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PICee DEVELOPMENT SYSTEM

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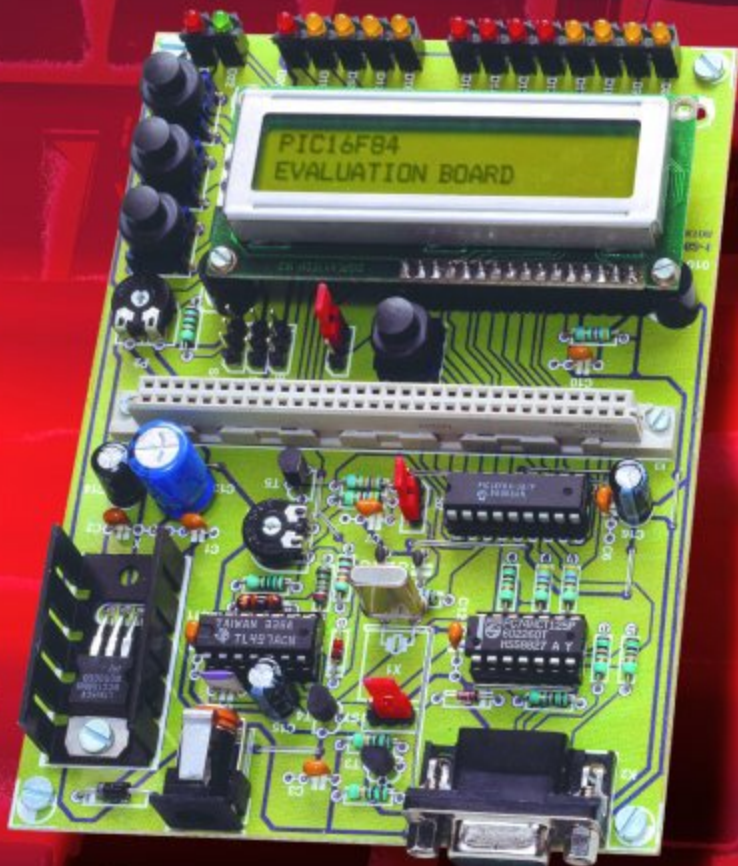
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Valve Amplifier Revival

return of the 'warm' sound

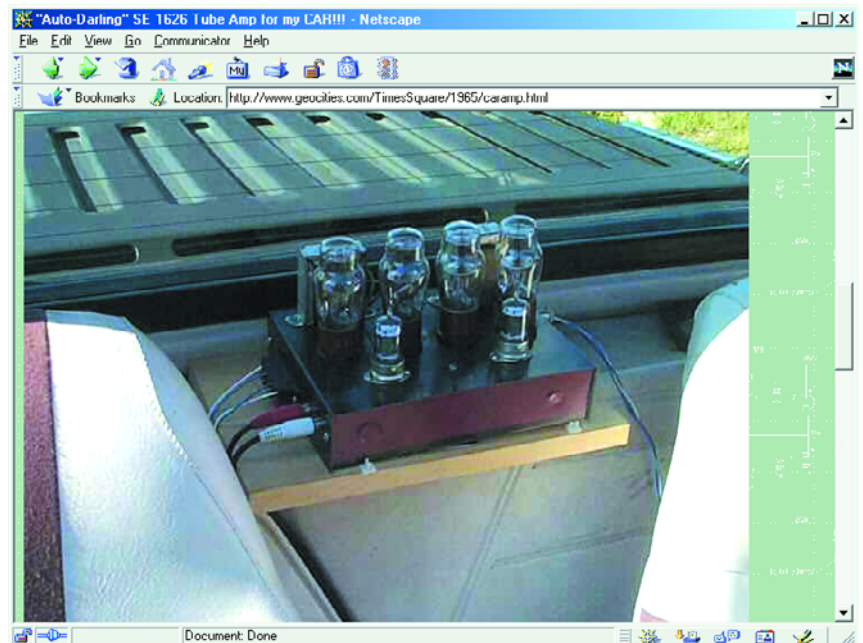
By Harry Baggen

Despite semiconductors being all around us, it seems that valves have never lost their appeal and even enjoy increasing interest from the 'all solid-state' generation. Among audio purists, valves have an excellent record for natural sound reproduction. The Internet is full of DIY valve amplifier designs, often complete with extensive description and photographs.

