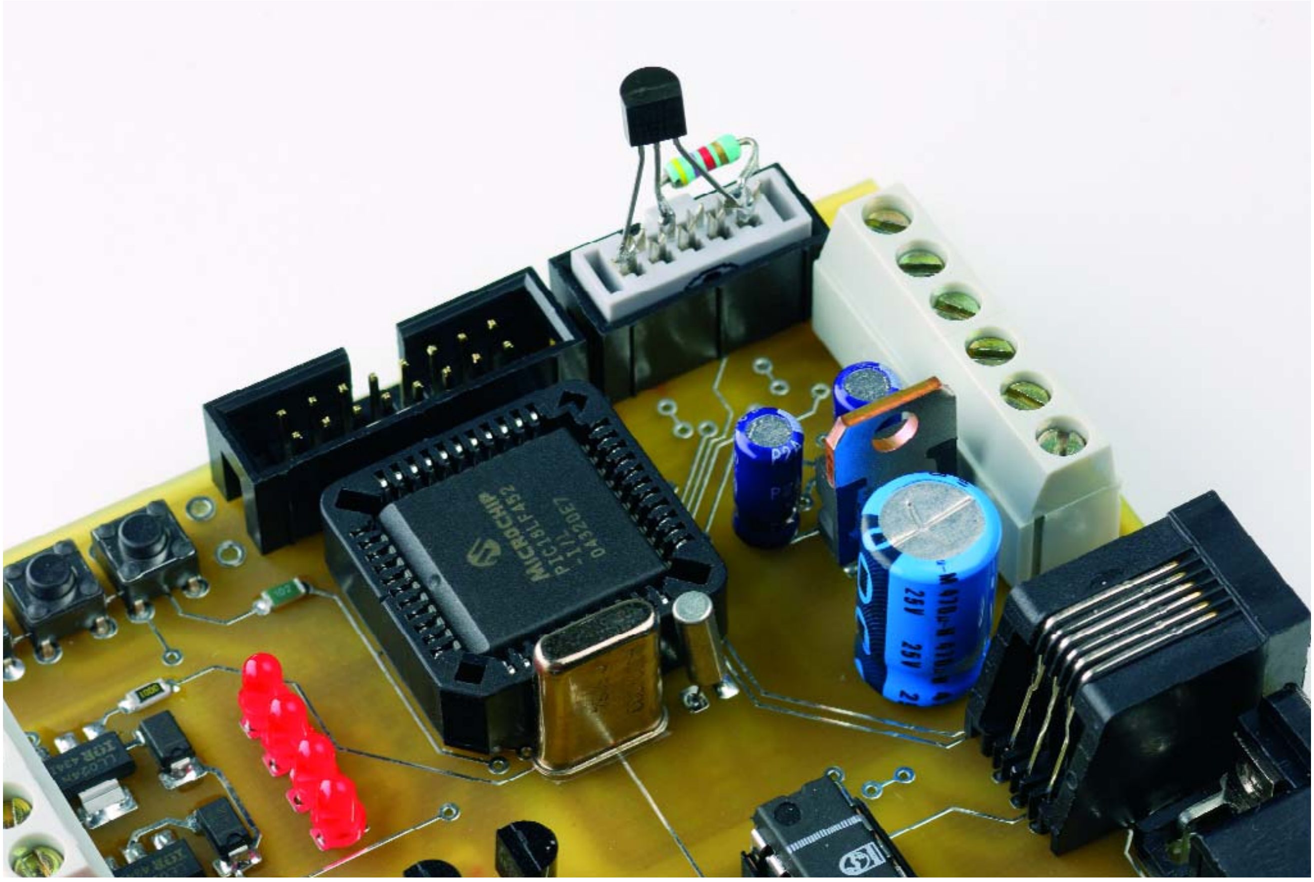
The ANC anti-noise technique can be applied not only to noise, but also to vibration. The techniques of AVC (active vibration control) and ASAC (active structure acoustic control), which aims to reduce noise produced as a result of vibration, look to have a bright future.

(040440-1)

Remote Temperature Logger for PIC18F Board

Using a DS1820 1-wire temperature sensor

Peter Moreton



In this short article, we describe how to bit-bang a DS18B20 or DS18S20 temperature sensor device using PIC firmware written in 'C', and output the temperature values to the RS232 port, for logging on a PC, all using the PIC18Flash Development Board published last month.

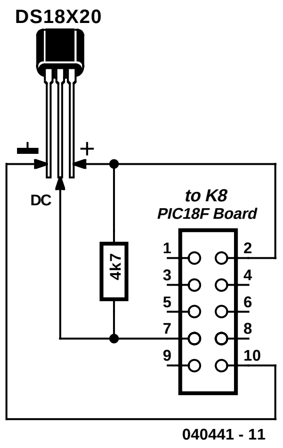


Figure 1. Circuit diagram of the Remote Temperature Logger hardware — all parts are connected to K8 on the PIC18F Flash Development Board.

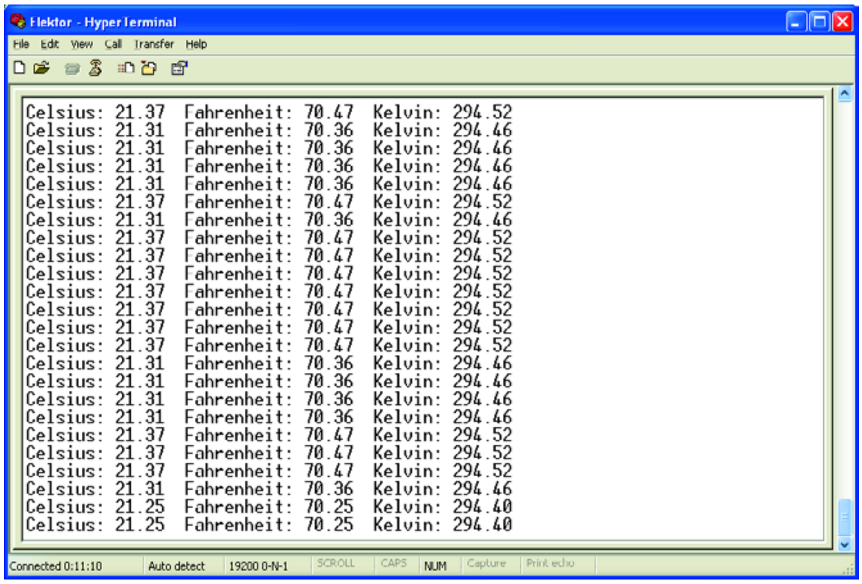


Figure 2. Screendump showing the Temperature Logger software in action.

The PIC18Flash development board described in *Elektor Electronics* January 2005 includes a header 'K8' that delivers the appropriate processor pins for SPI / Microwire, I^2C and 1-Wire communications to external devices. The SPI/Microwire and I^2C protocols are supported by dedicated PIC hardware modules, and are quite easy to use as the hardware can be accessed using standard C18 library functions.

The Dallas/Maxim '1-Wire' bus does not benefit from a dedicated PIC hardware module and therefore must be implemented by 'bit-banging' the appropriate MCU pin high and low in accordance with the 1-Wire protocol specification.

The 1-Wire bus

The Dallas / Maxim 1-Wire bus uses a single data line (hence '1-Wire') to transmit and receive data. Each 1-Wire device must also be connected to a common ground, and optionally to a power supply. If the power supply line is omitted, the 1-Wire device can be 'parasitically powered' from the data line. The 1-Wire bus is normally held 'high' using a 4k7 resistor to Vdd, and the devices pull the bus low and release the bus to go high, in order to signal data bits.

Pin RA4 of the PIC18F452 is particularly suited to the 1-wire protocol, as it is an open-drain pin, enabling the bus to be 'released' , and pulled high by the 4k7 bus pull-up.

The bus protocol works by defining a number of signal types: a reset pulse, presence pulse, write 0, write 1, read 0 and read 1. Each signal has timing parameters which must be strictly adhered to, and therefore an attribute of all 1-wire systems is the ability to generate accurate 'time-slots'. Readers wishing to fully understand 1-Wire signalling are referred to the in-depth documentation available from the Dallas/Maxim website.

DS18x20 Temperature Sensors

There are a number of similar devices in the DS1820 lineup; we have developed and tested PIC firmware for the DS18S20 giving 9-bits / 0.5 DegC resolution and the DS18B20 giving up to 12 bits/0.0625 DegC resolution. Both parts are accurate to within 0.5 DegC.

Either device may be used, by editing the DS1820.C source file appropriately to call one of the two available functions:

```
Celsius =
Read_Temperature_DS18S20();
// use DS18S20 device
```

or

```
Celsius =
Read_Temperature_DS18B20();
// use DS18B20 device
```

The pre-compiled HEX file included in the free download for this article (no. **040441-11.zip**) assumes the higher resolution DS18B20 device is connected.

Circuit Description

Figure 1 shows a Dallas DS18S20 or DS18B20 temperature sensor in a TO92 package and a 4k7 pull-up resistor connected to the K8 header, as follows:

– K8 pin 2 to DS1820 pin 3, V_{dd};
– K8 pin 10 to DS1820 pin 1, Ground;
– K8 pin 7 to DS1820 pin 2, DQ;
– Fit a 4k7 pull-up between DQ and V_{dd}.

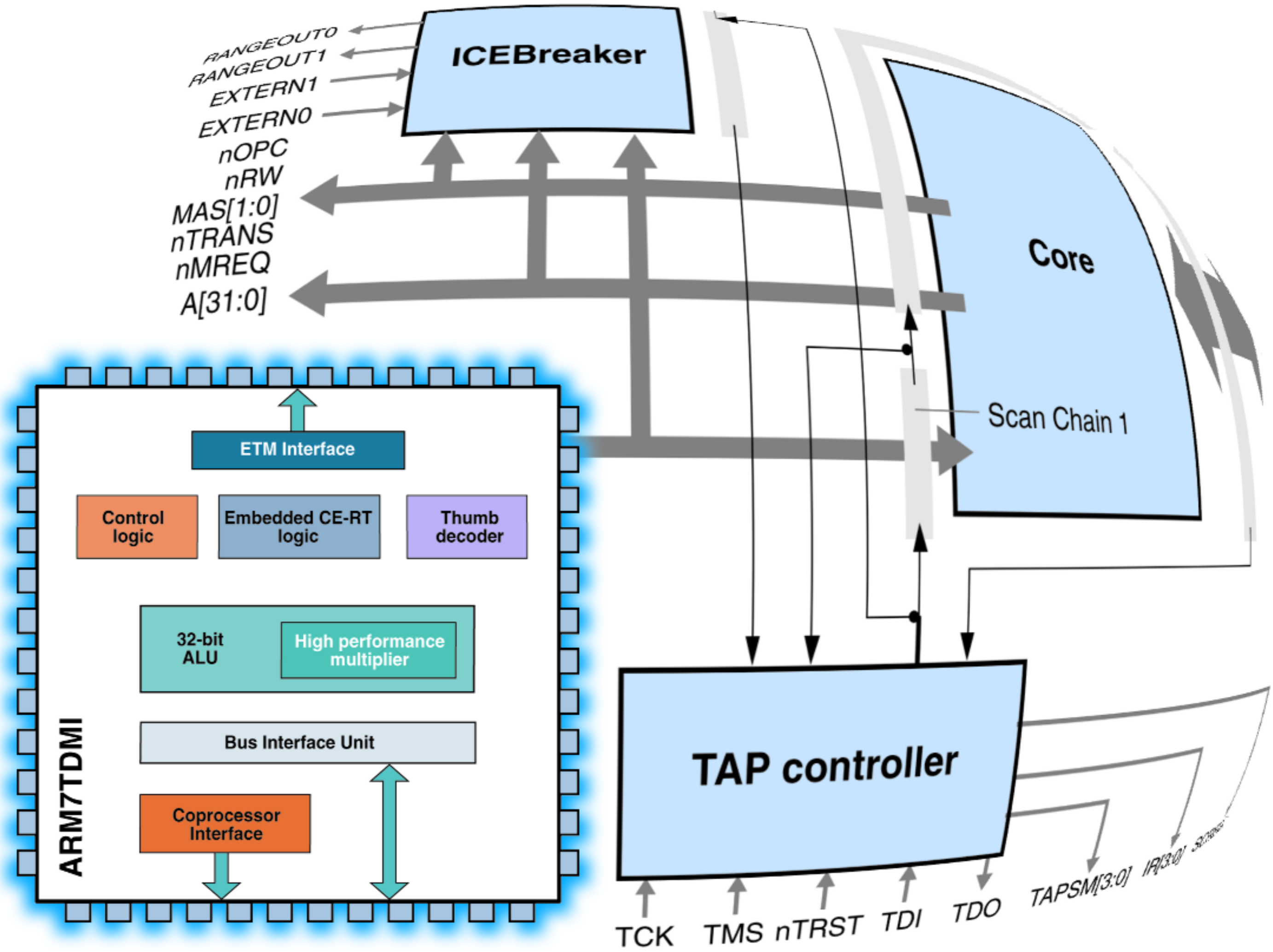
It is acceptable to extend the interconnecting cable up to several meters to allow for remote measurement, and we suggest that for external applications the DS1820 be protected from the elements using heatshrink tubing over a waterproof sealant.

(040441-1)

LPC210x 'ARMee'

Part 1: an ARM processor survey

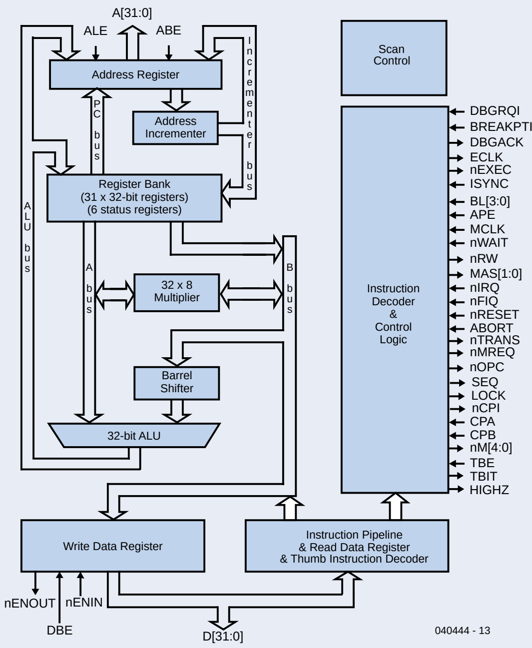
Tony Dixon



In the first instalment of a three-part article we will look at recent developments in 32-bit ARM based microcontrollers, concentrating on those devices that are available, do not cost an arm and a leg or require a major investment in surface mount soldering equipment!

DEVELOPMENT BOARD (1)

Overview of the ARM7TDMI core



The ARM7 core is a 32-bit Reduced Instruction Set Computer (RISC). It uses a single 32-bit bus for instruction and data. The length of the data can be 8, 16 or 32 bits and the length of the instruction word is 32 bits.

What Does TDMI™ mean?
The ARM7TDMI is an ARM7 core with 4 additional features identified with letter codes, as follows

T: support for the Thumb (16 bit) instruction set.
D: support for debug
M: support for long multiplies
I: include the EmbeddedICE module to support embedded system debugging.

Thumb mode (T)
An ARM instruction is 32-bits long. The ARM7TDMI processor supports a second instruction set that has been compressed into 16-bits, the Thumb instruction set. Faster execution from 16-bit memory and greater code density can usually be achieved by using the Thumb instruction set instead of the ARM instruction set, which makes the ARM7TDMI core particularly suitable for embedded applications.
However the Thumb mode has two limitations and these are: Thumb code usually uses more instructions for the same job, so ARM code is usually best for maximising the performance of the time-critical code.
The Thumb instruction set does not include some instructions that are needed for exception handling, so the ARM code needs to be used for exception handling. See ARM7TDMI User Guide for details on the core architecture, the programming model and both the ARM and ARM Thumb instruction sets.

Long multiple (M)
A 32-bit Multiplier function qualifies the core for complex arithmetic tasks usually performed by a DSP. The ARM7TDMI instruction set includes four extra instructions that perform 32-bit by 32-bit multiplication with 64-bit result and 32-bit by 32-bit multiplication-accumulation (MAC) with 64-bit result.

Debugging (D)
A special hardware extension allows for Debugging within an application. This is made possible by means of a boundary-scan cell array around the core driven by a JTAG port and a TAP controller.

EmbeddedICE (I)
The EmbeddedICE extends the debugging functions and this module contains the breakpoint and watch point registers that allow the code to be halted for debugging purposes. These registers are controlled through the JTAG test port with the aid of software debugging tools running on a computer. When a breakpoint or watch point is encountered the processor halts and enters debug state. Once in a debug state, the processor registers may be inspected as well as the Flash/EE, SRAM and the Memory Mapped Registers

ARM stands for *Advanced RISC Machine*, where RISC means *Reduced Instruction Set Computer*. The ARM 32-bit architecture has been around for a number of years and has been used in products where low power consumption is essential such as for mobile phones and PDA's.
Its 32-bit core is available in several guises including the ARM7, ARM9, ARM10 and the recently announced ARM11, each of which offers enhanced levels of computing power.

The ARM was usually only available as a microprocessor device, where it required external program and data memory to complete a system. However several companies are now offering a 32-bit ARM based microcontroller with sufficient memory options for them to be considered worthy microcontroller alternatives. **Table 1** shows a selection of ARM based microcontrollers from three companies, Analog Devices, OKI Semiconductors and Philips Semiconductors. Other companies such as Atmel, NetSilicon, Samsung and TI offer similar devices

Table 1. ARM processor comparison matrix.

Device	Package	RAM	Flash	Clock	I/Os	UARTs	SPI
AduC7020	40-pin LFCSP	8 kB	62 kB	0.3-45 MHz	14	1	1
AduC7021	40-pin LFCSP	8 kB	62 kB	0.3-45 MHz	13	1	1
AduC7022	40-pin LFCSP	8 kB	62 kB	0.3-45 MHz	13	1	1
AduC7024	64-pin LQFP	8 kB	62 kB	0.3-45 MHz	30	1	1
AduC7025	64-pin LQFP	8 kB	62 kB	0.3-45 MHz	30	1	1
AduC7026	80-pin LQFP	8 kB	62 kB	0.3-45 MHz	40	1	1
AduC7027	80-pin LQFP	8 kB	62 kB	0.3-45 MHz	40	1	1
LPC2104	48-pin TQFP	16 kB	128 kB	0-60 MHz	32	2	1
LPC2105	48-pin TQFP	32 kB	128 kB	0-60 MHz	32	2	1
LPC2106	48-pin TQFP	64 kB	128 kB	0-60 MHz	32	2	1
LPC2114	64-pin LQFP	16 kB	128 kB	0-60 MHz	46	2	2
LPC2124	64-pin LQFP	16 kB	256 kB	0-60 MHz	46	2	2
LPC2212	144-pin LQFP	16 kB	256 kB	0-60 MHz	112	2	2
LPC2214	144-pin LQFP	16 kB	256 kB	0-60 MHz	112	2	2
LPC2119	64-pin LQFP	16 kB	128 kB	0-60 MHz	46	2	2
LPC2129	64-pin LQFP	16 kB	256 kB	0-60 MHz	46	2	2
LPC2194	64-pin LQFP	16 kB	256 kB	0-60 MHz	46	2	2
LPC2292	144-pin LQFP	16 kB	256 kB	0-60 MHz	112	2	2
LPC2294	144-pin LQFP	16 kB	256 kB	0-60 MHz	112	2	2
ML674001	144-pin LQFP	32 kB	256 kB	1-60 MHz	42	2	1
ML674001	144-pin LQFP	32 kB	512 kB	1-60 MHz	42	2	1
ML675001A	144-pin LQFP	32 kB	256 kB	1-60 MHz	42	2	1
ML675001A	144-pin LQFP	32 kB	512 kB	1-60 MHz	42	2	1

but these are usually available in larger BGA (ball grid array) packaging only, making them less suitable for hand prototyping. So what's available on the ARM market we can actually obtain and handle? Let's have a look what a number of major manufacturers in the ARM arena have on offer.

Analog Devices

Analog Devices (www.analog.com) is not a name normally associated with microcontrollers, however they are about to change this view by creating a microcontroller product with a range of precision analogue interfaces. *Elektor Electronics* (leading the way!) have already published a series of articles on Analog Devices AduC812 8052 based controllers (Ref. 1). Analog Devices have released an updated range using an ARM7TDMI as the computing engine.

The ADuC702x family of devices from Analog Devices integrates a 32-bit ARM7TDMI core with a 12-bit data converter that can have up to 16 channels supporting one million samples/sec. The ADuC702X devices also feature up to four 12-bit DACs with a precision bandgap reference sensitive to 10 ppm/°C. Other peripherals include a comparator, a small programmable-logic array (PLA) for glue logic, an on-chip temperature sensor (±3°C) and a three-phase 16-bit PWM generator. Of these peripherals the programmable-logic array is the most interesting to find on a microcontroller.

A JTAG interface is provided for debugging the chip, while the UART can be used to program the Flash memory *in-situ*.

An on-chip oscillator will drive the ADuC702x at speeds of up to 35 MHz and is 2% accurate. An external clock is required to run at speeds up to the 45- MHz limit. Memory options include both a 32-kByte Flash memory for the ADuC7024 and a 62-kByte Flash memory for the ADuC7026 and all include 8 kBytes of RAM. Packaging options range from a 6×6-mm, 40-lead CSP, a 64-pin LQFP and an 80-pin LQFP. These 3-V devices can operate within a temperature range of –40 to +85, or at extended temperatures up to +105, or +125°C.

Analog Devices offer a low-cost quick start development system called QuickStart, which includes a power supply, cables, evaluation board, JTAG emulator and software-development tools from Keil Software and IAR Systems. The QuickStart Development System sells for $249 and is available directly from Analog Devices.