Despite the fact that semiconductors have been with us for more than 50 years, there is still a large number of audio lovers who prefer the warm sound of a valve amplifier over that produced by a transistor rig. Setting aside the air of nostalgia around valve amplifiers and their fine appearance, audio lovers will insist that valves are winners when it comes to sound reproduction with character. Despite the mechanical and electrical pitfalls encountered in the design and construction of valve amplifiers, like exotic transformers and dangerous supply voltages, thousands of enthusiasts enjoy tinkering with valves. Some do so with an innovative approach, while others focus on constructional aspects and strive to give their amplifier the best possible appearance.

On the Internet, a veritable galaxy of photographs may be found showing home built valve amplifiers. Once you start searching the net for websites covering valves, the amount of information is overwhelming. Besides true hobbyists you will also find companies professionally engaged in valve amplifier technology (still widely used in transmitters). Other firms specialize in kit, ready-built amplifiers or components. However these websites will not be discussed here since our aim was to see what was going on in the hobby department!

A great starting point for your explorations is the **World Tube Portal** [1], a website specialized in links to do with valves. Although these links have been sorted alphabetically, it is also possible to search in 105 categories.



Two of these links found under the heading 'DIY Tube Audio Sites', **DIY Tube Amplifiers** and **DIY Tube Audio Sites** [2] are of particular interest to hobbyists. Together these two links cover about 150 websites.

Once you start visiting a couple of the websites mentioned on the portal, you soon notice that there is much more to be discovered. We had to make a selection from the vast amount of information on offer, and have to limit ourselves to a couple of

sites we found either particularly attractively styled or rich in useful content.

To begin with, we should recommend **Audio Bizarro** from tube 'fanatic' Ralph Power. On his website Ralph shows a lot of photographs of valve amplifiers [3], which are a sight for sore eyes! Some amplifiers are complemented by a circuit diagram. However the feature *par excellence* of Ralph's website is his set of guidelines for



newcomers [4]. On 14 pages, Ralph tells you all you need to know to get started with valve amplifiers for home construction.

The hobby is international, too, as we soon discovered. **Bob Danielek** from the USA [5] designs and builds valve amplifiers large and small. Of the latter, the 'Darling', a super simple 1.5-watt stereo design, is fairly well known. Another, more unusual, design is a valve amplifier for use in a car — Bob has actually succeeded in fitting his sports car with a small, valved amplifier incorporating a power inverter for the high voltage supply.

The Japanese also have a soft spot for valves and the relevant websites we came across are often marked by original ideas. Of the many websites we came across we should mention **Explore the wonders of direct heating** [6] which describes the Sakuma principle.

The Canadian website run by **Rudy Godmaire** [7] allows you enjoy a visual impression of his beautiful designs of 70-watt power amplifiers. Rudy was obviously inspired by the designs of Menno van der Veen, an authority in the field and author of the book *Modern High End Valve Amplifiers* published by Elektor Electronics.

Claudio Bonavolta [8] from Switzerland is convinced that only

valves can bring across musical listening enjoyment. Besides a description of his own audio system (using a 300B single ended SRPP final amplifier), his 'Electronics' pages present a wealth of circuit diagrams employing transistors, valves and even hybrids. Here you will find nearly everything you need for a complete audio system, ranging from MC preamp to power amplifier.

Of German origin, **Roehrenfieber** [9] by Ulrich Romanski offers other valve amplifier constructors an opportunity to present their design on his website. Tuning tips are also discussed, and there is an interactive introduction into the basic operation of the valve.

The Dutch, too, have a reputation to keep up when it comes to 'all things valved'. **Aren's Attic** [10] not only presents a description of two amplifiers and a line amplifier, it also provides an excellent link list to glance through.