Of the devices offered, the 64-pin and 80-pin devices

I2C	CAN	Timers	PWM	ADC	DAC	Notes
2	-	2	-	5 x 12-bit	4 x 12-bit	PLA, Temp Sensor
2	-	2	-	8 x 12-bit	2 x 12-bit	PLA, Temp Sensor
2	-	2	-	10 x 12-bit	-	PLA, Temp Sensor
2	-	2	3	10 x 12-bit	2 x 12-bit	PLA, Temp Sensor, 3-Phase
1	-	2	3	12 x 12-bit	-	PLA, Temp Sensor, 3-Phase
1	-	2	3	12 x 12-bit	4 x 12-bit	PLA, Temp Sensor, 3-Phase
1	-	2	3	16 x 12-bit	-	PLA, Temp Sensor, 3-Phase
1	-	4 x 16-bit	6-ch	-	-	
1	-	4 x 16-bit	6-ch	-	-	
1	-	4 x 16-bit	6-ch	-	-	
1	-	4 x 16-bit	6-ch	4 x 10-bit	-	
1	-	4 x 16-bit	6-ch	4 x 10-bit	-	
1	-	4 x 16-bit	6-ch	8 x 10-bit	-	with external memory interface
1	-	4 x 16-bit	6-ch	8 x 10-bit	-	with external memory interface
1	2	4 x 16-bit	6-ch	4 x 10-bit	-	
1	2	4 x 16-bit	6-ch	4 x 10-bit	-	
1	4	4 x 16-bit	6-ch	4 x 10-bit	-	
1	2	4 x 16-bit	6-ch	8 x 10-bit	-	with external memory interface
1	4	4 x 16-bit	6-ch	8 x 10-bit	-	with external memory interface
1	-	7 x 16-bit	2-ch	4 x 10-bit	-	with external memory interface
1	-	7 x 16-bit	2-ch	4 x 10-bit	-	with external memory interface
1	-	7 x 16-bit	2-ch	4 x 10-bit	-	external memory i/f, 8K cache
1	-	7 x 16-bit	2-ch	4 x 10-bit	-	external memory i/f, 8K cache

are probably the most usable by our readership for their prototyping ease.

Philips

Of all the companies offering ARM microcontrollers Philips (www.semiconductors.philips.com) seem to be the company pushing the ARM microcontroller the most and have already released an extensive range of microcontrollers based on a 32-bit ARM7TDMI-S core (see **inset**).
Philips initially offered the LPC210x which featured 16 to 64 kB of RAM dependant on the device, together with 128 kB of Flash memory and all operating at 60 MHz. Other peripherals include two UARTs, SPI and I²C interfaces, 6-channel PWM and 32-bit digital I/O port. All of which are fitted in a small 48-pin LQFP package!
All three controller chips are based on a common system architecture approach which offers the same memory map, vectored interrupt controller and similar peripheral complements. Also common to them are the same Flash programming and updating mechanism, JTAG debugging and emulation facilities.
These devices operate from 1.8 V for the core CPU functions and 3.3 V for the I/O and peripherals, with the general I/O being 5 V tolerant.

Philips has extended the LPC21xx family to include new devices packaged either in a 64-pin or a 144-pin LQFP. These new family members offer larger Flash memory options, an additional SPI interface and additional digital I/O lines. They also included either a 4 or 8 channel ADC with 10-bit resolution, 2- or 4-channel CAN bus interface and the option of an external memory interface on the larger 144-pin devices.
The LPC210x devices have a number of development and evaluation boards from companies such as Hitek, Keil, IAR and Nohau.
According to press releases from Philips we can expect future members of the LPC21xx and LPC22xx family to include Ethernet, USB, and 802.11 capabilities. Something to look forward to!

OKI Semiconductors

OKI Semiconductors (www.oki.com) are a Japanese company who offer a broad range of ICs and have been providing 32-bit ARM-based solutions for a number of years. Oki have extended its microcontroller portfolio by introducing a new series of general-purpose 32-bit microcontrollers based on an ARM7TDMI core.
These new two lines consist of the ML674001 and the

ML675001 series. The ML674001 series comprises of three products: the ML674001, the ML67Q4002 and the ML67Q4003. While the ML675001 series consisting of the ML675001, the ML67Q5002 and the ML67Q5003. The ML674001 and ML675001 are ROMless parts.

The ML67Q4002/3 and ML67Q5002/3 microcontrollers offer large Flash memory options up to 512 kB and 32 kB of RAM. Other peripherals include 1 × system timer, 6 × general purpose timers, 2 × PWM, watch dog timer, general purpose I/O ports, ADC converters and 2 × DMA channels. Communications are provided 2 × UARTs; one UART is an industry standard 16550A and has 16 bytes FIFO for both send and receive, with the other having no FIFO; an I^2C and SPI interface. The chips also include an external memory interface that features a SDRAM controller allowing for ROMs (including Flash memories), SRAMs, DRAMs, or I/O devices can be directly connected to the on-board SDRAM controller. A standard JTAG interface is provided for debugging and device programming. These chips can also be programmed by using a special Boot mode program built into the device. In boot mode, the on-chip boot ROM downloads a Flash writing application into the internal RAM area of the MCU. This application then handles the serial transfer and writing of internal Flash through the UART interface of the MCU.

The chips require 2.5 V for the core CPU functions and 3.3 V for the I/O and peripherals. The series operate in a wide temperature range of –40°C to +85°C. The ML674001 series can operate at a maximum frequency of 33 MHz, while the ML675001 series operates at a maximum frequency of 60 MHz. The ML675001 series has an 8-kB unified cache memory allowing the chip to operate at the higher clock speed.

The ML67Q4002/3 and ML67Q5002/3 are packaged in a 144-pin LQFP and all the microcontrollers are of a pin-compatible design, allowing for easier upgrade from the ML674001 series to the ML675001 series with a minimum of program and board layout change.

Next month's issue

has a heavy focus on microcontrollers and it is no coincidence that you will be able to read about an extremely powerful ARM microcontroller development system you can build at home, in class or in the lab. As far as we know, this is a first in electronics magazine publishing but then again who else but Elektor?

(040444-1)

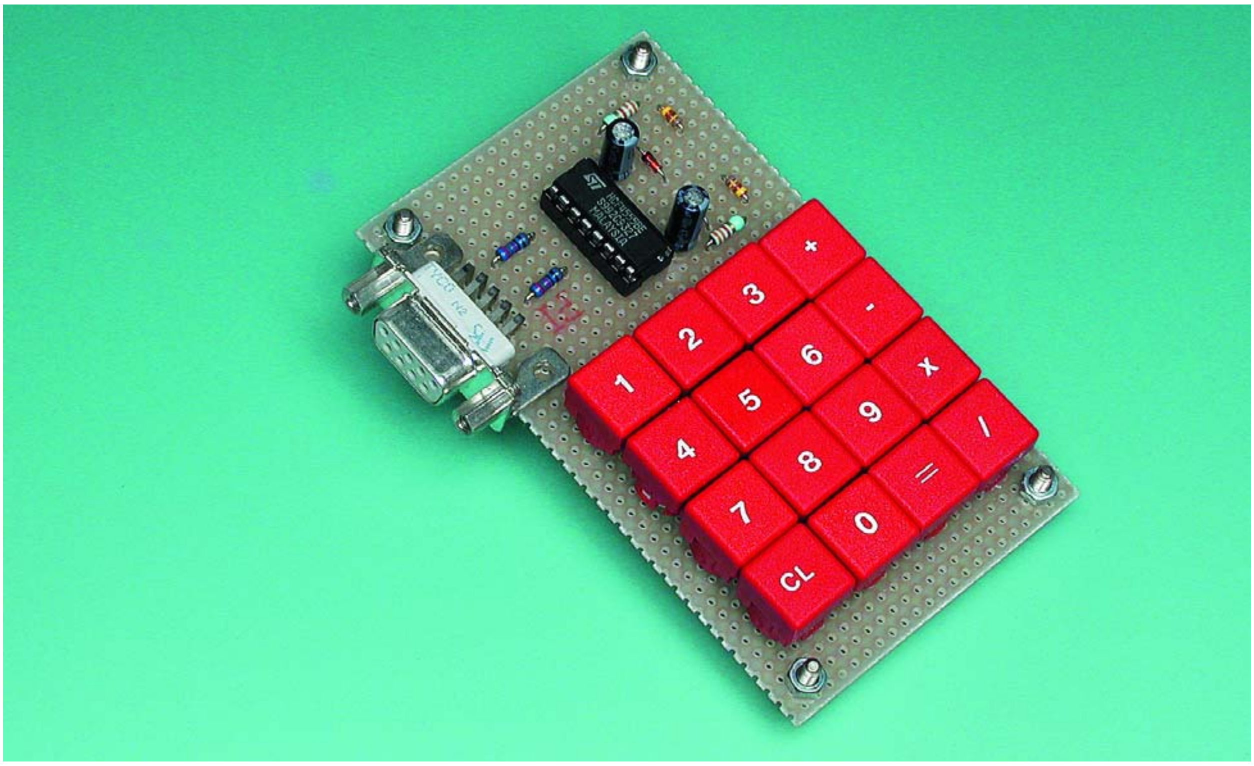
Reference:

1. Intelligent Sensor/Actuator Controller (ISAC), parts 1-4, Elektor Electronics October – December 2001 and January 2002.

COURSE

Delphi for Electronic Engineers

Part 3 – An external calculator



Herman Bulle with thanks to Anton Vogelaar

In the previous article we saw how we could implement a simple calculator using Delphi. This time we connect an external keypad to the serial port, to allow the calculator to be controlled externally. We'll use the program from the last article (slightly modified) to put the numbers onto the display and to perform the calculations.

The external keypad, which we'll build ourselves, consists of 16 keys in a configuration of 4 rows by 4 columns. When a key is pressed, a single row is connected to a single column. The combination of the exact row and column is unique for each key.

By activating each column in turn and then checking which row has a signal, we can determine which key has been pressed.

Before we continue with the practical side we would like to point out that the complete listing for this project would take up too much space in this magazine. The listing (project files) can be downloaded from the Elektor website from the Delphi course for March. We recommend that you download this first and load it into Delphi; this makes it easier to follow the next description.

Connecting the hardware

A CMOS type 4555 data selector is used to activate the columns (see **Figure 1**). This is a dual 1-to-4 decoder/multiplexer. Each decoder in the IC has a 2-bit digital input and 4 outputs. Depending on the binary value of the input (00, 01, 10, 11), one of the outputs of the IC will be driven high.
The RS232 port of the computer has two output signals: **DTR** (Data Terminal Ready, pin 4) and **RTS** (Request To Send, pin 7). As a reminder we have shown the full connection details of a 9-way RS232 connector in *Figure 1*. These two signals are used to drive the line-selector inputs of the 4555. Before we can do this we should provide a supply voltage to the IC. Since a CMOS IC consumes very little power it is quite easy to derive the supply from these two signals. When we're not looking for a key-press there is no need to drive the data selector inputs. We can then keep DTR at '1' and RTS at '0'. It is very easy to derive a symmetrical supply of ±7.5 V (V_{DD} = 15 V), using a handful of resistors, zeners and electrolytic capacitors (R1, R2, D1-D4, C1, C2 in Figure 1). The capacitors are sufficiently large to supply a stable voltage to the IC during a complete scan period. All that's left is to add a little protection to the inputs since the input voltage could be larger than the supply voltage. By adding a 10 kΩ to the signal path the input current is kept to a safe level.

If you're using a laptop you should first measure the RS232 output voltages, as these occasionally deviate from the official RS232 specification (some Dell laptops had outputs of only ±5 V). In this instance the zeners are no longer needed. We have also seen asymmetric outputs (+5 V/0 V), so it is worth checking this out.

Building the circuit of Figure 1 is fairly straightforward. The IC, passive components and 16 keys are easily mounted onto an experimenter's board, and it doesn't take long to do the wiring either. You could use an old RS232 cable for the connection to the PC, with a plug cut off from one end (Make sure that you keep the female connector on the cable, since that is required to plug into the PC). The wires can then be soldered directly to the board. As an alternative to individual keys you could also use a matrix keypad (obtainable from Conrad and many other suppliers). Often these come in sizes of 4x3, in which case two keypads can be 'connected in parallel', using just a single column from the second one.

Software switching

To drive the DTR and RTS lines we'll have to write a bit of Delphi code.
To start with, we have to open the serial port. We have already covered this in Part 1 for the burglar alarm:

```
FHandle:=CreateFile(PChar(Port),Generic_Read+
Generic_Write,0,Nil,Open_Existing,0,0);
If FHandle = Invalid_Handle_Value Then
  Begin
    ShowMessage('Unable to open communication port.');
    Exit
  End;
```

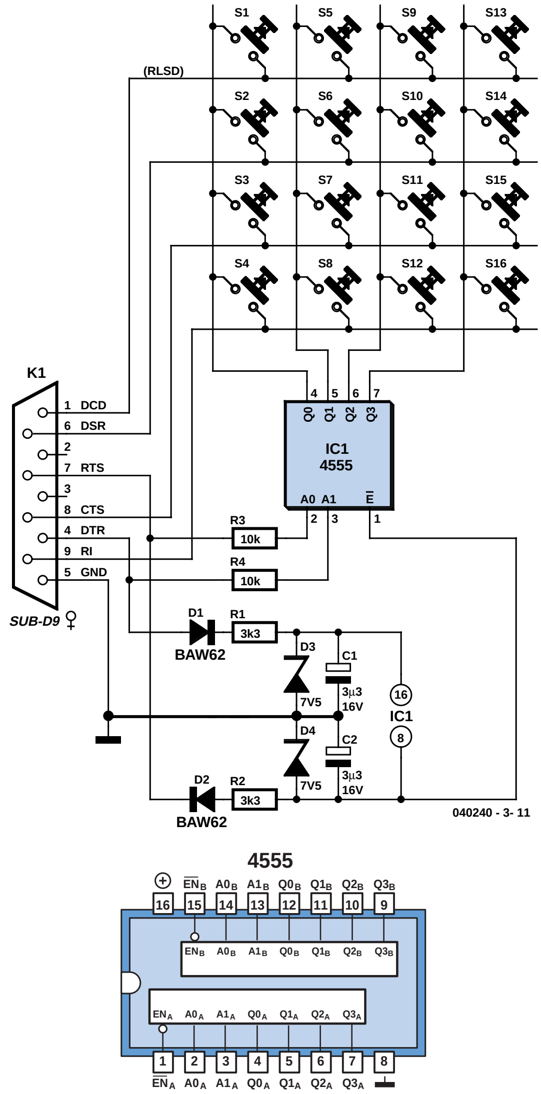


Figure 1.The circuit for the external keypad, which connects to the PC's serial port.

(As a reminder: Delphi programs are not 'case sensitive', unlike C and several other programming languages. Because of this, it may happen that we don't always use capitals and small letters the same way in different parts of the program.)

In the program mentioned above this line had an error-check added: if the port was already open it wouldn't be opened again.

Windows keeps a list of various objects such as programs, ports, drivers etc. These objects are given a unique code (or handle). The function **CreateFile**, called above, returns the value of this code. If this value is equal to **Invalid_Handle_Value**, which is defined in Delphi, something went wrong during the initialisation of the port and the program is halted with a message and the **exit** command.

On successful completion of this phase we have to activate a few outputs. DTR has to be set to +12 V and RTS to –12 V.

To help with programming the UART, a data structure (**DCB**, see inset) is defined in Windows that contains a large set of parameters. Once all parameters have been set, this data structure is sent to the UART. For our program the **DCB.Flags** field is the most important. In this field virtually all bits have a separate function. We have defined two constants in uDriver, which are required to turn DTR and RTS 'on' and 'off'.

First the current state of the UART is read with the command **GetCommState(FHandle, DCB)**.
Next we set a bit in the FLAGS field to turn RTS on:

```
DCB.Flags := DCB.Flags OR RtsEnable
```

The RTS output then becomes –12 V.
We do something similar for DTR, but in this case we have to set the bit to '0' while keeping all the other bits the same:

```
DCB.Flags := DCB.Flags AND (NOT DtrEnable)
```

We can now send the modified DCB back to the UART using the command **SetCommState(FHandle, DCB).** Since we need to use all possible combinations of DCD and RTS when selecting the columns, we have written the following procedure:

```
Procedure Key_Out(RTS, DTR: Boolean);
Var DCB: TDCB;
Begin
  IF FHandle > 0 Then
  Begin
   GetCommState(FHandle, DCB);
   If DTR Then DCB.Flags := DCB.Flags Or DtrEnable
   Else DCB.Flags := DCB.Flags And (NOT DtrEnable);

   If RTS Then DCB.Flags := DCB.Flags Or RtsEnable
   Else DCB.Flags := DCB.Flags And (NOT RtsEnable);

    SetCommState(FHandle, DCB)
  End
End;
```

To turn on the supply voltage to the IC we call this function with the following parameters: **Key_Out(False, True)**. This is done every time after scanning the keys, thereby keeping the capacitors fully charged.

Scanning the matrix

The next building block required for this project has to take care of the reading of the status of the four input lines. With the help of the procedure **GetComm-ModemStatus(FHandle, MdmSts)** these details are stored in the variable MdmSts. By applying a mask we can determine which bit, if any, is set to '1'. A mask is a number which has just a single bit set to '1', for example binary 1000 (= hex 10), binary 10000 (= hex 0020) etc. Delphi has already defined some important values for us:

```
MS_CTS_ON = hex 0010
MS_DSR_ON = hex 0020
MS_RING_ON = hex 0040
MS_RLSD_ON = hex 0080;
```

If we now apply the mask using the Boolean AND function we can determine if this bit is set. This therefore tells us if a key in this row is pressed:

```
If MdmSts And MS_RLSD_ON = MS_RLSD_ON Then
    Result := 1
```

The same method is used to check if a key is pressed in the other rows.
The result will be 1,2,3 or 4, depending on the selected row.
These lines of code have been combined in the function **Key_Inp**, which returns the value of the selected row.

Combining the routines

In principle that is all there is to do: we know how to select a column and how to read on which row a key is pressed. However, these functions still have to be 'glued together' and the result has to be shown on the display. This takes place as follows: a timer is used to call the function **GetKey** at regular intervals. This calls the procedure **Key_Out** to select a column, for example Key_out(False, False). Next we use **Key_Inp** to check if a key in that column has been pressed. A small section of this procedure is shown below: if no key is pressed in the first column, the variable Row becomes 0. If a key is pressed, Key_inp returns the row number.

```
If row=0 then
  Begin
    col:=2;
    Key_Out(False, True);
    row:=Key_Inp;
  End;
If row=0 then
    col:=3
```

In this way the entire keypad matrix is scanned. Once the row and column are known, the correct key number is looked up using the conversion matrix **Map [row,col]**, as follows. We do have to make sure that we don't read 2 key presses if a key is held down a bit longer. This is where the variable **KeyLast** comes to the rescue. If we insist on at least one scan without a key press between two successive key presses, we can use that to distinguish two real key presses from a single one, which is held down too long.
The previous key press should therefore always be an empty scan (with key number 0) for the current scan to be valid.
Every old key press is stored in the variable **KeyLast**. When we're scanning for a new key press the result will only be valid if the old scan (stored in **KeyLast**) was 0. Adding the following line to the end of the **GetKey** procedure stops the generation of false key presses:

```
If (KeyLast = 0) Then Result := Key
else result:=0;
KeyLast := Key;
```

We then store the value of the scanned key in **KeyLast.** This nearly completes this section. We just have to make sure that the supply voltages are restored at the end of a scan. This is done by calling Key_Out(False, True) once more, which sets DTR to +12 V and RTS to –12 V.
Apart from the procedures mentioned above for driving and reading the RS232 port, there is another procedure (KeyGet), which is used during the simulation of the program. The intention is that only one of the two functions is active. The other is removed using curly brackets (in Delphi any text between curly brackets is interpreted as a comment).
In the simulation the tag of the key is read, which we

Device Control Block

The Device Control Block contains a list of parameters that are required to configure a serial port. The elements BaudRate, StopBits etc. are easily recognised. It is defined as a record of type TDCB and has the following structure:

	Type	Size
dcblength	dword	4
baudrate	dword	4
flags	longint	4
wreserved	word	2
xonlim	word	2
xofflim	word	2
bytesize	byte	1
parity	byte	1
stopbits	byte	1
xpmcjar	char	1
xoffchar	char	1
errorchar	char	1
eofchar	char	1
evtchar	char	1
wreserved11	word	2

The flags are of particular interest here. Each bit in the 16-bit word has a unique function, which is shown in the table below:

Binary	$00000001
ParityCheck	$00000002
OutxCtsFlow	$00000004
OutxDsrFlow	$00000008
DtrControlMask	$00000030
DtrControlDisable	$00000000
DtrControlEnable	$00000010
DtrControlHandshake	$00000020
DsrSensitvity	$00000040
TXContinueOnXoff	$00000080
OutX	$00000100
InX	$00000200
ErrorChar	$00000400
NullStrip	$00000800
RtsControlMask	$00003000
RtsControlDisable	$00000000
RtsControlEnable	$00001000
RtsControlHandshake	$00002000
RtsControlToggle	$00003000
AbortOnError	$00004000
Reserveds	$FFFF8000

When the symbol '$' is in front of a number it means it is a hexadecimal number.

In this case we only make use of the numbers $00000010 (DtrControlEnable) and $00001000 (RtsControlEnable). The others don't really concern us here.

More information can be found in the Windows SDK section accessed from the Delphi Help menu.

described in the previous article. This value is put into the variable KeyPressed. In the simulation mode this value is passed as the result of the function GetKey and subsequently processed. From this point onwards the program is the same whether in simulation mode or when using an external keypad.

We now know how to read the keys. From the main program (in the unit uConsole) we start a timer that calls a procedure every 50 ms to read the key matrix. The procedure ControlExe (in uControl) separates the received IDs (1 to 16) into two groups. When ID ≤ 10 a digit was pressed and a number is being entered. When ID > 10 an action is requested. In that case we subtract 10 from the ID and execute the appropriate action from the list defined in Tmath. As an example: 'subtraction' has an ID or tag of 12. Process 2 is mtSub (counting from 0) and this generates a subtraction. When the process has finished, a Screenrefresh is executed, which shows the result in the display of the calculator.