For those interested in the renowned Williamson amplifier design, the web pages put together by **Bert van der Kerk** [11] are worth visiting. There, the construction of a Williamson amplifier is described.

Triode Dick's page [12] is the third but by no means least Dutch website we should mention in this article. On his homepages, Dick reports on his own ventures in

building valve amplifiers. Of particular interest are his extensive descriptions and carefully finished photographs that go with the designs. There are also quite a few tips for music to play through your valved stereo — apparently Dick is a great fan of Mobility Fidelity Soundlab.

Finally, two websites with lots of schematics of valve amplifiers from the past — an excellent source of inspiration for new designs! The British website **The Circuit Archive** [13] holds an extensive archive of all designs by Dynaco and Heathkit, which were once famous for their kits. The schematics cover a lot of valve designs.

The company **Triode Electronics Online** [14] has a website with an extensive archive of antique schematics. Most amplifier brands are found in the archives. If not, a link is given to another collection of schematics on the net.

(025012-1)

Internet addresses

- [1] World Tube Audio Portal:
www.worldtubeaudio.com/
- [2] DIY Tube Amplifiers en DIY Tube Audio Sites:
www.worldtubeaudio.com/diy_tube_amplifiers.htm
www.worldtubeaudio.com/diy_tube_audio_sites.htm
- [3] Audio Bizarro Constructors Page:
www.spiritone.com/~tube/constructor.htm
- [4] Building your own tube amp:
www.spiritone.com/~tube/ralphbuild.htm
- [5] Bob's Music/Vacuum Tube Audio/electronics Page:
www.geocities.com/TimesSquare/1965/music_etc.html
- [6] Explore the wonders of direct heating:
www10.big.or.jp/~dh/index.html
- [7] Rudy Godmaire DIYer zone:
www3.sympatico.ca/r.godmaire/rg32e.html
- [8] Claudio Bonavolta:
www.infomaniak.ch/~bonavolt/audio.htm
- [9] Roehrenfieber:
www.roehrenfieber.de
- [10] Aren's Attic:
<http://members.tripod.lycos.nl/Aren/index.htm>
- [11] The Williamson Tube Amplifier:
www.spiritone.com/~tube/constructor.htm
- [12] Triode Dick's Page:
www.triodedick.com/
- [13] The Circuit Archive:
www.circuitarchive.co.uk/heath.htm
www.circuitarchive.co.uk/dynaco.htm
- [14] Triode Electronics Online:
www.triodeel.com/schindex.htm

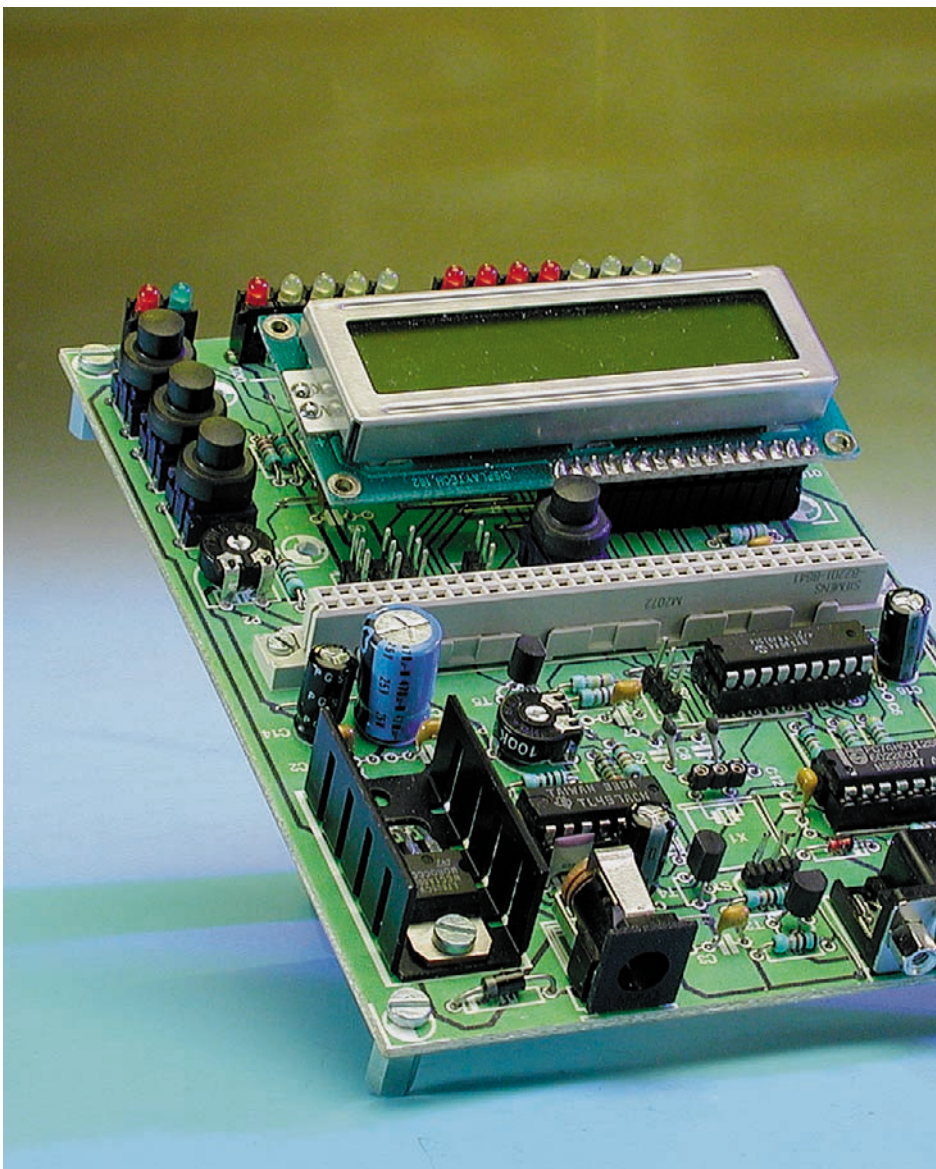
PICee Development System

a PIC16F84-based single-board computer

by Reinhardt Weber, DC5ZM

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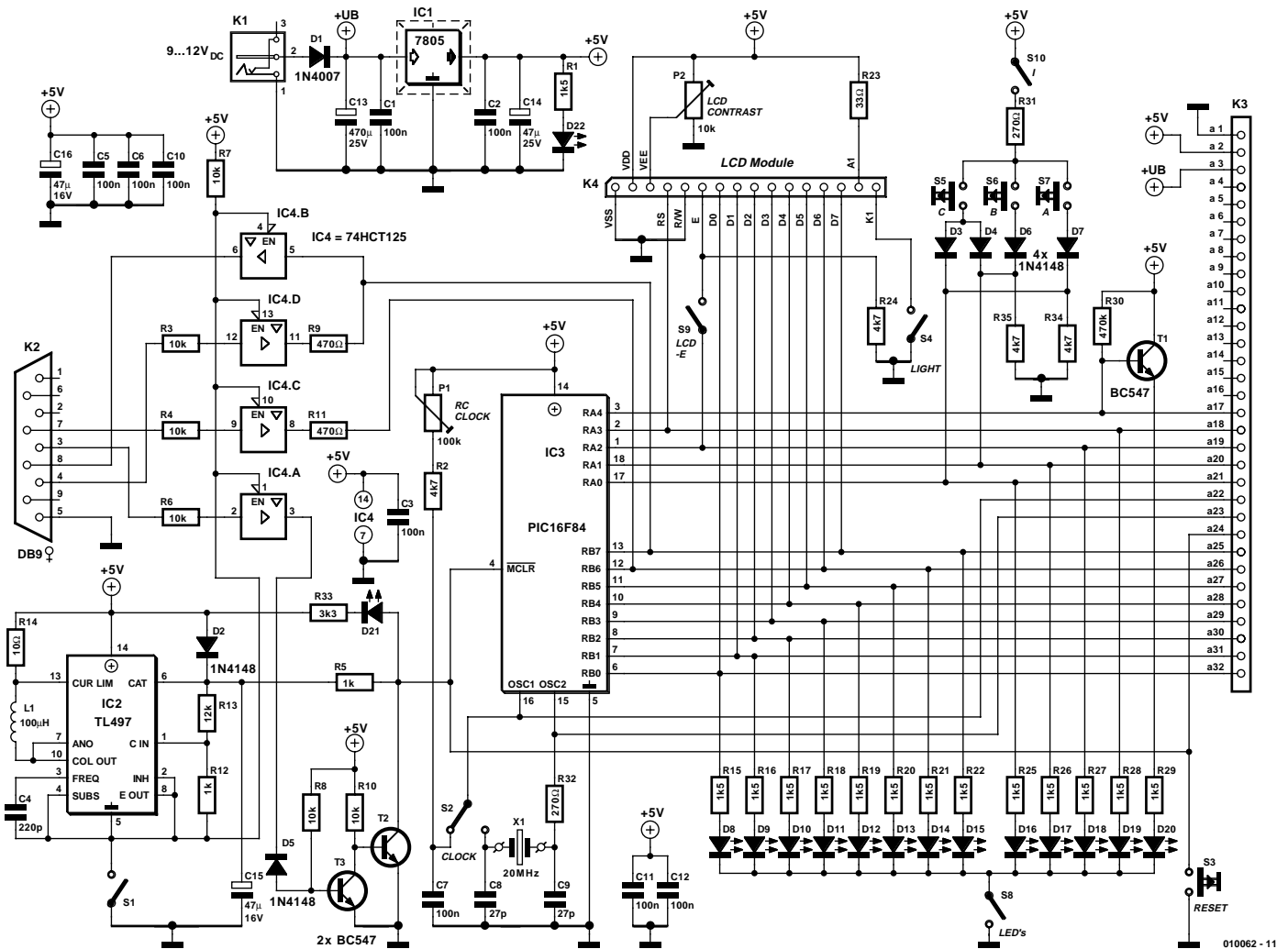
This single-board computer, using the popular low-cost PIC16F84 microcontroller, has been developed with educational applications in mind.



The PICee single-board computer described in this article is a versatile training and development system based on the well-known Microchip PIC16F84 microcontroller. The microcontroller's flash memory is electrically erasable: hence the 'ee' in the title. In contrast to the 89C8252 Flash microcontroller board described in our December 2001 issue, the 16F84 processor used here is a so-called 'RISC' (reduced instruction set computer) microcontroller with only a small number of instructions. The PICee board allows experimentation with all 35 of the processor's instructions without additional hardware. The board encompasses a wide range of applications, from a simple LED flasher to an elegant crystal-controlled clock.

The programming hardware

The socket can accept all varieties of the Microchip PIC16F84, with clock frequencies from 4 to 20 MHz. The clock can be generated using a quartz crystal or an RC oscillator. Switch S2 selects between the two oscillator types. Using the slower RC oscillator is particularly convenient during experimentation or development. The clock



010062 - 11

Figure 1. The microcontroller training board includes a wide range of peripherals for experimentation.

frequency is continuously variable via trimmer P1. Diode D2 and resistor R5 ensure that 5 V is fed to the MCLR input of the microcontroller in normal operation. Pushbutton S3 can be used to reset the processor.

If a DC voltage of around 13.5 V is applied to the MCLR input, the microcontroller switches into programming mode. The programming voltage is generated on the circuit board using a TL497 step-up converter (IC2) and enabled using switch S1, which takes pin 5 of IC2 to ground. LED D21 indicates when programming mode has been activated. The board is connected to a PC over the serial interface via 9-pin sub-D connector K2. Connection can be made using a normal RS232 cable (not a null modem cable) with a 9-pin D-type plug at the PICee end and the usual 9-pin D-type socket at

the PC end. **Connection should be made with the unit switched off** — only when the unit is connected should power be applied and the programmer software started up. The two programming signals DATA and CLOCK are taken via drivers IC4.C and IC4.D to microcontroller inputs RB6 and RB7. Driver IC4.B delivers data read from the microcontroller back to the interface. Driver stage IC4.A is used to control the programming voltage and reset the microcontroller.