Functional design

Observant readers will have noticed that this program consists of three functional blocks, which are the three units. The basis for this division is that complicated problems are best described as a block diagram, which is also common practice in electronics. If you have a clear division between these blocks you can describe and test the working of the individual blocks better, and the chance of introducing design faults becomes smaller. In this project we have split the software into three layers (in the software world we talk about layers in this context).

We have a presentation layer (uConsole), which receives the results from the simulated keys, a control layer (uControl), which runs the processes in the procedure ControlExe and a driver layer (uDriver), which takes care of the communications with the outside world via the RS232 port.

In this instalment we have shown you how to implement 4 command lines and 4 status lines under Windows for use external to the computer, with only minimal hardware and a serial port. Since most desktops have two serial ports as standard, a total of 8 command lines and 8 status lines are available for external use.

(040240-3)

Ordering Delphi 7

Borland has made the Personal version of Delphi 7 available cheaply especially for this course. The CD costs € 10.00 (ten euros) and contains Delphi 7 as well as several extra files for this course. It can be paid for by credit card (see website below) or bank transfer (in the EC) by transferring to (please copy exactly):

Bank:	ABNAMRO
IBAN:	NL31 ABNA 0577002562
BIC:	*ABNANL2A*
Name:	DETLEF D. OVERBEEK
Address	EDELSTENENBAAN 21 IJSSELSTEIN
Post code:	3402 XA
Country:	THE NETHERLANDS
Reference:	DELPHI ELEKTOR

IBAN/BIC payments should not incur bank costs when processed correctly — ask you bank for details. Cheques are not acceptable. The HCC PGG has set up a special website in support of this course: www.learningdelphi.info/

Here you can find the most up-to-date news and extra files for the course, as well as credit card payment options.

PLAY SOLITAIRE

...packed in an AT90S micro

Andy & Rose Morrell

This version of Solitaire is played with an array of LEDs instead of pegs, with a microcontroller behind the scenes to check if you're any good at playing. As you'll soon find, the game is challenging and addictive!

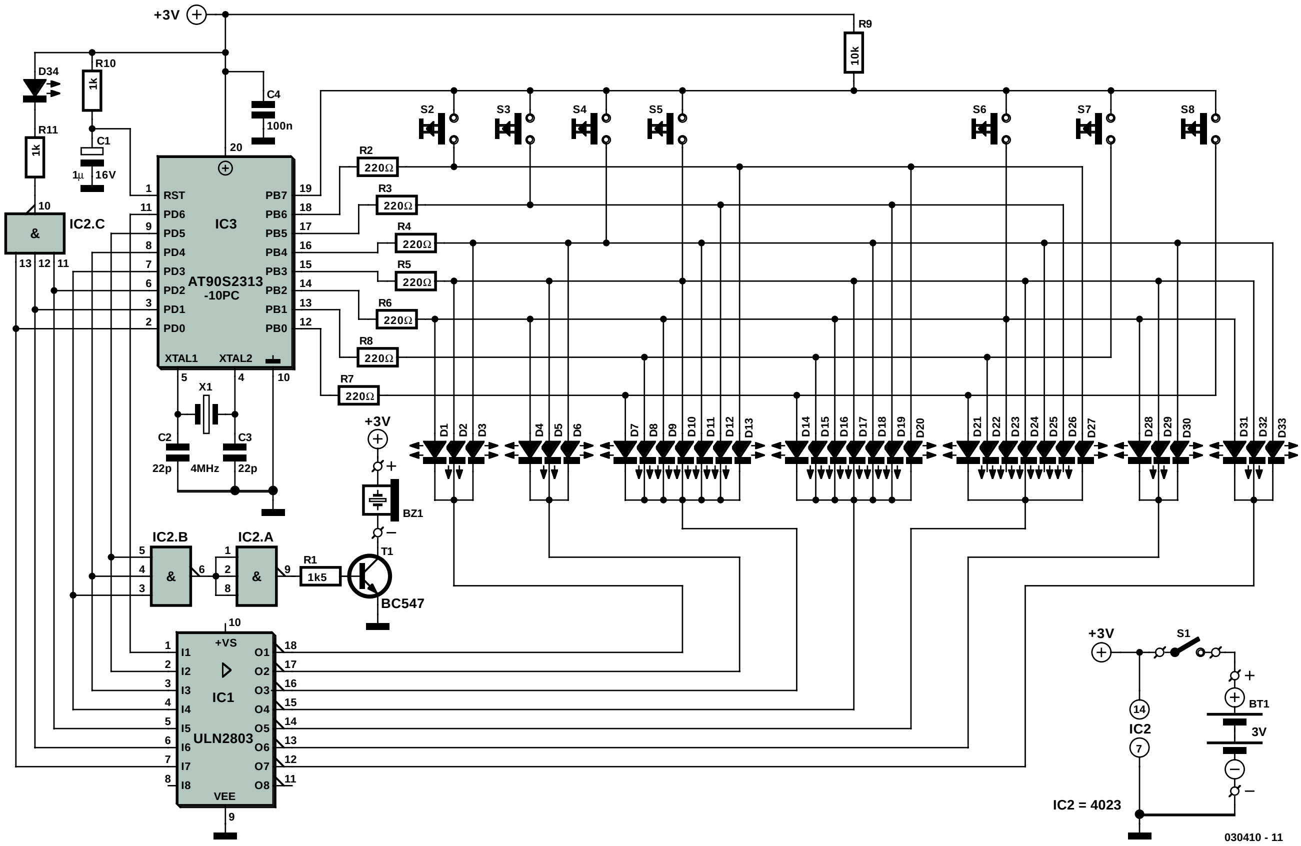


Figure 1. At the heart of the Solitaire game you'll find an AT90S2313 microcontroller running some really clever software.

This project is a fine instance of a microcontroller and its resident software slashing component count when implementing a logic circuit, as well as providing yet an another answer to the perennial "great-but-what-do-you-do-with-it?" question from outsiders and newbies when discussing those multilegged beasts commonly known as microcontrollers.

The design is based on the Atmel AT90S2313 microcontroller which was selected for its low cost, speed, number of I/O lines and ease of programming in assembly code using free tools.

The hardware

If you think of a microcontroller acting like a black box and sitting between input and output devices like switches and LEDs, then the circuit diagram in **Figure 1** is nothing special. It does, however, contain a few ingenious solutions when it comes to marrying software to hardware that's to remain as simple and reproducible as possible.
The AT90S2313 micro, IC1, runs at a clock frequency of 4 MHz obtained with the aid of quartz crystal X1. The two 22-pF capacitors serve as parallel load devices for the oscillator/crystal configuration (note that the oscillator is on board the AT micro).
Output devices controlled by the AT90 micro comprise active buzzer Bz1, LEDs D1-D33 mimicking the playing field and LED D34 to confirm that the move has been performed. The buzzer sounds in response to the microcontroller software activating the PD3, PD4 and PD5 lines at the same time. This condition is signalled by NAND gate IC2.B, with IC2.A acting as an inverter only. A similar arrangement but without logic inversion is used for LED D34, which lights when PD0, PD1 and PD2 are logic High at the same time. Here, IC2.C is the responsible NAND gate.
All Port lines PD0 through PD6 are applied to the inputs of an ULN2803 driver IC. Note the inverting action of this IC — it will pull the commoned cathodes of the LED sets low in response to a high level on the port lines. The anodes of the LEDs are connected to port lines PB0 through PB7 via 220-ohm current limiting resistors. In this way, each individual LED in the playing area can be turned on an off using just 15 control lines instead of… right, 33! The process is called multiplexing. Here, it relies on software.
As to input devices, a number of PB on IC1 lines are also connected to switches, requiring them to act as inputs as well as outputs — in a controlled manner, of course! Pushbutton activity is detected on port line PB7 which is exclusively an input.
Switches S1-S4 are the cursor (direction) controls, while S6, S7 and S8 act as the SELECT, CANCEL and STALEMATE controls.
The microcontroller is reset at power-on by a brief logic Low level obtained from network R10-C3.
The circuit does not have a voltage regulator and is powered directly from a 3-volts supply made from two series connected 1.5-V dry batteries.

Rules for playing Solitaire

As the name suggests, Solitaire is a game to be played on your own. The board is made up of 33 LEDs laid out in a cross shape, where a piece on the board (or 'peg') is indicated by an active LED.

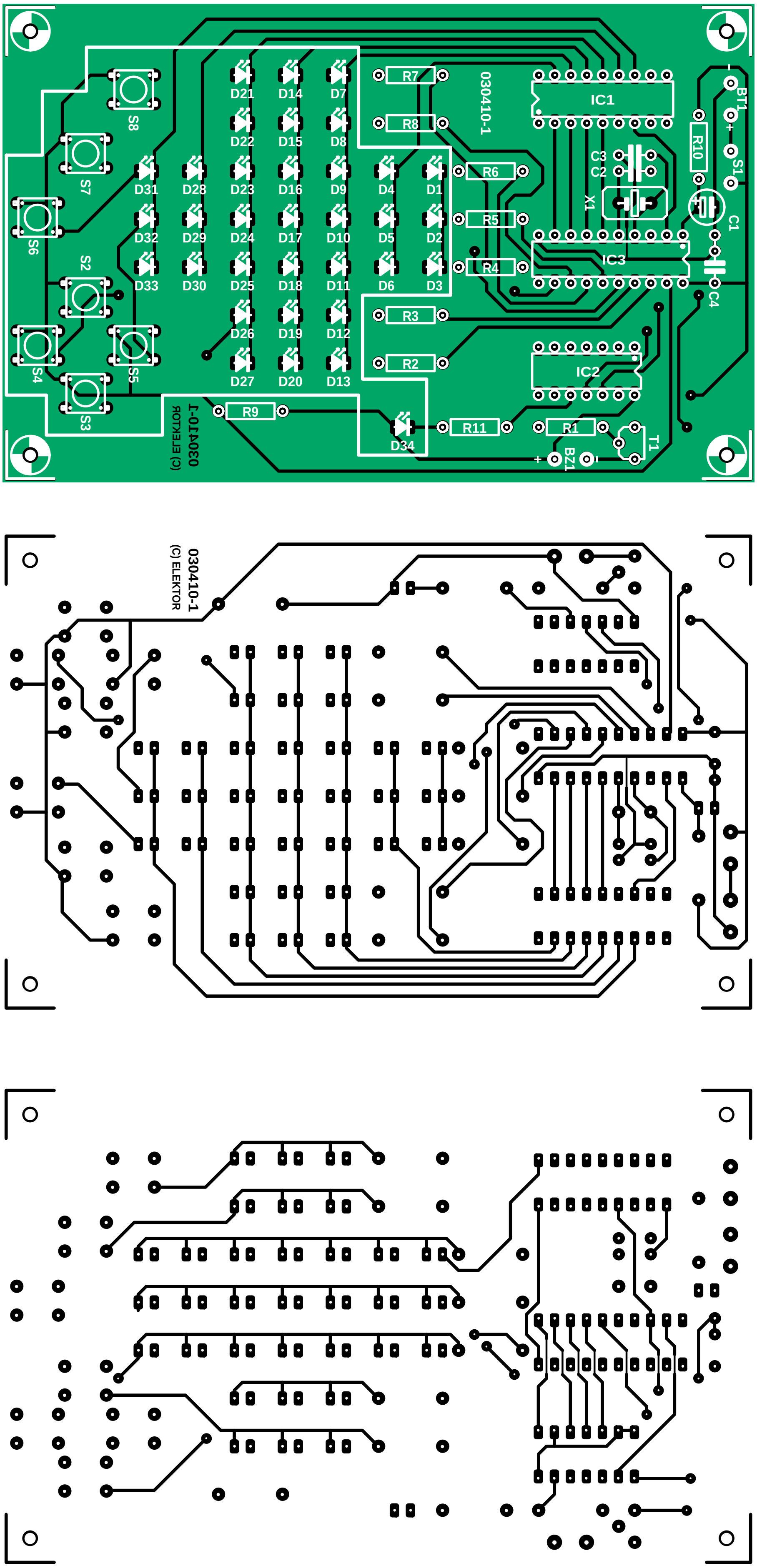


Figure 2. PCB designed for the Solitaire game.

COMPONENTS LIST

Resistors:
R1 = 1kΩ5
R2-R8 = 220Ω
R9 = 10kΩ
R10,R11 = 1kΩ

Capacitors:
C1 = 1µF 16V, radial
C2,C3 = 22pF
C4 = 100nF

Semiconductors:
D1-D34 = LED, 3mm, low current, red
T1 = BC547
IC1 = ULN2803
IC2 = 4023
IC3 = AT90S2313-10PC, programmed, order code **030410-41**

Miscellaneous:
S1 = on/off switch, one contact
S2-S8 = pushbutton, PCB mount, 1 contact, 5mm, e.g., T60 (Conrad Electronics # 700479)
X1 = 4MHz quartz crystal
BZ1 = 5VDC (active) buzzer
BT1 = 2 x penlight (AA) battery with holder
Enclosure, e.g., Hammond type 1591DTRD (150 x 46 x 84mm)
PCB, ref. 030410-1, from The PCBShop

The aim of this game is to be left with as few pieces as possible at the end of the game. You can claim victory if you have only one piece left in the centre of the 'board'. As you will discover, that's very hard to do! So how do you arrive at this desired state?

When you turn the game on, there will be 32 LEDs lit up continuously and one, at the centre, flashing. The flashing LED shows your position on the board. The first step is to decide either to play your turn

Project Downloads

AT90S microcontroller software.
File number: **030410-11.zip**

www.elektor-electronics.co.uk/dl/dl.htm, select month of publication.

where your piece is or to move the position of your piece. So how do you move your position should you wish to?

1. To move your position simply use the four direction switches S1-S4. You can move your position to wherever you like.

2. When you are positioned to where you want to be, press the SELECT button. The LED that you are positioned on will flash even faster. This means that you are in jump mode. To jump a piece, which must be next to yours — up, down, left or right, but not diagonal — press the appropriate direction switch. The piece you are positioned on will jump over that piece onto the space on the other side, and the jumped piece will disappear. You can only jump a piece if there is a space on the other side of that piece to be jumped, and if there is a piece where you are positioned. It is not always obvious if there is a piece where you are positioned, because the flashing LED looks the same whether it is just your position or whether you actually have a piece there. If the jump is requested and it is impossible, you will hear a beep. The jump mode will be cancelled for that move and you can then continue as before.

3. You continue this procedure until you can jump no more. The smaller the number of pieces left, the better. If (you think) a 'move' is no longer possible, then the player can press the STALEMATE button. Depending on the score, the buzzer will produce a number of beeps indicating your result:

   1 beep = ≤ 10 remaining pieces
   2 beeps = 8 remaining pieces
   3 beeps = 4 remaining pieces
   4 beeps = 1 remaining piece!

   Where 'piece' should be taken to mean 'peg' or, in electronic parlance, 'LED'! In the assembly code listing, you'll find labels like 'bad_luck', 'excellent', 'well_done', 'very_good' and 'good' to describe relevant routines.

Listing 1. LED control routine
(AT90S program extract)

```
INIT_RED:

         LDI      TEMP, 0B11111111 ;7 BIT TABLE FOR 7X7 LED PATTERN
         STS      $60, TEMP       ;
         LDI      TEMP, 0B11111111
         STS      $61, TEMP       ;
         LDI      TEMP, 0B11111111
         STS      $62, TEMP       ;
         LDI      TEMP, 0B11110111
         STS      $63, TEMP       ;
         LDI      TEMP, 0B11111111
         STS      $64, TEMP       ;
         LDI      TEMP, 0B11111111
         STS      $65, TEMP       ;
         LDI      TEMP, 0B11111111
         STS      $66, TEMP       ;
         ;
         LDI      TEMP, $80       ;TERMINATOR
         STS      $67, TEMP       ;
         ;
         LDI      TEMP, $80       ;TEST
         STS      $68, TEMP       ;
         RET
;=========================================================================
REFRESH_RED:
         CLI
         PUSH     R27
         PUSH     R26
         ;
         LDI      R26, $60
         LDI      R27, $00
         LDI      COUNT1, 01
         RCALL    CONVERSION_ROUTINE
         ; INPUT IN COUNT3
         ; RESULT IN COUNT2
         ;
REFRESH_LOOP:
         CLR      TEMP              ;
         LD       TEMP, X+          ;GET RAM DATA AND INCREMENT POINTER
         CPI      R26, $68          ;END OF COLUMN?
         BREQ     EXIT_ROUTINE     ;YES
         CPI      FLAG,1            ;NO
         BREQ     FLASHA            ;
         ;
         CP       COUNT1,COUNT2    ; COUNT1 = COUNT3 EFFECTIVELY
         BRNE     WRONG_ROW         ;
         EOR      TEMP,CURSOR       ;

WRONG_ROW:
FLASHA:
         OUT      PORTD, COUNT1    ;
         OUT      PORTB, TEMP       ;
         RCALL    DELAY             ;
         ; NOW SHIFT COUNT 'LEFT'
         ROL      COUNT1            ;
         RJMP     REFRESH_LOOP     ;

EXIT_ROUTINE:
         POP      R26               ;
         POP      R27               ;
         RET                        ;
```

Construction

The game is built on a double-sided printed circuit board of which the copper track layout and component mounting plan are shown in **Figure 2**. **The parts in the enclosed area (LEDs and pushbuttons) should be mounted at the solder side of the board**. You may want to do the same with the on/off switch, S1. This is necessary to allow the board to be mounted as close as possible to the inside of the enclosure top panel (with appropriate cutouts for the switches). If you want to use solder pins for the battery connections, these are also best fitted at the solder side.

The photograph in **Figure 3** shows the board before it was fitted behind the opaque panel. The red filter enhances the visibility of the LEDs. Actually, that's achieved by making the board and the other component less visible!

Software

The complete source code listing for the program executed by the AT90S micro is available free of charge from the Publishers' website at www.elektor-electronics.co.uk as file number **030410-11.zip**.

The program listing is a good example of simple code with comment where necessary. An extract of the program appears in **Listing 1** — this part looks after the LED activity on the 'board'. Finally, those of you unwilling or unable to program your own micro for the project will like to know that the AT90S2313 can also be bought ready-programmed from us under order code **030410-41**.

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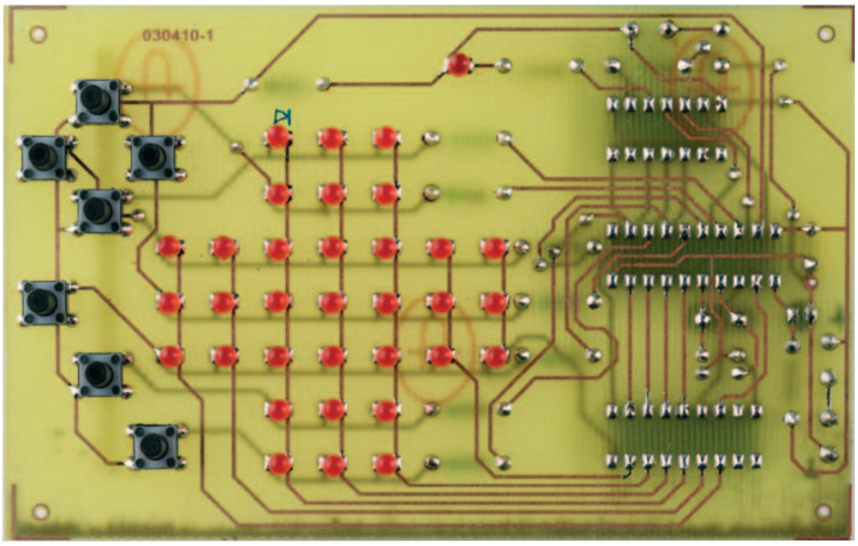
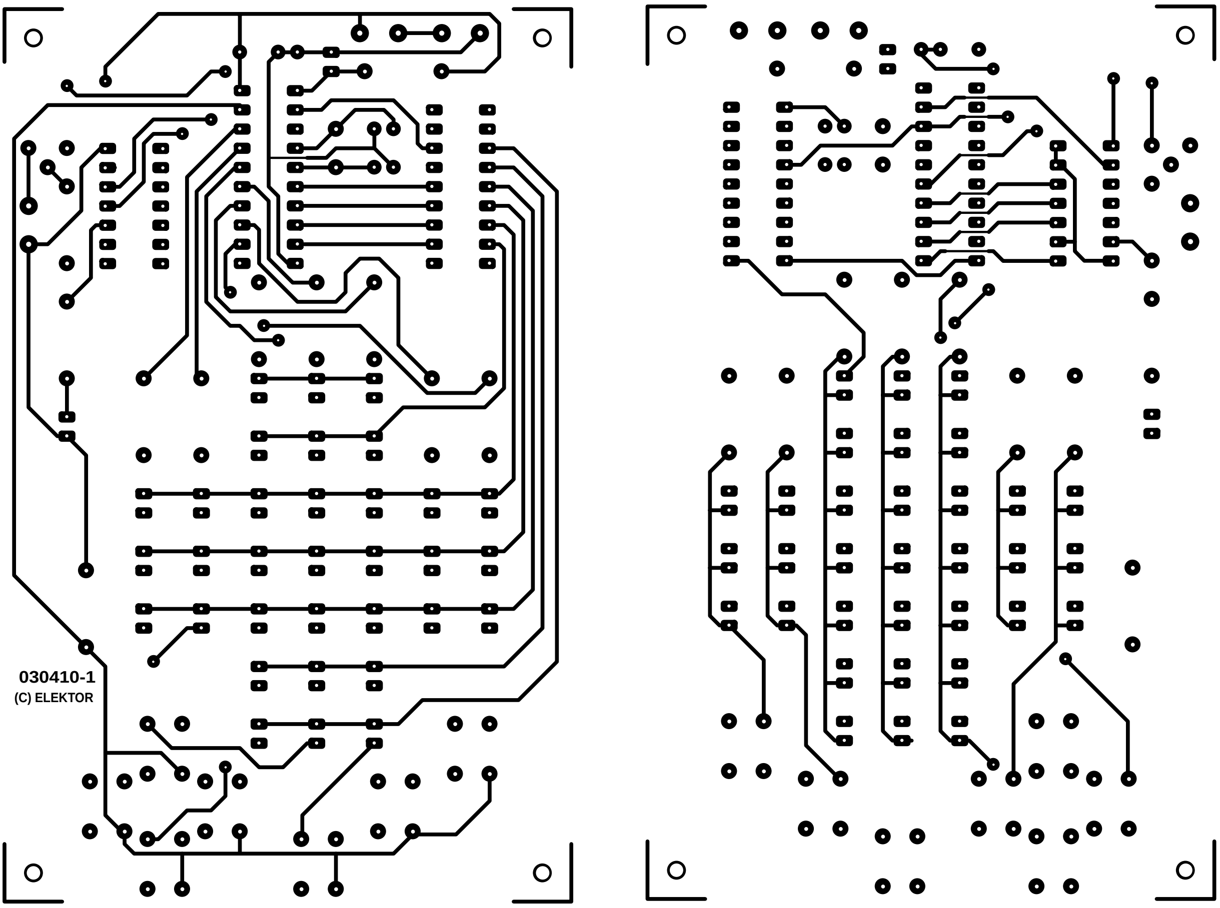


Figure 3. Completed board before mounting in the enclosure.