A microcontroller can become the heart of a microcomputer system with the addition of peripheral components. The circuit board includes three typical applications that allow experimentation with the microcontroller's instruction set. These expansion circuits can also be built into other applications you may develop.

LEDs D8 to D20 indicates the logic values present on the RA and RB ports of the microcontroller. These are very helpful when debugging applications at low speed using the RC oscillator. They can also be used in your first programming exercises: for example, a LED flasher, running light, bar graph display, or LED dimmer. The port LEDs can be enabled or disabled via miniature switch S8.

Switches S5, S6 and S7 form a mini-keyboard. Pressing S6 or S7 produces a logic '1' at port inputs RA1 or RA0, while pressing S7 produces a logic '1' at both inputs, thanks to the wired-OR configuration around D3 and D4. It can be determined in software which button has been pressed. Three buttons can give rise to a wide range of control possibilities if one button is allocated to a 'mode' function to select a value to be changed, while the other two are used to adjust the selected value, up or down. Think of how alarm clocks or time switches are set, or how car radios are controlled.

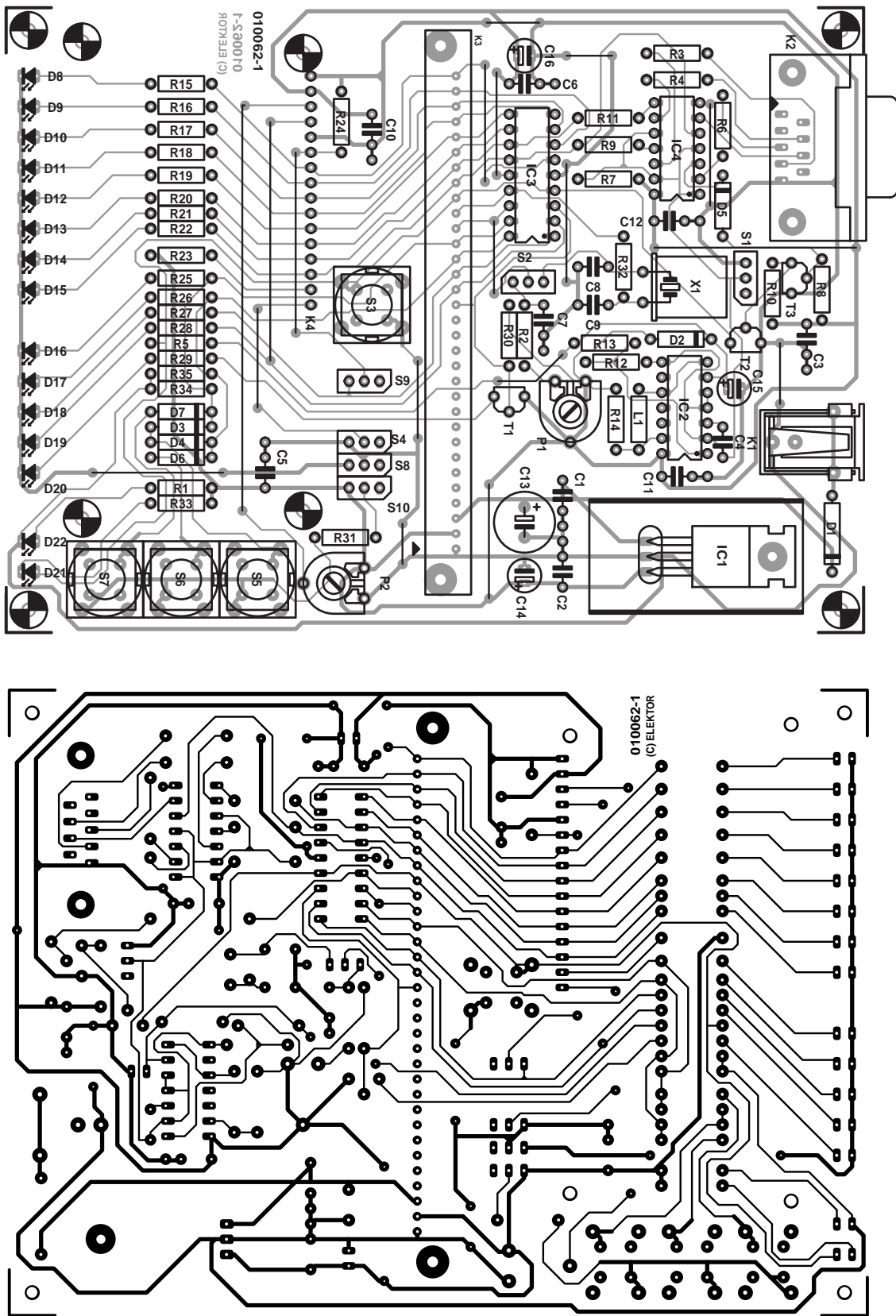


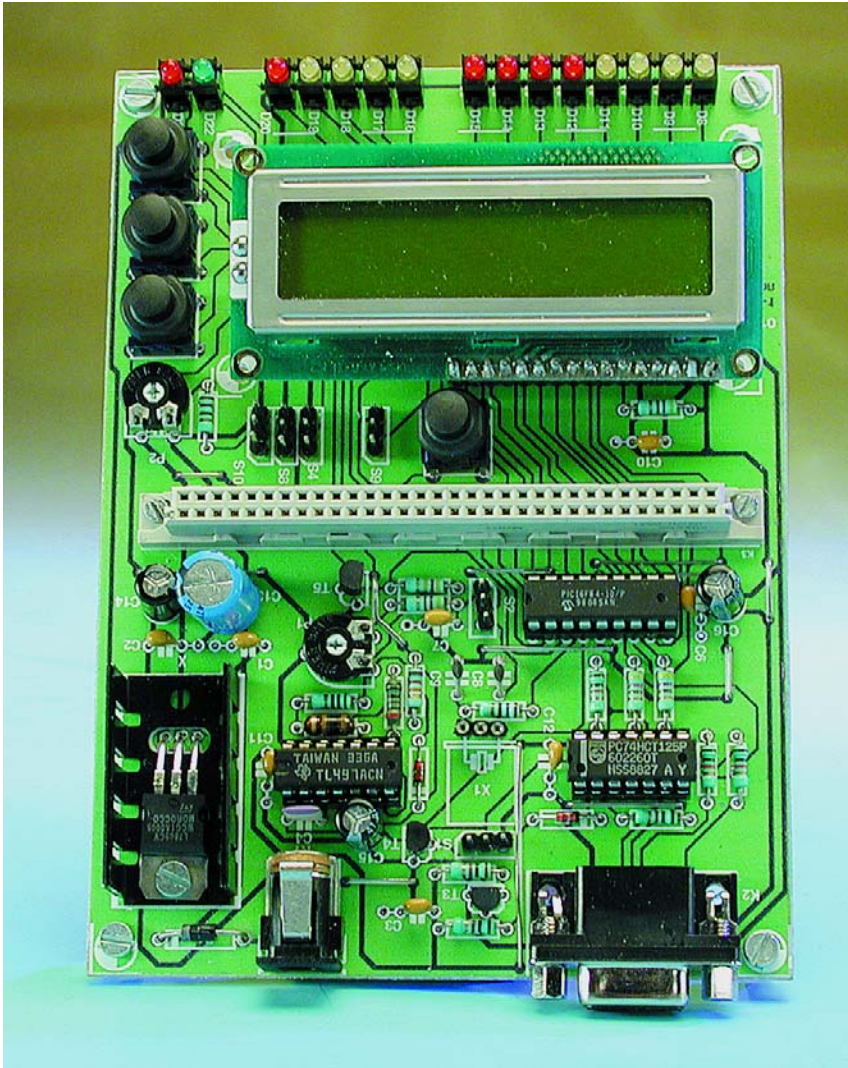
Figure 2. The layout of the single-sided printed circuit board for the PICee single-board computer.