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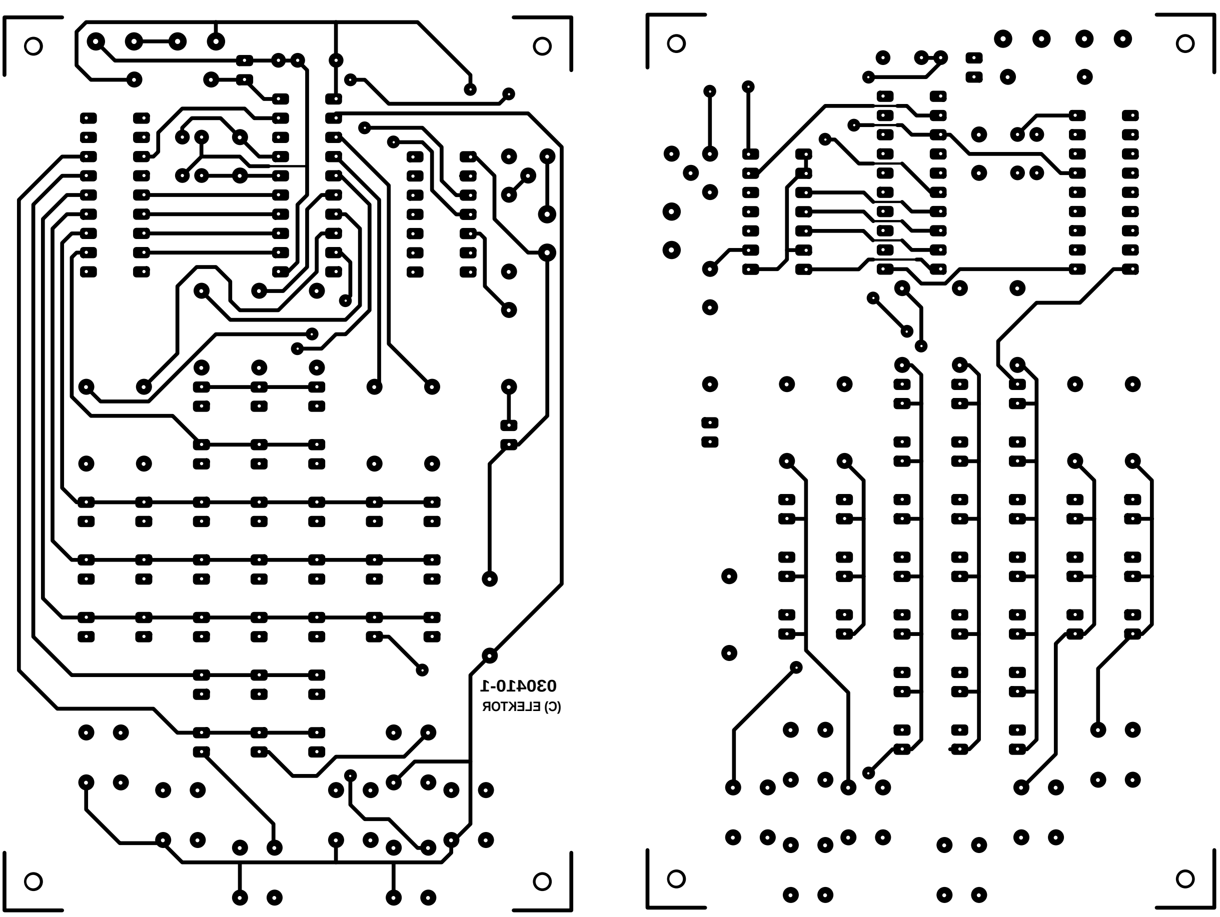

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Altium FPGA LiveDesign Kit

Designing with FPGAs

Paul Goossens

Modern circuits make ever more extensive use of FPGAs. Special software is required when designing with these devices. One of the companies that produces such software is Altium. In order to allow their software to be evaluated, they offer a cheap evaluation kit. Once the evaluation period has expired, the kit is still very useful as a development platform for your own designs!

Altium's background is that of a developer of various software products for electronics designers. Their product range contains, among other things, compilers for microcontrollers, schematic and PCB drawing software and FPGA design software. To allow their software to be evaluated, Altium has offered for some time now a development platform called nanoBoard together with an evaluation version of their software. Recently, they started offering a slimmed-down version of this board plus the above-mentioned software, which goes by the name of LiveDesign. This evaluation board costs 99 Euro (approx. £68) (excl. P&P). You get quite a bit for this money.

Contents

The kit contains, in addition to the development board itself, a mains power adapter and a programming cable. There are also 2 ribbon cables included that allow the I/O connectors of the board to be connected to your own expansion PCB. From the software perspective there are four CDs. These contain the evaluation software, examples and two presentations for Altium products. Finally, the box contains various items of documentation.

It is very important NOT to discard the docket on the packaging carton. This contains the customer number and release-code that you will need when installing the software.

This installation itself is very straightforward, but it is still useful to read through the installation instructions first. Especially since in addition to the Altium software, you will also need to install software from the manufacturer of the FPGA (depending on your choice this is either Altera or Xilinx).

Hardware

The development board is provided with a number of standard I/O features, such as we are accustomed to seeing on these kinds of boards (also refer to side bar). However one thing is striking, it is possible to adjust the corner frequency of the output filter of the DAC with a resistor array.

The heart of the circuit is the FPGA, of course. Depending on your choice, this is either a Cyclone-FPGA (EP1C12F324C8) from Altera or a Spartan-3 (XC3S400-5FG456C) from Xilinx. As you will see shortly, these two FPGAs are very powerful!

Otherwise there is not much more to say about the hardware, except that with our board the wires between the speakers and the PCB had become disconnected. Fortunately, this was easily fixed with a little bit of soldering!

Software

The accompanying software is an evaluation version of the Protel/Nexar software suite from Altium. This version is time limited (30 days). Keep in mind that these 30 days start counting down from when the package is sent! It makes sense therefore, to install and try everything out immediately once you receive the package.

This software looks very polished and it is definitely worth the effort to try all its features. There are too many features for all of them to be described here. It is certainly no exaggeration to state that this software contains everything that is necessary to complete your own FPGA design from beginning to end. Even the firmware for so-called 'soft-core' processors can be developed with this software! While trying the various examples you will quickly become familiar with most of the functionality of this software.

In addition to the enclosed software, you will need to

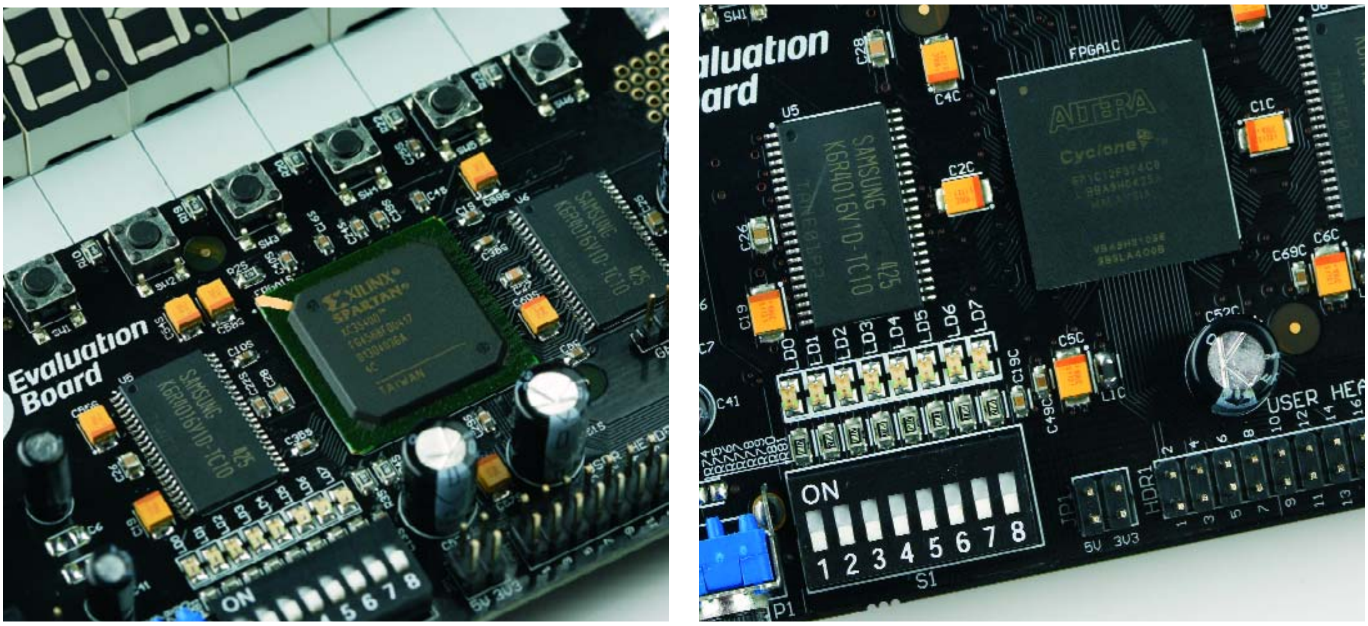


Figure 1. The board is available with two different types of FPGA.

install software from the FPGA manufacturer. If your board is fitted with a Xilinx FPGA, then you will need the free Web-ICE-Pack software. In the case of the Altera FPGA you will need the Quartus software (also a free download). With our version of the software, the latest version of Quartus was not yet supported. On Altium's website you can find how to solve this problem very easily. It is only a case of tweaking a small text file. Very simple, but you just need to know how.

Soft-core

This development kit is naturally accompanied by a number of examples. These can be roughly divided into 2 categories, namely pure hardware and soft-core-processor based designs. Before we go any further we will explain the term soft-core-processor.

A soft-core-processor is a microprocessor that is programmed inside an FPGA. These processors are usually designed in a higher description language, such as VHDL or Verilog. An advantage is that any signal from inside the processor can be examined and routed to the outside world. It is also not inconceivable to add some 'custom' instructions to the instruction set of the processor, just for this particular application!

There are three different soft-core-processors among the examples. The TSK51 is an 8051-compatible processor, the TSK80 is compatible with the Z80 and finally there is the TSK165, which is compatible with the PIC16-series from Microchip.

In addition to the design of the processor, there is also the necessary firmware to set this processor to work. The software suite therefore also contains a C-compiler, which supports these processors. Designing hardware, software and now even your own processor can be done in just one software suite!

Examples

The supplied examples range from simple to extremely complex. So there is something to appeal to anyone. There are examples that produce some simple effects using the LEDs, but there is also an example of a

Hardware Specifications

- FPGA choose between: Cyclone (Altera) **or** Spartan-3 (Xilinx)
- 2 x 512 Kbyte SRAM
- 6 x 7-segment display
- 6 pushbuttons
- 8-way DIP switch
- 8 LEDs
- TEST/RESET pushbutton
- system clock 50 MHz
- JTAG-interface
- 2 x I/O-connector with 20 unused I/O-signals
- RS232 connector
- 2 x PS/2 connection
- VGA output with R/2R-DACs
- stereo delta/sigma DACs
- line-out output
- headphone output (100 mW)

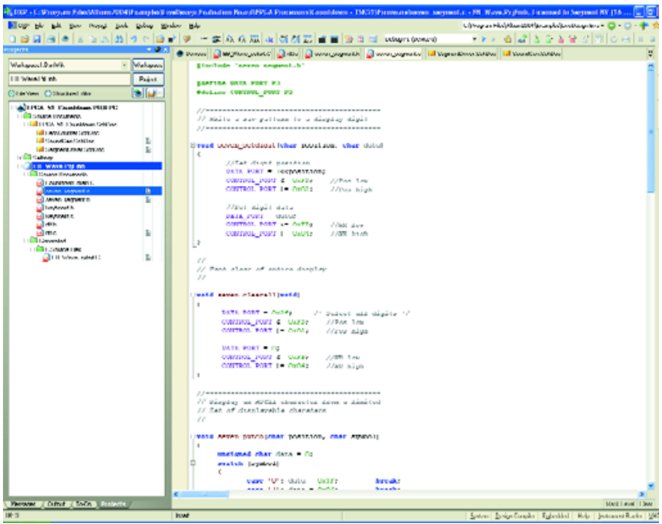


Figure 2. Working with the source code for a soft-core-processor.

Figure 3. With these virtual instruments you can make measurements inside the FPGA.

complete 'Pacman' video game with PS/2-inputs for mouse and keyboard and a video output for the screen! There are also some intermediate complexity examples such as a clock with alarm function.

To get an impression of the scale of the designs in the FPGA, we compiled an included example that contains no less than 4 processors (TSK165). Then we looked at a report from the compiler software and it appeared that this design used only about 1/3 of the available resources of the FPGA!

All examples can be programmed in the FPGA after compilation and be tested directly on the board.

Debugging

The real power of the software is really only utilised when you want to carry out measurements on your design. It is obviously impossible to measure internal signals of the FPGA with a probe, so most examples are provided with a JTAG interface.

With this you can (without disturbing the operation) make real-time measurements inside the FPGA. For this purpose, several measuring instruments are built into the software. A logic analyser, an oscilloscope (digital, obviously) and some I/O-blocks that let you determine logic levels with some buttons.

To make this all complete, there is the possibility to debug the processors with their firmware. Registers can be changed, breakpoints can be placed, memory can be examined, etc. All this can be done!

Finally

This FPGA development kit offers much for the money. Unfortunately the software is no longer usable after 30 days. However, you can continue to use the board in combination with the free software from the FPGA chip manufacturers. Although, it will take a little getting used to after having been spoilt with the software suite from Altium.

The diagram for the design-kit is supplied as a Protel schematic file. This means that after the end of the evaluation period the schematic cannot be viewed again. If you would like to continue to use the board after the software has expired it makes sense to print the schematics early on. Without these schematics it is not easy to figure out which pins from the FPGA are connected to what sections of hardware and connectors!

Purchasing this software is not really in the league of the average hobbyist (both parts of the software together cost around 10,000 Euro!). The free software from the FPGA manufacturers, however, is still very usable to realise your own designs with this kit, after the 30-day period.

Because an FPGA with sufficient capacity has been used, it is not very likely that your design won't fit in this device. Furthermore, the FPGAs are so quick that it has become relatively easy to design complex and fast equipment at home. The nicest aspect of this is that when changing the design there is no need to immediately reach for the soldering iron. A simple button push is sufficient, the rest is carried out by the PC and hardware.

This development kit offers more than enough possibilities to begin exploring and experimenting with FPGAs. You will quickly be surprised as to what you can do with it!

(040414-1)

Internet links

www.altium.com

www.altera.com

www.xilinx.com

What are FPGAs?

FPGA stands for Field Programmable Gate Array. These are chips that are provided with a large number of small digital building blocks, each of which can perform a simple function. The inputs and outputs from these building blocks can be connected to inputs and outputs from other blocks via a matrix of signal lines, hence the term Gate Array.

When the power is first turned on, the inputs and outputs are disconnected from the signal matrix. The chip can be programmed via a programming port. This means that the designer, with the aid of a programming file, determines which inputs and outputs are connected to each other, so that the chip will perform a certain function.

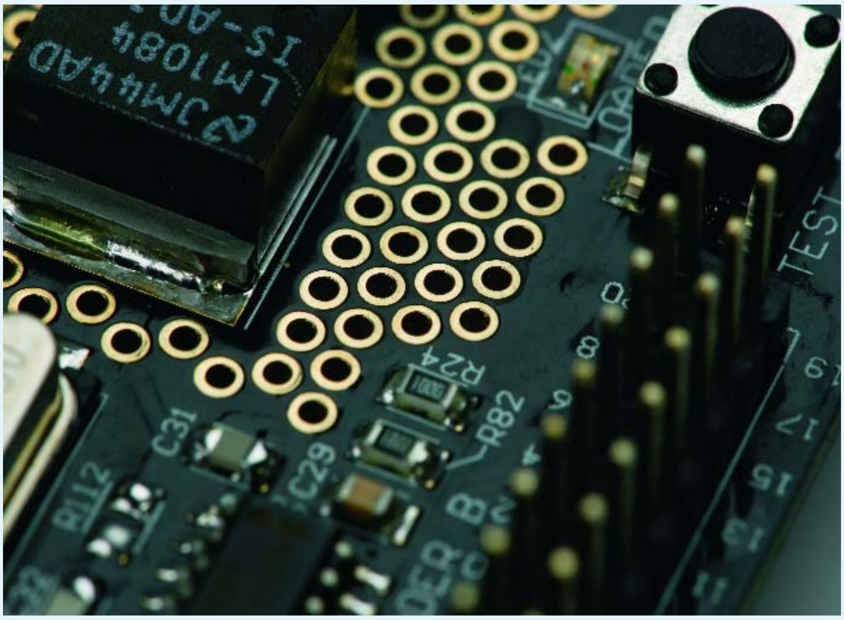
It is easy to imagine the FPGA as a (very) large experimenting board with tens of thousands (or even hundreds of thousands) of logic ICs, which are not interconnected. Connections are made between these chips according to the wishes of the designer.

An advantage of an FPGA is that all these building block are on one piece of silicon and that each building block has a very high propagation speed (in the order of a few nanoseconds). This means that very fast digital circuits can be realised with an FPDA, which cannot be done with a handful of logic ICs.

FPGAs are often also provided with a number of additional features, such as block of memory, hardware multipliers, etc. Some FPGAs also contain a built-in processor that can be connected with the design in the FPGA.

Depending on the capabilities of the FPGA, it can sometimes be difficult (as we described in this article) to implement and apply your own processor in an FPGA!

Personal ideas



In the process of authoring this article I've gradually become very enthusiastic about the possibilities for this board. One of my hobbies (in addition to electronics, of course) is building and flying model aeroplanes.

As it happens, I have a nice aeroplane that has a reasonably spacious fuselage, which accommodates this board easily. Originally I had the idea of fitting a camera with accompanying transmitter in there, so that I could see from the ground on a TV the world from the cockpit of my model aeroplane. Afterwards I wanted to add all sorts of bells and whistles.

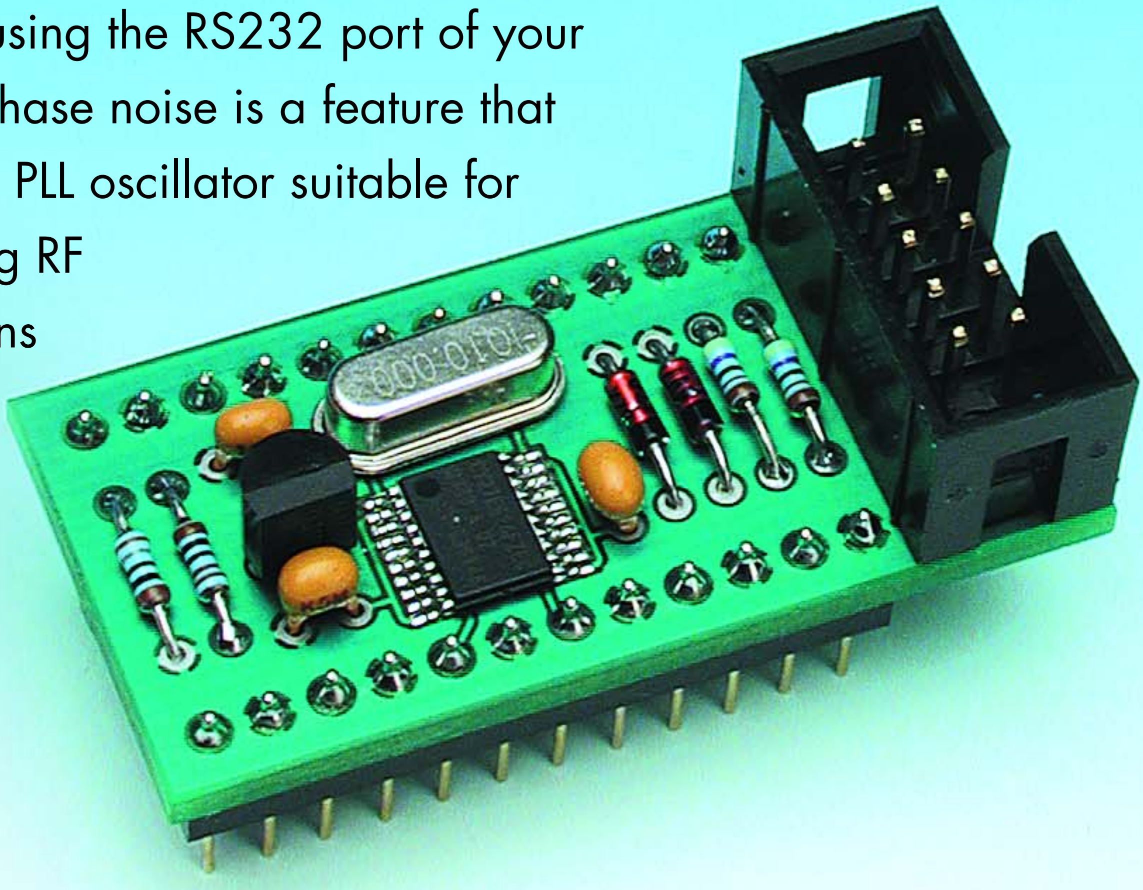
While experimenting with this board from Altium, I noticed how powerful the FPGA is that has been used here. That's why I immediately took up the plan to go a little further, namely converting the video signal from the camera to a digital signal with a video decoder and send it to the FPGA. In addition I would like to place a number of sensors for measuring the air pressure (i.e. height and vertical speed) and motor temperature, and possibly a GPS module to determine the position. This information is also (in digital form) sent to the FPGA. The FPGA will then combine the measurement data with the camera signal and send the output video signal first to an encoder and then to the transmitter. In this way I cannot only see the real-time picture from the cockpit but also see various relevant data at the same time!

Who knows, maybe I will even fit an autopilot in my aeroplane. That would make my hobby a lot cheaper!

Serially Programmable
Compact and universal

Burkhard Kainka

The right crystal controlled frequency at the right time: With this DIP outline oscillator there's no need to wait for a special crystal to be made up. You can program its frequency precisely using the RS232 port of your PC. Low phase noise is a feature that makes this PLL oscillator suitable for demanding RF applications also.



This multipurpose clock generator circuit uses the CY27EE16 programmable oscillator of the CyberClock family of devices from Cypress. Regular readers will remember this chip from the November 2004 issue of *Elektor Electronics.* The entire oscillator circuit shown in **Figure 1** comprises of little more than the clock generator chip together with a voltage regulator and fits neatly onto a small PCB. Altogether it makes an extremely versatile clock generator that will no doubt find a use in many applications. The 10 MHz reference crystal shown in the parts list is fitted to the PCB via a socket. A different value may be substituted if it is necessary to provide two or more specific output frequencies simultaneously for a particular application.

The chip communicates with the PC over an I^{2}C bus using handshake signals of the RS232 computer interface.

Crystal Oscillator

Data from the slave is read on the CTS line. The signals DTR and RTS are read by DCD and DSR and these may be used later in software for possible expansion of the design, they are also used to control the data rate in a USB/RS232 adapter.

The software allows the user to load settings to the internal EEPROM or RAM register. In many applications the oscillator chip will be built into equipment and its clock frequency parameters will be programmed in the EEPROM. It will always be possible to change the value later if the frequency needs to be altered. In other cases where the application calls for the output frequency to be selectable the connection cable will remain attached to the PC and new data can be sent to the clock chip whenever necessary.

The PCB is the same outline as a 24 pin DIL IC. Two rows of pins can be fitted to allow the assembly to be plugged into a standard 24 pin IC socket. A

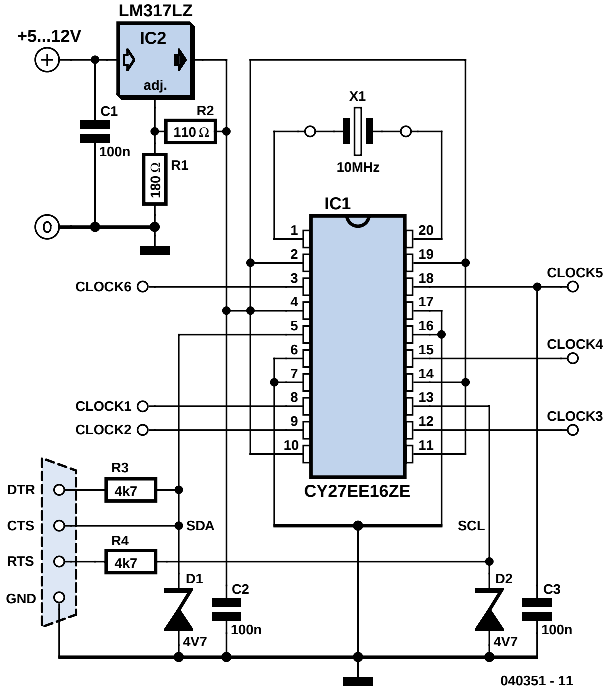


Figure 1. The programmable clock circuit diagram.

COMPONENTS LIST

Resistors:
R1 = 180Ω
R2 = 110Ω
R3, R4 = 4kΩ7

Capacitors:
C1,C2,C3 = 100nF ceramic

Semiconductors:
IC1 = CY27EE16ZE
IC2 = LM317LZ, TO92
D1, D2 = Zener diode 4.7V 0.5W

Miscellaneous:
10.0MHz quartz oscillator module with socket
10-way boxheader
10-way flatcable with IDC socket and 9-way sub-D socket
PCB, ref. 040361-1 from The PCBShop

Note:
Ready populated and tested boards as well as bare boards available from AK-Modul-Bus GmbH, Germany (www.ak-modul-bus.de).

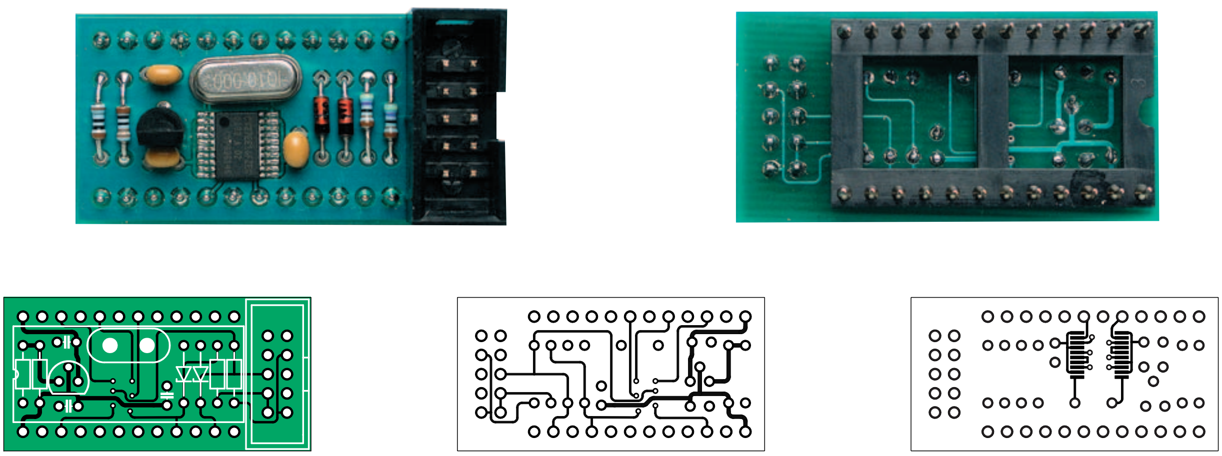


Figure 2. The PCB is arranged as a 24 pin DIP outline.

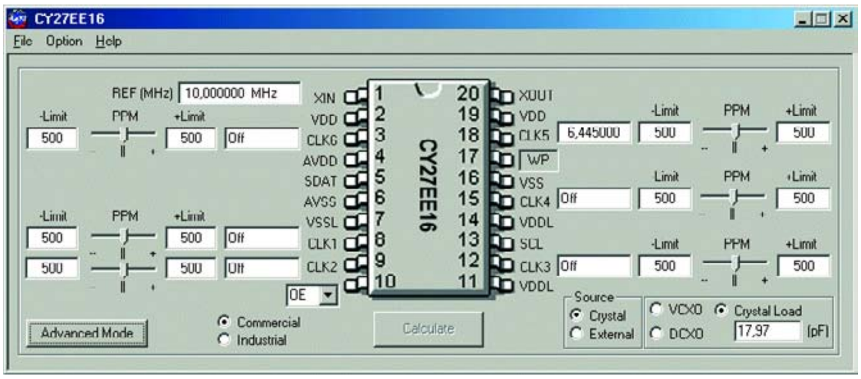


Figure 3. Frequency setting using CyberClocks.

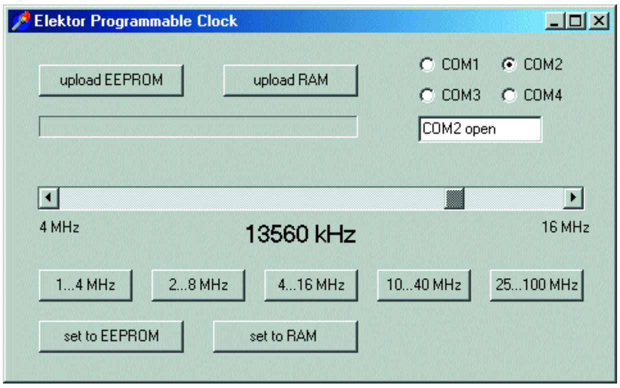


Figure 4. The Clock.exe program.

10-way upright connector accepts the cable connecting it to the PC. This is not too much of a problem because the cable is only needed when the oscillator requires reprogramming. The data and clock signals to the CY27EE16 are also connected to pins on the IC socket so that an on-board processor in the target system with suitable software routines could reprogram the clock generator directly. **Table 1** shows the pin-out assignments of the 24-pin socket.

Assembling the components onto the PCB should not be too much of a problem for anyone with a little experience in mounting SMD components. One method using generous amounts of solder and desoldering braid to clean up has often been described in *Elektor Electronics*. When the finished PCB is connected to a suitable power supply a voltage of 3.3 V can be measured on the CY27EE16 supply pins and the crystal should be oscillating. All signal output pins will however be set to high impedance mode until the chip is programmed.

The program *Clock.exe* together with some helpful examples can be downloaded free of charge from the *Elektor Electronics* web site www.elektor-electronics.co.uk. The file number is **040351-11.zip**. For complete chip programming flexibility it will also be necessary to download the *CyberClocks* program and this can be found at www.cypress.com.

Change the settings with CyberClocks

All the configuration data for the chip can be programmed using *CyberClocks*. It is necessary to specify the reference crystal frequency and also which output pin (or pins) will be used and the desired output frequency. **Figure 3** shows an example using the

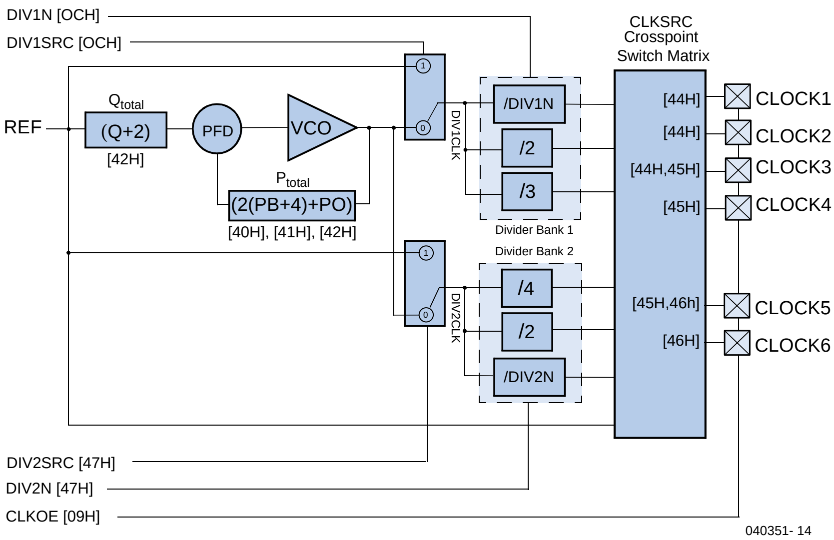


Figure 5. The programmable divider and register.

Table 1. Pin-out assignments for the oscillator PCB.			
1	3.3 V O/P	+5 V	24
2	Ground	Ground	23
3	Clock1	Clock6	22
4	Ground	Ground	21
5	Clock2	Clock5	20
6	Ground	Ground	19
7	Clock3	Clock4	18
8	Ground	Ground	17
9	SCL	Ground	16
10	Ground	Ground	15
11	SDA	Ground	14
12	Ground	Ground	13

CLK 5 output. Information containing all register values and settings are stored as a binary file in the PC and downloaded to the chip.

Data transfer using Clock.exe

The *Clock.exe* program writes data into either the internal EEPROM of the CY27EE16 or into RAM. The register contents in RAM become immediately effective whereas the settings stored to non-volatile EEPROM will only become effective after the chip undergoes its next power up when all EEPROM data will be written to RAM. The software includes some example files to generate different frequency outputs from a 10-MHz reference clock. The buttons labelled 'upload EEPROM' and 'upload RAM' transfer the binary file generated in the Cypress *CyberClocks* program. In the lower part of the program window there is possibility to directly change the frequency settings without recourse to the *CyberClocks* program. It is assumed that a 10 MHz crystal is used and a sample file with the desired output configuration has already been loaded. *Clock.exe* does not change all the registers but just programs new values to the PLL and the 7-bit divider DIV1N for divider bank 1. The maximum PLL range from 100 MHz to 400 MHz is used with the smallest step change of 250 KHz. The different output frequency ranges are produced by the post divider and give the following step changes in output frequency:

Output frequency in MHz	Steps in kHz
1 to 4	2.5
2 to 8	5
4 to 16	10
10 to 40	25
25 to 100	62.5

Multiple clock sources from one oscillator

The *cyberclock* clock generator has six frequency outputs and its important to understand how the internal divider arrangement places limitations on programming the output frequencies. There are two 7-bit programmable internal divider banks that can be configured to divide down the output of the crystal or the PLL. Two alternative dividers in each bank offer fixed division ratios. The PLL output can for example be made available on three output pins, the first pin will be the PLL frequency divided by 127, the second will be the PLL frequency divided by 2 and the third will be divided by 3. The other divider bank can be used to divide the crystal frequency. A more detailed description can be found in the data sheet from Cypress.

In a typical RF application there is often the need for more than one frequency source in the whole system. The CY27EE16 contains only one PLL but it can generate several different frequencies if the crystal frequency is chosen carefully. **Figure 5** shows an example where two oscillator frequencies are produced by the clock generator for a DRM type receiver. The desired second oscillator frequency of 467 KHz is achieved with a deviation of 300 Hz using a division factor of 19 after the crystal oscillator. The first oscillator frequency uses the PLL and achieves the desired frequency of 6550 KHz with an offset of just 16 Hz.

The *Cyberclock* also allows fine adjustment of the crystal frequency with a programmable load capacitor. Its value can be adjusted in the range from 7.8 pF to 32 pF in steps of 0.1 pF. To try out this feature the author required a highly stable precise frequency of 45.012 MHz for a receiver with a 45 MHz crystal filter to give a 12 kHz output signal. A rummage around the spares box soon provided a number of suitable crystals and with the help of the calculator it was determined that a standard 8.867238 MHz PAL crystal would again be suitable and would provide an output frequency closest to the required signal.

The frequency difference was only –210 Hz or 4.7 ppm. The crystal would only need to be 'pulled' by 40 Hz to achieve exactly the required output frequency. This last adjustment was made using the time-honoured principle of trial and error and the final 'trimmer' value was found to be 22 pF.

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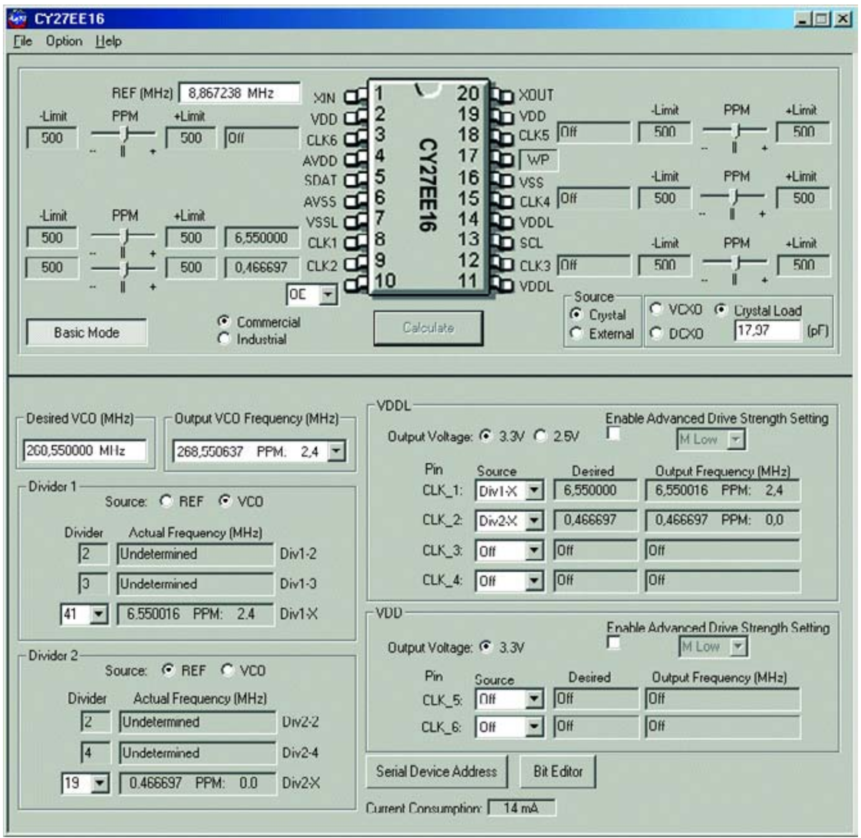
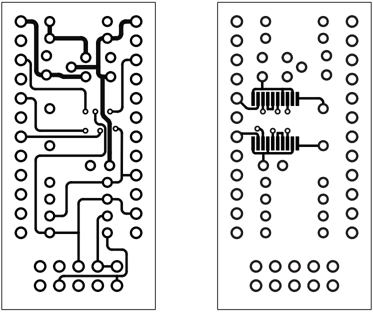


Figure 6. Advanced Mode settings.



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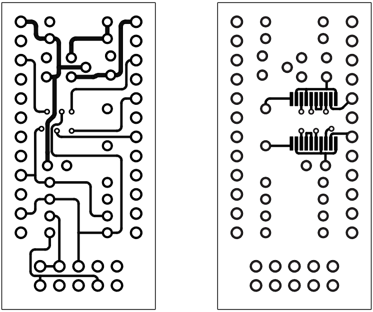

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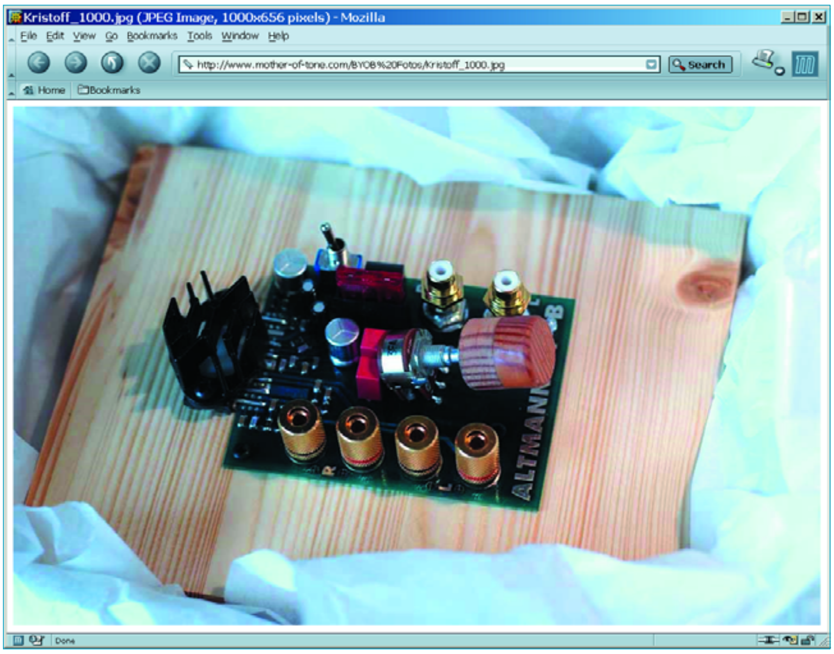
AUDIO-TWEAKS

From demagnetising to varnishing

Harry Baggen

For true audio enthusiasts, nothing's too far out when it comes to improving the sound quality of their hi-fi or surround-sound systems. Naturally, you can spend a lot of money on cables and other accessories, but there are a lot of possibilities that go even further, such as equipment modifications and special treatments for LPs and CDs.



Despite the mountains of cheap audio equipment sold every year, there is still a sizable group of people devoted to true hi-fi reproduction in both stereo and surround-sound versions. Many of these confirmed high-enders even resort to rather unusual methods to squeeze the last drop of performance out of their gear and obtain just a smidgen more detail from the speakers. Here we have collected several of these methods, which range from quite serious to rather questionable.

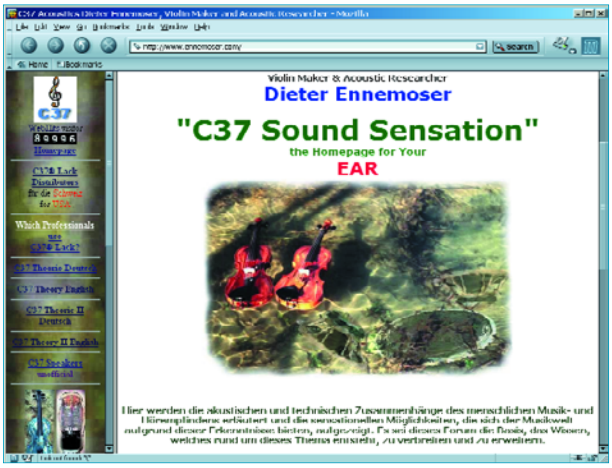
Let's start off with a few in the latter category. There are various tips going around for making CDs sound better. For instance, it's supposed to help if you put the CD in the freezer for a few hours before playing it. That doesn't cost anything, so it's easy to try it out. Another way to improve the sound of a CD is to demagnetise it before you play it. Several manufactures market commercial devices for this purpose.

One of the strangest tricks for improving the sound of a CD is one that we recently found on the **Audio Tweaks** website [1]. There someone recommends spraying a bit of Pledge (a sort of wax polish for wooden furniture) into the air and then dragging the data side of the CD through the aerosol fog hanging in the air [2]. That will cause microscopic drops of liquid to adhere to the surface of the CD, which apparently affects the playback. This seems rather risky to us from a technical viewpoint, since residues of the polish will most likely be spun off from the CD and end up in the drive, and that can have disastrous consequences for the laser and the slide mechanism. At least you can't say we didn't warn you. This site has a collection of more than 300 audio tweaks, and it's quite amusing to browse through them.

Another rather unusual tip involves 'C37 lacquer', a sort of magic potion formulated by the Austrian violin maker **Dieter Ennemoser** [3]. It's a sort of violin varnish that also appears to have special properties for audio equipment, and everything that is coated with it will only vibrate at natural harmonic frequencies (based on carbon at a body temperature of 37 °C – hence the name). We must admit that if you coat your speaker cones with this stuff, it's bound to affect the sound quality. Herr Ennemoser's site also has various links to sites with descriptions of treating amplifiers and the like with this varnish. Even coating the ICs on a circuit board is supposed to lead to a perceptible improvement in sound quality. But you should first read what various people have to say and draw your own conclusions.
The activities the American company **CyroPlus** [4] are less strange, and they also have a technical explanation. What they do is to chill interconnecting cables and speaker cables, as well as individual valves and ICs or even entire circuit boards, down to absolute zero. The idea behind this is that it restores the original crystalline structure of the materials in the cables or components. You can buy treated cables from them, but you can also place an order to have your own cables chilled. The prices are quite reasonable, aside from the cost of shipping to the US.

It used to be that tuning or tweaking hi-fi equipment was primarily something done by hobbyists, but in the meantime it has turned into a professional business. Well-known companies, such as **Van Medevoort** [5] in the Netherlands and **Swoboda Audio** [6] in Germany, offer modified consumer equipment with all sorts of tweaks, such as optimising the damping of the enclosure, upgrading the power supply, or modifying subcircuits (such as using special opamps in a CD output stage). There's no doubt that such extensive measures definitely affect the sound quality and can yield very good end results. In most cases, though, it's a costly enterprise, since it takes a lot of work and time to modify an existing piece of equipment.

Finally we have a nice idea for the minimalists among our hi-fi enthusiasts. When you think of a hi-fi amplifier, you usually think of a big box with a hefty power supply (and just now we want to leave aside controversial issues such as how much feedback gives the best results). Under the motto 'keep the circuitry and circuit-board tracks between the input and output of the amplifier to a minimum', the German **Charles Altmann** [7] has created an amplifier stage consisting of nothing more than a power IC on a circuit board with input and output sockets

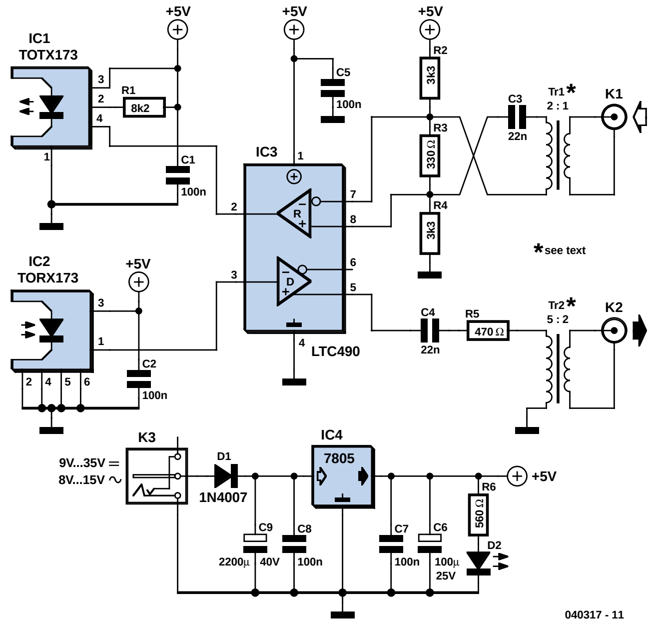


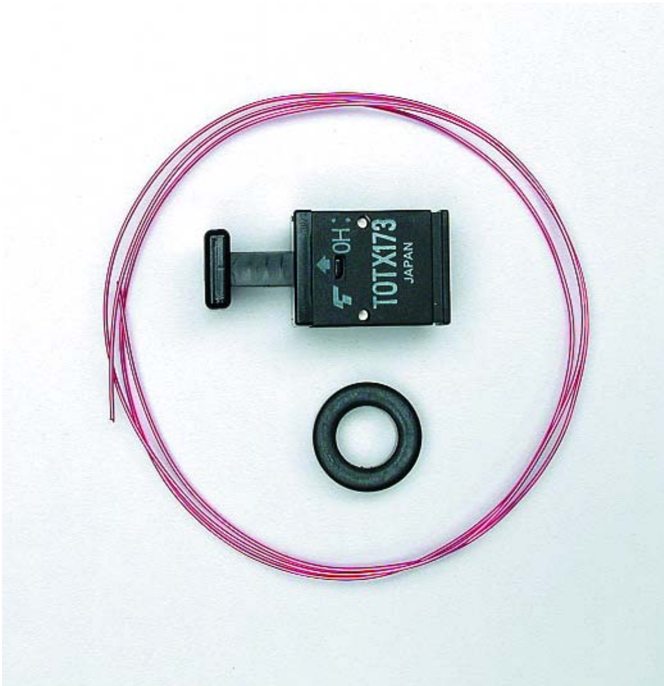
and a potentiometer. It bears the suggestive name 'BYOB amplifier', and the circuit is powered by a car battery. As Altmann finds that a metallic enclosure generates undesirable vibrations, the amplifier is supplied on a small, nicely finished wooden board. The photos on his website should inspire quite a few audio enthusiasts to make something similar on their own.
As you can see, hi-fi is truly an open-ended subject!

(045111-1)

Internet addresses

[1] Audiotweaks: www.audiotweaks.com/collection_ascend.htm
[2] Pledge-tip: www.audiotweaks.com/tweaks/tweak_265.htm
[3] Dieter Ennemoser: www.ennemoser.com/
[4] CryoPlus: www.cryoplus.com/audio.html
[5] Van Medevoort upgrades: www.audioart.nl/nl/upgrade.html
[6] Swoboda Audio: www.swobodaaudio.de/
[7] BYOB amplifier: www.mother-of-tone.com/byob.htm





Bidirectional S/PDIF Converter

Joseph Kranz

Although S/PDIF is the standard when it comes to conveying digital audio signals in consumer electronics equipment, difficulties may be encountered if you try to connect up two apparatus having digital connections. Cheap equipment usually has only one connection (coaxial or optical), while more up-market products will generally offer both variants. When interconnecting CD and DVD players, MD recorders and surround receivers you'll soon find that a converter from optical to coaxial or the other way around is useful if not indispensable.

The bidirectional (two-way) S/PDIF converter shown here employs just one IC to handle the conversion both ways. The LTC490 (IC3) is a differential line transceiver originally intended for use in RS485 applications. Using R2, R3 and R4 the driver part used here to get from coaxial to optical is biased to the centre of the permissible common-mode voltage (between –7 V and +12 V for this IC). Moreover, a small offset is added (approx. 0.2 V) to ensure a low level is defined in the absence of an S/PDIF signal.

The input signal is supplied by Tr1, a small transformer you need to wind yourself on an Epcos core type 64290-L44-X830. The primary consists of six turns of 0.3 to 0.5 mm diameter enamelled copper wire (SWG30 – SWG26). The secondary side is made by winding 12 turns of the same wire. Incidentally, C3 prevents the transformer secondary from short-circuiting R3. The capacitor has a negligible effect on the /PDIF signal.

That concludes the more complex part of the circuit, the receiver section of IC3 being followed by a dead standard application circuit around the TOTX173, an optical S/PDIF driver.

The other part of the circuit, i.e., the converter from optical to coaxial, is even simpler. Optical receiver IC2 works happily with the addition of just one capacitor that decouples the supply voltage. The output signal is applied directly to the driver section inside IC3. At the output of that IC, we find another DIY transformer and a coupling capacitor to prevent short-circuiting. Resistor R5 serves to adapt the output level to the S/PDIF standard.

Transformer Tr2 is wound on the same type of core and using the same wire as Tr1, but this time with 20 turns at the primary and 8 turns at the secondary.

The home made transformers are, of course, not necessary if you do not require electrical isolation between the input and output signals. In that case, you may replace Tr1 with a 100-Ω resistor between ground and the signal line (directly at the input, i.e., ahead of C3). The same applies to Tr2 — this transformer, too, may be replaced by a 100-Ω resistor to ground. It, too, is connected directly to the input, so in this case ahead of R5. Note however that R5 then takes a value of 330 Ω instead of 470 Ω.

We would like to add that this circuit may also prove useful in a computer environment, some sound cards having an on-board S/PDIF output that supplies a signal at TTL level. Such a TTL signal may be applied to the D input of IC3. Components IC2 and C2 are then omitted to add a coaxial S/PDIF output to your trusty PC. Alternatively, the sound card output signal may be applied directly to pin 4 of the TOTX173. If you do so, do not forget to connect the 8.2-k resistor and the 100-nF capacitor (R1 and C1 in the diagram).

The reverse applies as well — you can also use the circuit to apply an S/PDIF signal to the sound card (that is, if it has a TTL compatible input). In that case, components IC1, R1 and C1 or C4, R5 and Tr2 are omitted (for the coaxial and optical input, respectively).

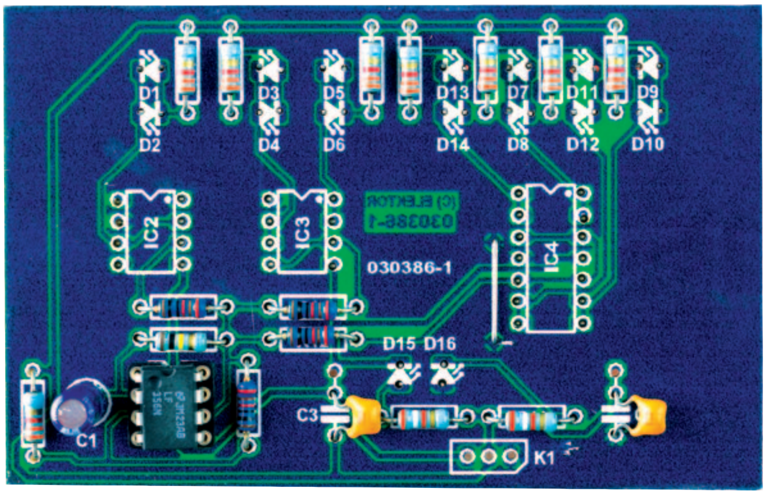
The supply section consists of a 7805 voltage regulator in its standard configuration. Diode D1 not only serves as a reverse polarity protection device if a DC-output battery eliminator is used, but also allows the circuit to be powered from an 8-15 VAC source. If you use (parts of) the circuit in combination with your PC, the 5-V supply voltage may, of course, be 'stolen' from the PC's power supply. This makes the circuit around IC4 unnecessary, except of course the IC decoupling capacitors C1, C2 and C5.

(040317-1)

OPAMP TESTER

Quick go/no-go testing for operational amplifiers

Dirk Schumacher



Not a microcontroller in sight, and yet this item of test equipment will be found very handy in any electronics lab.

If you ever want to use an operational amplifier salvaged from an old circuit or from the junk box, you will take into account that the device may no longer work properly. Unfortunately, it is not as easy to perform a quick go/no-go test on an operational amplifier using a multimeter as it is with a resistor, coil, fuse, diode or even a capacitor. Although an opamp tester would obviously be a useful thing to have, a dedicated instrument is not easy to come by. However, you can build this item of test gear yourself in a few minutes, and it certainly won't break the bank.

The test circuit just consists of a simple squarewave oscillator (IC1) which oscillates with a frequency of about 1 Hz. The output of the oscillator, on pin 6, therefore swings between 'high', $+(V_b–0.5$ V), and 'low', $–(V_b–0.5$ V), with a period of about 1 s.

Most operational amplifiers come in packages containing one, two or four identical circuits. All the manufacturers have standardised on three pinouts, irrespective of the part number of the device. (There are some devices with non-standard pinouts, but they are very rare.) These are shown in the circuit diagram as 'type 1', 'type 2' and 'type 3'. The part numbers of a few 'general purpose' opamps are also shown. Single and dual opamps come in eight-pin packages, quad opamps in 14-pin packages. All the opamps under test are identically wired, as voltage followers/impedance converters. The output voltage is equal to the voltage at the non-inverting input, and so the squarewave from IC1 will therefore be present on all the relevant pins.

The results of the test are displayed using low current LEDs. If the output of the opamp is high, the red LED will light; if the output is low, the yellow LED will light. The opamps under test will need to be able to sink and source a current of at least 2 mA.
The test unit is powered from two 9 V PP3 (6F22) type batteries (BT1 and BT2). D15 and D16 indicate when the supply voltage is present on all the relevant pins of the oscillator and of all the test sockets.

As a glance at the printed circuit board layout in **Figure 2** will reveal, populating the board should present no difficulties. There are many LEDs, which must all, of course, be fitted the right way around; the same goes for the small electrolytic capacitor and the oscillator IC. Don't forget the one wire link, which connects the two parts of the ground plane. It is worth clearly marking the position of pin 1 of the test sockets on the front panel of the enclosure, to ensure that devices under test are never inadvertently inserted incorrectly. Otherwise the opamp being tested will quickly give up the ghost, as you will be able to demonstrate when you subsequently insert it correctly!

(030386-1)

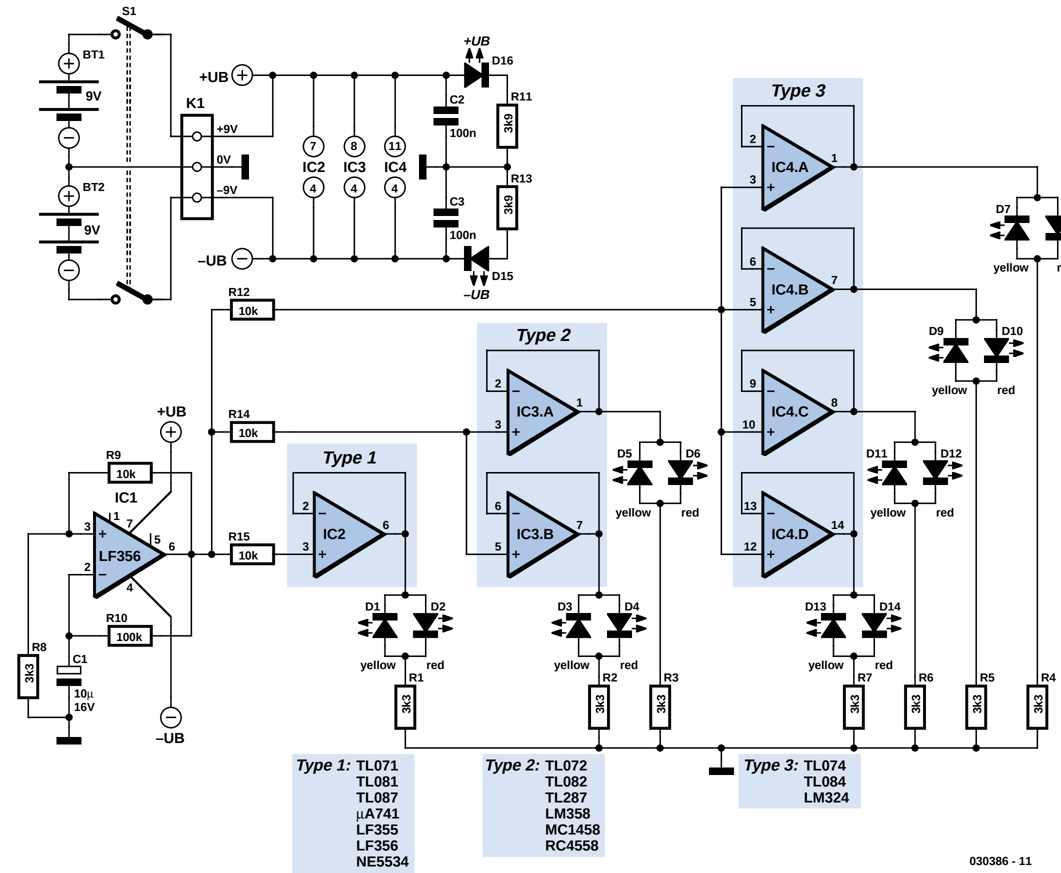


Figure 1. The circuit can test single, dual and quad opamps.

COMPONENTS LIST

Resistors:
R1-R8 = 3kΩ3
R9,R12,R14,R15 = 10kΩ
R10 = 100kΩ
R11,R13 = 3kΩ9

Capacitors:
C1 = 10µF 16V radial
C2,C3 = 100nF

Semiconductors:
D1,D3,D5,D7,D9,D11,D13,D15,D16 = LED, 3mm, yellow, low current
D2,D4,D6,D8,D10,D12,D14 = LED, 3mm, red, low current
IC1 = LF356

Miscellaneous:
2 8-way IC socket
1 14-way IC socket
S1 = double-pole on/off switch
BT1,BT2 = 9-V battery with clip-on connector
Enclosure, Hammond type 1591B
1 wire link
PCB, order code **030386-1**, see Readers Services page

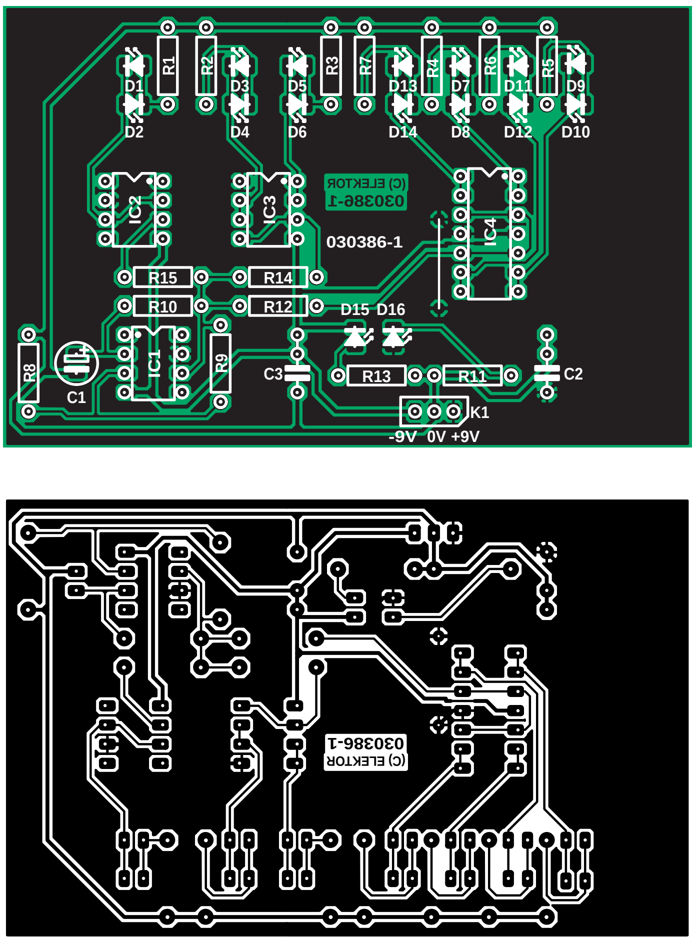
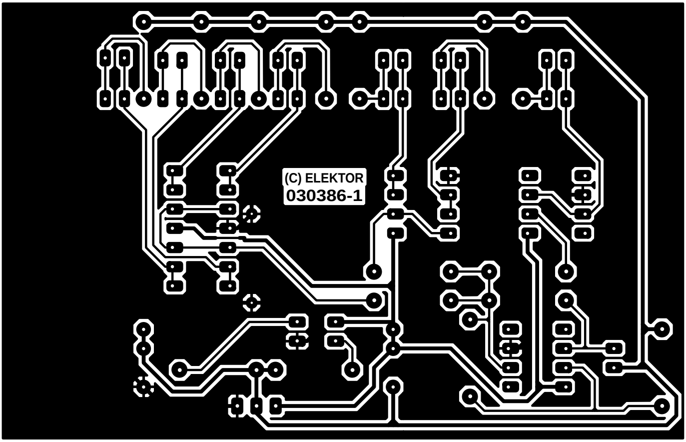


Figure 2. The printed circuit board is clearly laid out.



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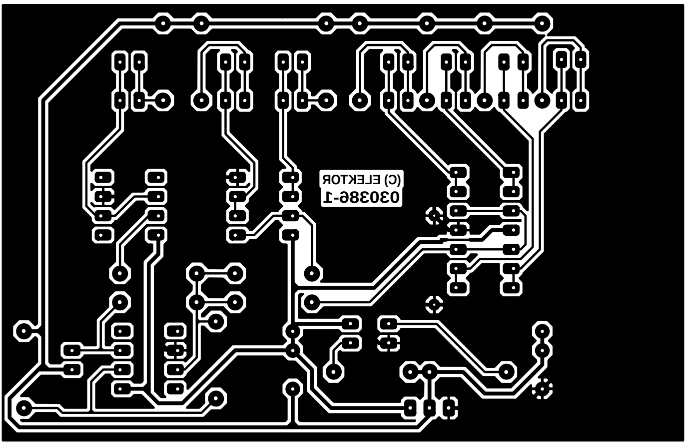

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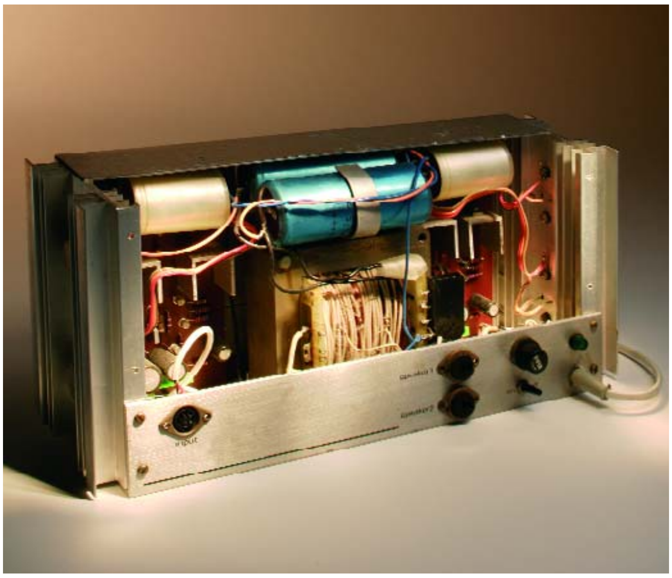
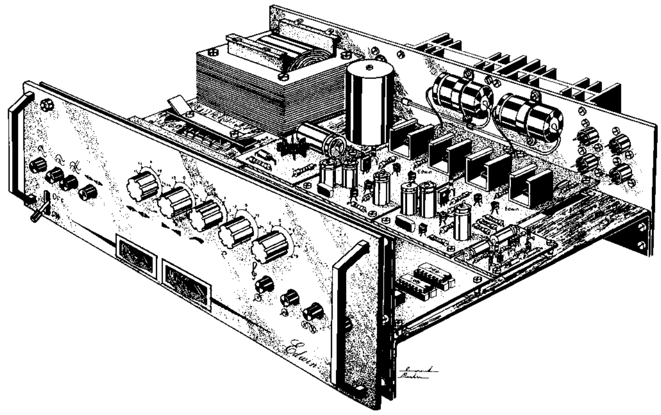




'Edwin' Audio Amplifier (1975)

Jan Buiting

Blending in beautifully with this month's focus on all things Audio I'd like to take you back to September 1975 when *Elektor* magazine published an audio amplifier named 'Edwin'. 1975 was the first full year the English-language version of Elektor was published. Not surprisingly, the then editors looked at designs that had already proved extremely popular in the German and Dutch editions of the magazine, which had been on the market for some years already. 'Edwin' was a beefed-up version of an even older design which I have been unable to dig out of the archives. Compared to the earlier design, Edwin doubled the output power from just 20 to a mighty 40 watts (maximum).

The huge success of the Edwin publication can be gauged from the resultant PCB sales which, according to some of my older colleagues, achieved many thousands — something we can only dream of in this day and age of DIY audio electronics having taken a very distant back seat to microcontrollers and PC ancillaries.

The September 1975 'EPS' (Elektor Print Service) column lists the 'Edwin' bare board as item no. '97-536' at a price of £1.20 including 25% VAT, excluding 15p P&P. If that strikes you as odd, you need to know that in 1975 a copy of the magazine would set you back a whole 35p, and a 2N3055, 70p.

The design of the Edwin would make modern audio amplifier designers either shudder or burst out laughing. The Edwin power output stage comprised 'workhorse' transistors like the BD138, BD130 and 2N3055, while a 2500-microfarads output capacitor was used. The amplifier had common-or-garden small-signal transistors like the famous BC107/BC108 in the prestages and worked from a single-rail 42-VDC supply. Two no-load supply voltages were stated. One was 42.5 VDC for 20 watts into 8 ohms, or 34 watts into 4 ohms. The other was 46.5 VDC for 24 watts into 8 ohms, or 42 watts into 4 ohms. These values were given to help readers in the choice of the mains transformer's secondary voltage — 30 V_{rms} or 33 V_{rms}.

The amplifier was claimed to be "*unusual in that it embodies two types of output stage in one amplifier*". The nitty-gritty was shifting biasing levels of two trannies in the design — depending on the signal level applied, your Edwin would behave as a class-A or a class-B(-ish) amplifier, allegedly without running into too much cross-over distortion. The design was advertised as "unconditionally stable" and "short-circuit proof", which was certainly revolutionary at the time.

In 1975, no photographs were printed with any of the published designs, probably owing to cost or the poor paper quality. Instead, enticing 'artist's impressions' were drawn by our former colleague Laurent Martin and printed, in the case of the Edwin, across a full page. The above photograph shows a stereo version of the Edwin I found in our lab's relics cabinet. It was built with, in my view, complete disregard for electrical safety — note the way the transformer winding was adapted, probably to tweak the secondary voltage.

I was not a little amused by the editors first calling Edwin a "*high-quality 40-W audio design*" in the introduction and then concluding the same article with "*Whilst the Edwin amplifier meets an exacting specification this is no reason to recommend its construction by the Hi-Fi enthusiast*", which strikes me as much more realistic and modest at the same time. No matter, thousands of readers enjoyed building and using this legendary amplifier!

(045106-1)

Retronics is a monthly column covering vintage electronics including legendary Elektor designs. Contributions, suggestions and requests are welcomed; please send an email to editor@elektor-electronics.co.uk, subject: Retronics EE.

QUIZZ' AWAY

Martin Ohsmann is a Professor of Electrical Engineering and Information Technology at FH Aachen and a long-time contributor to Elektor Electronics. Through Quizz'away he aims at stimulating thought, speculation, construction and simulation as well as raise interesting questions.

Oddball Oscillator

This time we present yet another problem covering elementary circuit design. The schematic in **Figure 1** shows an opamp circuit supplying a rectangular voltage with a frequency of about 34 kHz. As a remarkable aspect, components establishing positive feedback and normally required to create oscillation, are not available.

The oscillogram in **Figure 2** shows the voltages measured on the coil (U1, upper trace) and at the output (U2, lower trace). The photograph in **Figure 3** shows the circuit built in flying wire fashion. The oscillation frequency is so low that parasitic capacitances, inductances or stray coupling have no effect.

This month's question to you all:
How does it work?

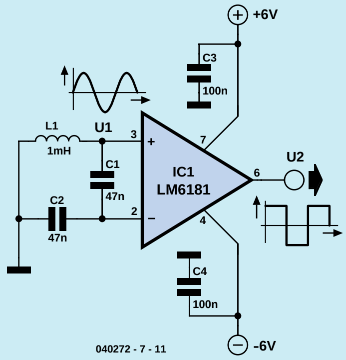


Figure 1. An oddball oscillator.

Tip: the circuit will not function with any old operational amplifier, but only with a 'certain kind', so it may be a good idea to get your hands on a copy of the LM6181 datasheets (National Semiconductor).

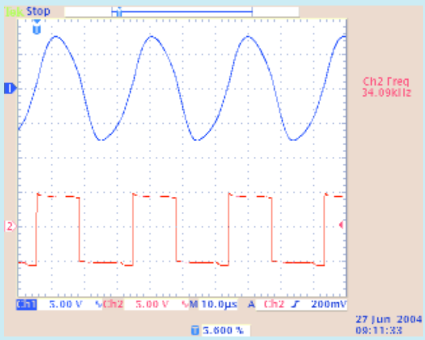


Figure 2. Oscillograms produced by the circuit.

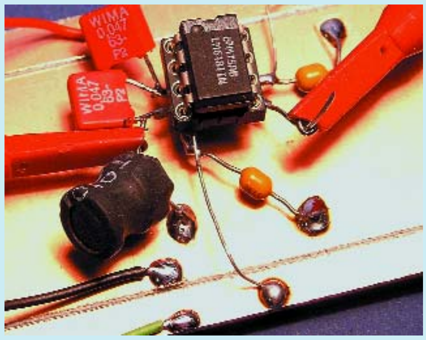


Figure 3. The oscillator in flying-lead construction.

Quizz'away and win!

Send in the best answer to this month's Quizz'away question and win a ready-built and tested

Elektor Gameboy Digital Oscilloscope (GBDSO) extension board with PC software, worth £103.

All answers are processed by Martin Ohsmann in co-operation with Elektor editorial staff. Results are not open to discussion or correspondence and a lucky winner is drawn in case of several correct answers.

Please send your answer to this month's Quizz'away problem, by email, fax or letter to:
Quizz'away, Elektor Electronics, PO Box 190, Tunbridge Wells TN5 7WY, England. Fax (+44) (0)1580 200616. Email: editor@elektor-electronics.co.uk, subject: 'quizzaway 3-05'.

The closing date is 25 March 2005
(solution published in May 2005 issue).
The outcome of the quiz is final.
The quiz is not open to employees of Segment b.v., its business partners and/or associated publishing houses.

As of the September 2004 issue Quizz'away is a regular feature in Elektor Electronics.
The problems to solve are supplied by Professor Martin Ohsmann of Aachen Technical University.

Solution to the January 2005 problem

(p. 78; Measurements using a probe — never a problem! For sure?)

For the second time since starting this series, surprisingly few answers were received, although the majority of quiz participants did manage to solve the problem!

The circuit consists of two parts. The first opamp is configured as a classic Schmitt trigger oscillator causing a nearly triangular-shaped voltage on the capacitor.

So far, nothing unusual. However, the configuration of the second opamp, shown here in **Figure 4** should strike you as rather less conventional. With the non-inverting input held at half the opamp's supply voltage, it is effectively connected to 'ground'. The T-junction formed by the 1-megohm resistor and the 1-microfarad capacitor is only effective at very low frequencies. In fact, it only ensures the DC biasing of the opamp remains undisturbed if the oscilloscope input is set to 'AC'.

For the 150-Hz triangular signal, then, the circuit may be further simplified to that shown in **Figure 5**, which now includes the oscilloscope (input) and the probe. The bottom line, then, is that the internal resistance of the probe or the oscilloscope input now determines the opamp's feedback path!

The opamp will attempt to adjust its output voltage in such a way that the differential voltage between its inputs becomes zero. Since negligible currents flow into the opamp inputs, the current that maintains a constant potential of U/2 at the inverting input must be supplied by the opamp's output, and, therefore, flow through the feedback impedance. Consequently, the feedback path carries a triangular-shaped current whose excursion (level) is independent of the feedback impedance value. Seen from the oscilloscope, the circuit presents a constant-current source! At a constant current, the 9-megohm series resistor in the probe is largely irrelevant, while the current through the scope's 1-megohm input resistance is independent of the probe setting.
The above effect only occurs within the drive margins of the opamp. Also, the opamp circuit has to be supplied from a battery to allow the oscilloscope to be connected up in 'floating' fashion as indicated. It is also not allowed to connect the 'scope the other way around because junction 'A' in the circuit s extremely likely to react to hum and other noise.

As the decisive factor, the circuit demonstrates that a probe will only 'divide' properly if the internal resistance of the signal source is small relative to the probe impedance.

Moral of the story:
You will realise, of course... the resistance of the source!

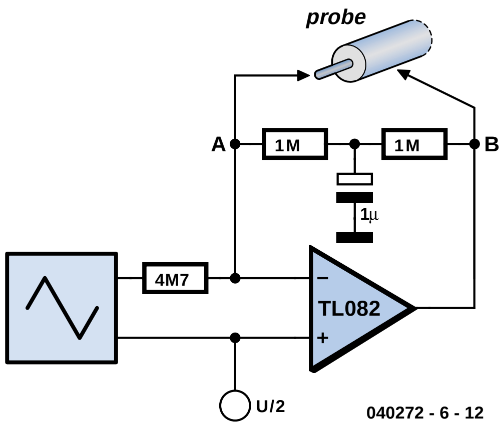


Figure 4. Simplified circuit of the 'probe mix up'.

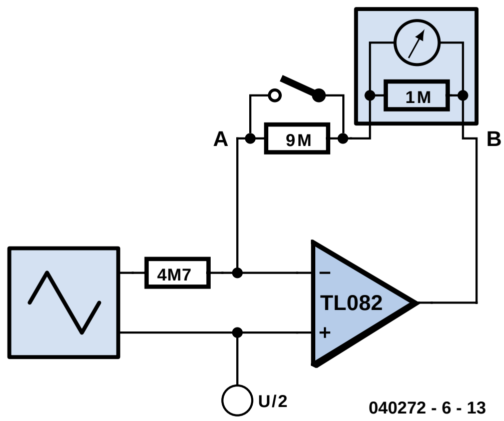
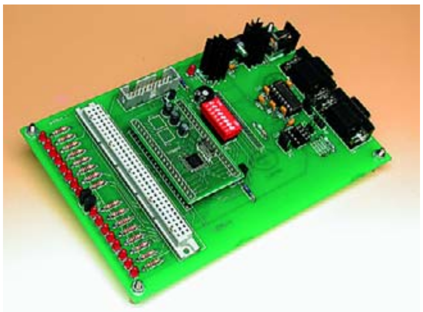


Figure 5. The bare essentials.

sneak preview



LPC210x 'ARMee' Development System (2)

Having digested the background article on ARM processors in the March 2005 issue you're ready to proceed with an ARM development board that follows in the footsteps of illustrious predecessors like our 'PICee' and 'AVRee' systems. A special feature of the project is the transportability (literally) of the ARM CPU sub-board — once programmed on the development system it can be unplugged from and migrated to a target application. Tony Dixon has the full story.

USB-GPS

Unfortunately this article could not be included in the current issue due to lack of space. It describes an interface between navigation applications and a miniature GPS module coupled to an equally small active antenna. The modular structure of the project allows you to use selected parts for your own applications.

SC-Analyser

Transistors are found in lots of electronic circuits. However, as many a repair engineer will avow, these components are not always easy to identify, which makes checking them — and finding a spare or substitute — a bit of a problem. Our SemiConductor Analyser is not only able to discriminate between FETs and bipolar transistors, but also snoop the pinout and extract the main electrical parameters like H_{FE} from an unknown device.

Theme Plan for 2005

January . . . . .Power Supplies
February . . . .Wireless
March . . . . . .Sound
April . . . . . . .Microcontrollers
May . . . . . . .Sensors
June . . . . . . .Environment
July/August . .Summer Circuits
September . . .Test & Measurement
October . . . . .Security
November . . .CAD Software
December . . .Optoelectronics

Also...

Long Lines in Chips; Delphi Course (4); Simple LiPo Charger; 27C512 Emulator; The Digital Future.

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UK subscribers will receive the magazine a few days before this date. Article titles and magazine contents subject to change.

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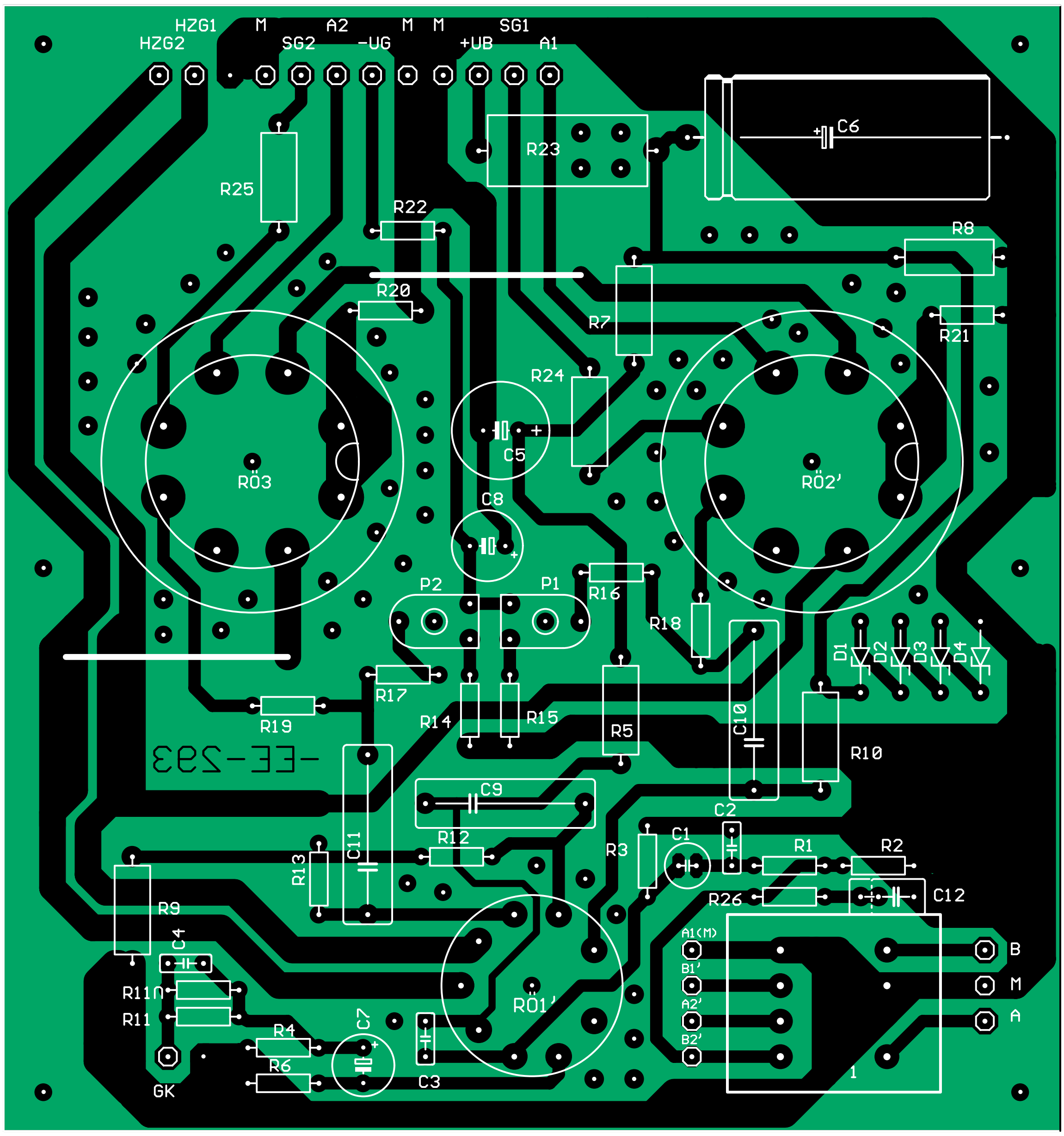
EL156 Audio Power Amplifier



COMPONENTS LIST

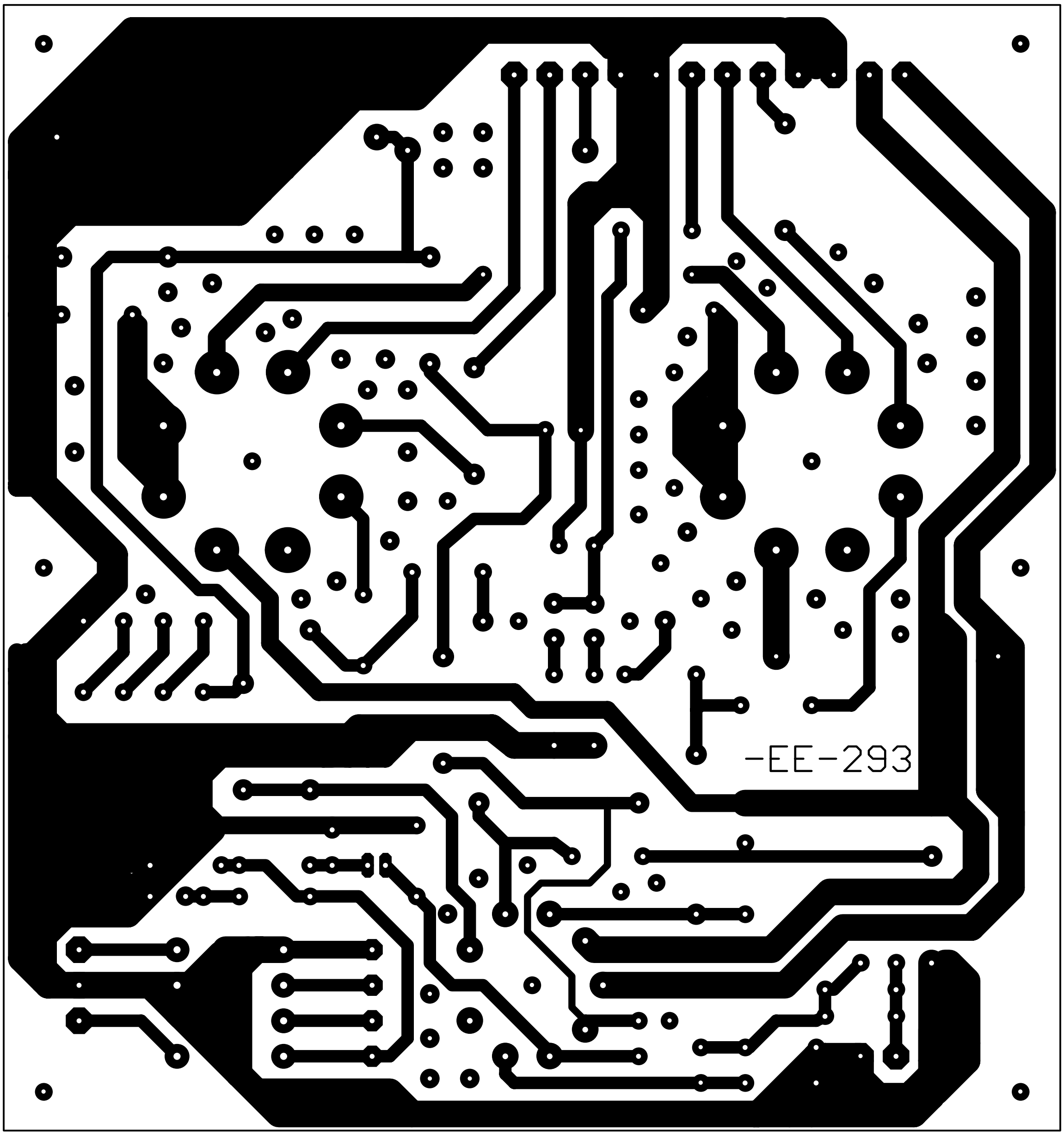
Main board

Resistors:
R1 =1k
R2,R3 = 100 k
R4 = 220 Ω
R5 = 18 k, 2W
R6 = 180 Ω
R7 = 2k2, 2W
RS = 10 k, 2W
R11 = 1k3
R12 = 680 k
R13 = 220 Ω
R14,R15 = 10 k
R16,R17 = 68 k
R18,R19 = 1 k
R20,R21 = 10 Ω, 2W
R22 = 1k5
R23 = 10 k, 4.5 W
R24,R25 = 100 Ω, 2W
R26 = 4k7
P1,P2 = preset 5 k

Capacitors:
C1 = 2µ2, 50 V bipolar
C2 = 10 p ceramic
C3 = 33 p ceramic
C4 = 1n5 ceramic
C5 = 10 µ, 450 V
C6 =22 µ,500 V
C7 = 220 µ,25 V
C8 = 100 µ, 63 V
C9-C11 = 470 n, 630 V MKS4
C12 = 2n2 ceramic

Semiconductors:
D1 = Zener diode 56 V, 1.3 W
D2-D4 = Zener diode 100 V, 1.3 W

Miscellaneous:
Rö1 (V1) = ECC81
Rö2,Rö3 (V2,V3) = EL156
Ü1 (Tr1) = transformer type E-1220
Ü2 (Tr2) = transformer type B-2156
1 Noval (9-way) socket, ceramic, PCB mount
2 Octal (8-way) socket, ceramic, PCB mount



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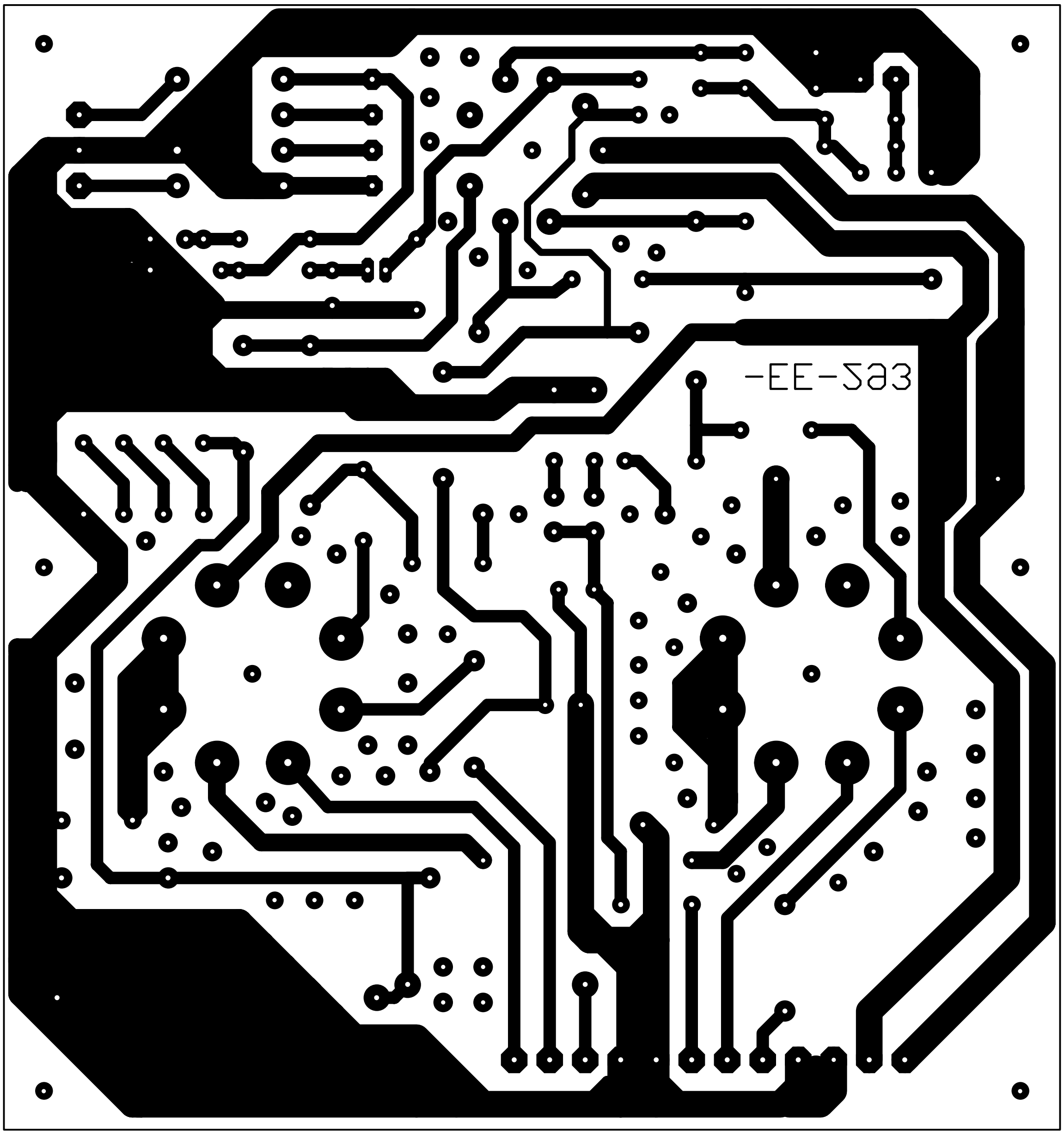

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COMPONENTS LIST

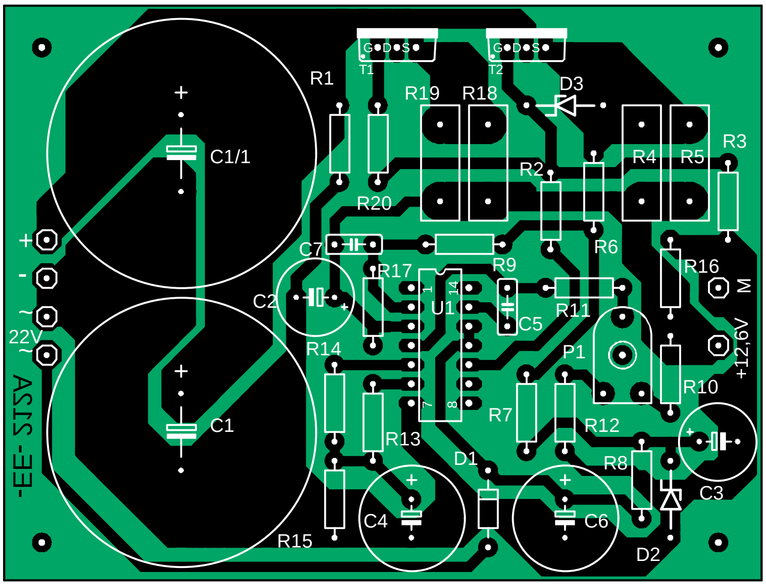
DC filament supply

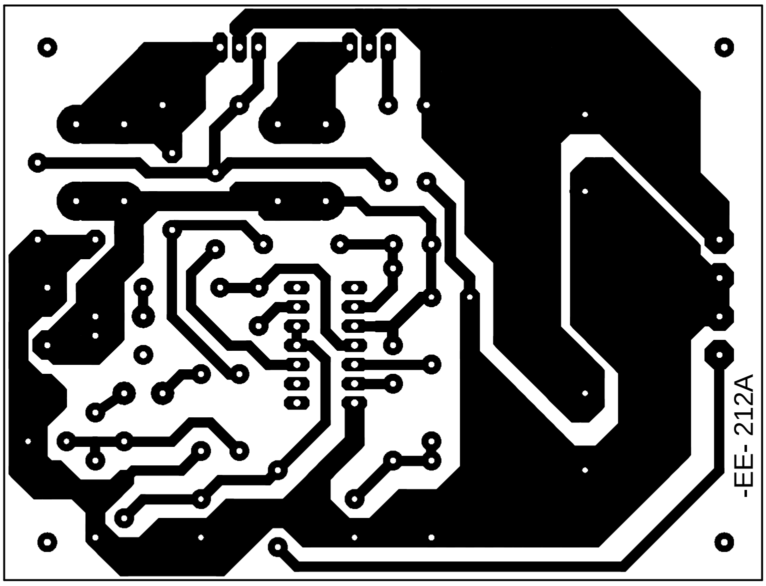
Resistors:
R1,R10,R15,R17 = 4k7
R2 = 100Ω
R3,R9,R11,R14,R16 = 1 k
R4,R5 = 0Ω22, 5 W metal film
R6 = 47 Ω
R7 = 2k2
R8 =470 Ω
R9 = 1 k
R12,R13 = 2k7
P1 = preset 1 k

Capacitors:
C1 = 10.000 μ, 25 V
C2 = 47 μ, 40 V
C3 = 220 μ, 40 V
C4 = 1000 μ, 25 V
C5 = 470 p ceramic

Semiconductors:
D1= 1N007
D2 = ZTK22
D3 = Zener diode 18 V, 1.3 W
Gl1 = rectifier KBU 8 B
IC1 = 723 (DIL)
T1 = BUZ12

Miscellaneous:
1 mains transformer type NTR-15





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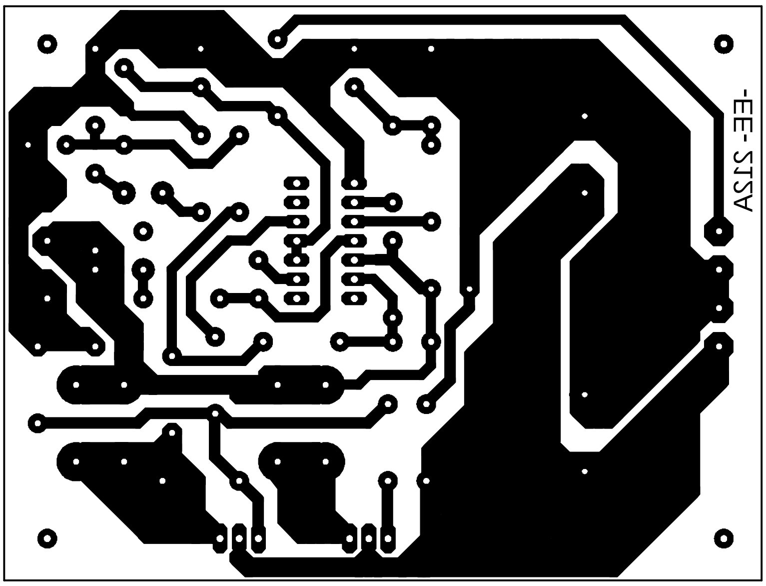

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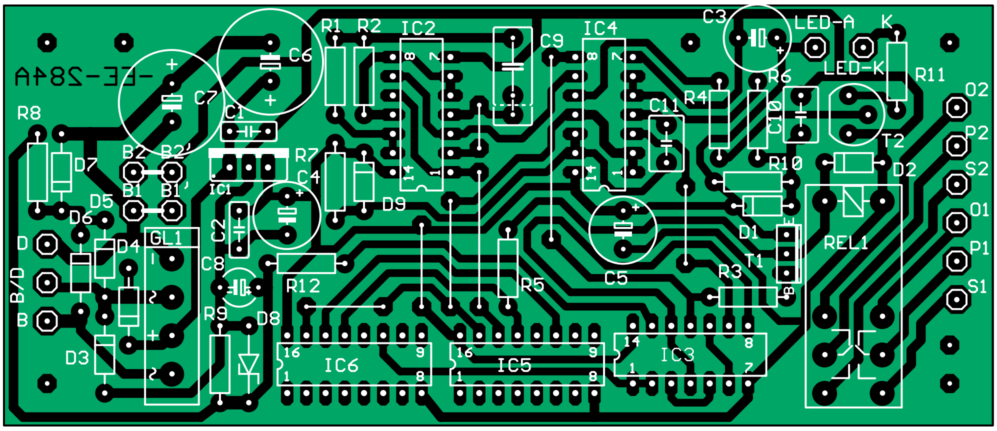
COMPONENTS LIST

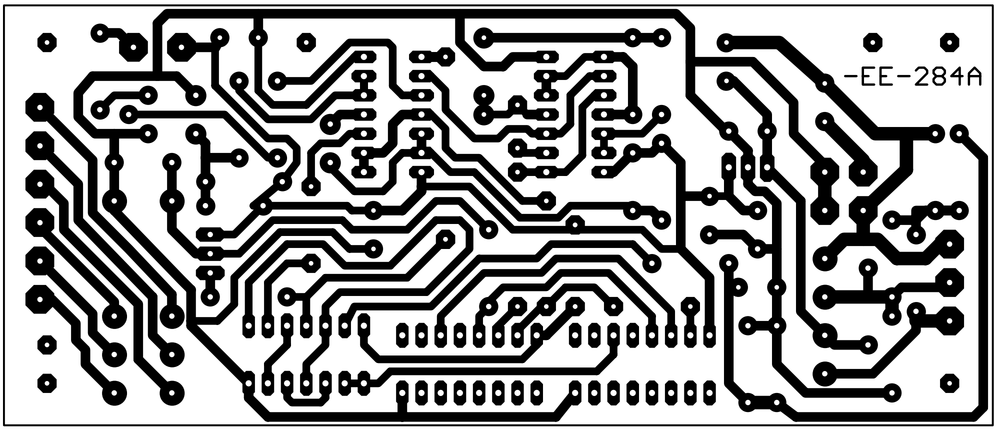
Power-on delay

Resistors:
R1,R5,R12 = 100 k
R2 = 680 k
R3,R4,R6,R7,R10 = 10 k
R8 = 2k2
R9=33k
R11 = 1k5 (see text)

Capacitors:
C1,C2 = 100 n ceramic
C3-C5 = 10 µ, 63 V
C6,C7 = 220 µ, 25 V (470 µ, 25 V at 6.3 V)
C8 = 1 µ,63 V
C9 = 680 n MKT
C10= 100 n MKT
C11 = 10 n MKT

Semiconductors:
D1,D2,D9 = 1N4148
D5,D6 = 1N4007 (only with 15-V supply
D3-D5,D7= 1N4007 (only with 6.3-V supply)
D8 = Zener diode 12 V, 1.3 W
T1 =BD140-16
T2 = BC549
Gl1 = B80C1500
IC1 = 7812 (TO220)
IC2-IC4 = 4011
IC5,IC6 = 4017





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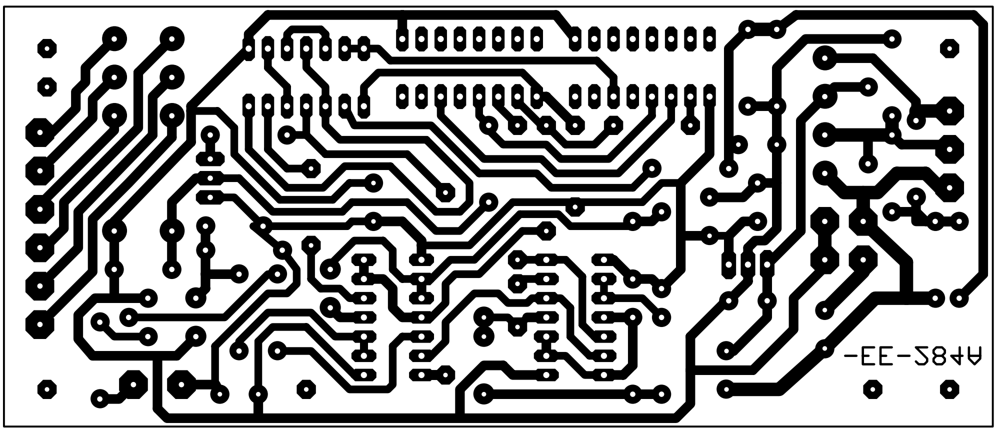

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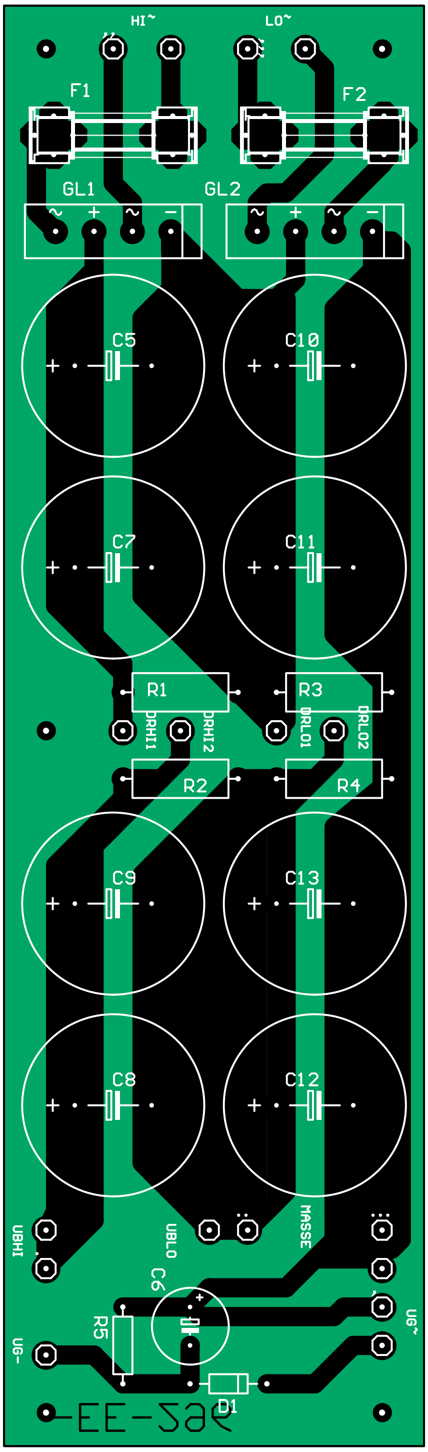

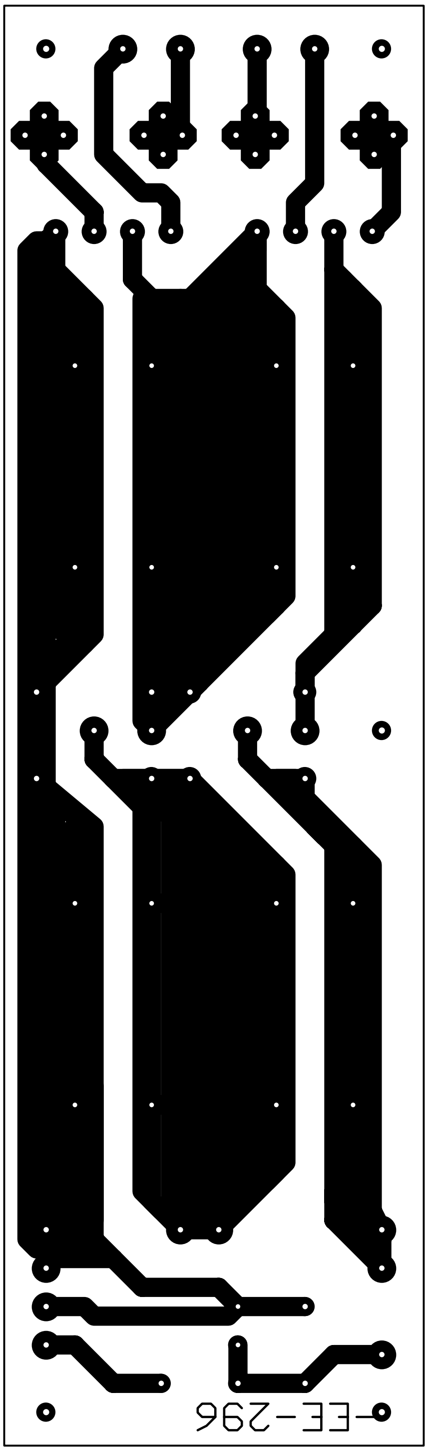

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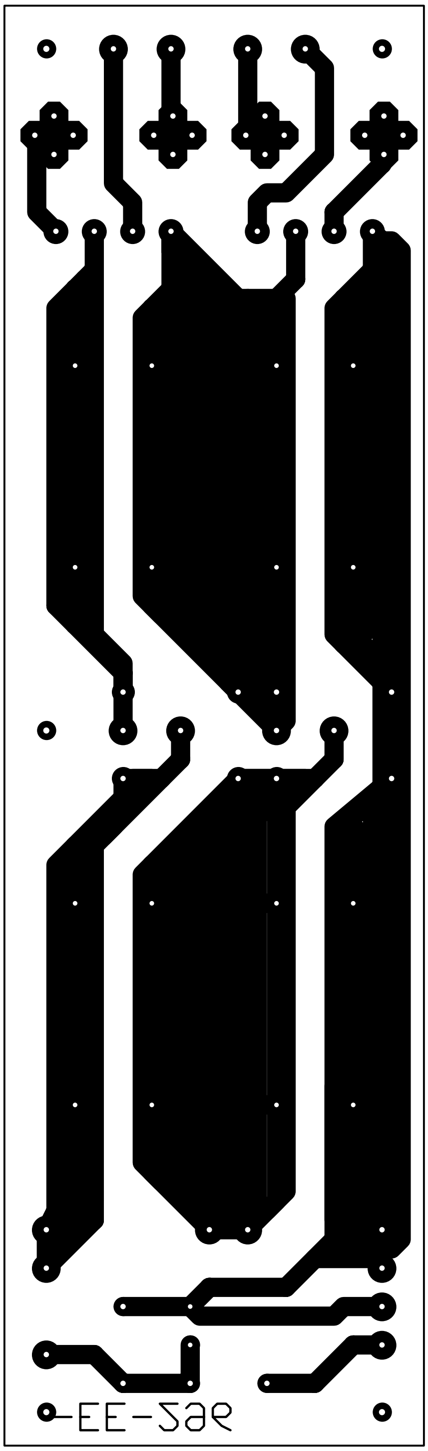

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COMPONENTS LIST

High-voltage PSU

Resistors:
R1-R4 = 150 k, 2 W
R5 = 47 k

Capacitors:
C1-C8 = 100 µ,500 V
C9 = 100 µ, 63 V

Semiconductors:
Gl1,Gl2 = B500C1500
in flat case
D1 = 1N4007

Miscellaneous:
Si,Si2 = 0.63 A slow, with
holder
Dr1 = 2H3, 0.3 A (D-2360)
Dr2 = 4 H, 0.18 A (D-4060)