When bright light falls on the LEDs, a voltage can be developed at the high-impedance inputs to the microcontroller that can lead to

false readings: for this reason R34 and R35 are fitted across monitor LEDs D16 and D17 to pull the input

signals down to ground, avoiding this unwanted effect.

A 2-line by 16-character alphanu-



meric LCD dot matrix module can be fitted to allow the system to display more complex information. This module is based around the Hitachi HD44780 controller, which has become something of an industry standard. The display is driven using the E (enable) and RS (register select) signals, which are connected to port pins RA2 and RA3. The read/write (R/W) input of the module is tied permanently to ground (GND), since it is only rarely that read mode is wanted. Trimmer P2 allows the display contrast to be set, and the backlight can be turned on and off via switch S4. The LCD controller can be enabled and disabled via switch S9.

All the microcontroller pins as well as the input power supply voltage and the regulated 5 V supply are available on a 32-way DIN41612-style female connector. Special application circuits can be constructed on low-cost prototyping board and con-

nected to the single-board computer via a complementary 32-way male connector.

Power for the single-board computer comes from an external mains supply connected via low-voltage connector K1. The input voltage can be anywhere between 9 V and 12 V. The built-in fixed voltage regulator IC1 produces a stabilised 5 V output. LED D22 indicates when power is applied. If the input reverse polarity protection diode D1 and the regulator are dispensed with, battery operation from four NiCd cells (4.8 V total) is possible.

Construction of the circuit on the printed circuit board shown in **Figure 2** should present no problems; sockets should be used for all ICs and for the crystal. The circuit board — surprisingly for a single-board computer — is only single-sided and therefore inexpensive. The price for this is 18 wire links in the well-spaced layout.

COMPONENTS LIST

Resistors:

R1,R15-R22,R25-R29 = 1k Ω 5
 R2,R24,R34,R35 = 4k Ω 7
 R3,R4,R6...R8,R10 = 10k Ω
 R5,R12 = 1k Ω
 R9,R11 = 470 Ω
 R13 = 12k Ω
 R14 = 10 Ω
 R23 = 33 Ω
 R30 = 470k Ω
 R31,R32 = 270 Ω
 R33 = 3k Ω 3
 P1 = 100k Ω preset
 P2 = 10k Ω preset

Capacitors:

C1,C2,C3,C5,C6,C7,C10,C11,C12 = 100nF
 C4 = 220pF
 C8,C9 = 27pF
 C13 = 470 μ F 25V radial
 C14,C15,C16 = 47 μ F 16V radial

Semiconductors:

D1 = IN4007
 D2-D7 = IN4148
 D8-D11,D16-D19 = LED, 3mm, yellow, low current
 D12-D18,D20,D21 = LED 3 mm, red, low current
 D22 = LED, 3 mm, green, low current
 T1,T2,T3 = BC547
 IC1 = 7805
 IC2 = TL497AC
 IC3 = PIC16F84
 IC4 = 74HCT125

Miscellaneous:

K1 = mains adaptor socket, PCB mount
 K2 = 9-way sub-D socket (female), angled pins, PCB mount
 K3 = DIN41612 connector, model B (Conrad Electronics # 741582)
 K4 = 16-way SIL connector*
 L1 = 100 μ H
 S1,S2,S4,S8,S9,S10 = toggle switch, 1 c/o contact, or 3-way pinheader with jumper
 S3,S5,S6,S7 = pushbutton, e.g., ITT/Schadow type D6
 X1 = quartz crystal socket with crystal*
 Heatsink for IC1, e.g. type ICK35 (Fischer) (Dau Components)
 PCB, order code **010062-I**
 Disk, contains example programs, order code **010062-II**

All 13 of the microcontroller's port pins as well as the power supply are brought to the connector in the middle of the circuit board, next to which, as can be seen from the main photograph, the LC display is fitted. The electrical connection to the display can be made

with a combination of SIL connectors or alternatively short wire links can be used. The switches shown in the circuit diagram can be replaced with jumpers.

Free programming software

A wide selection of literature is available on learning to program the PIC microcontroller family. A few starting points are given in the references, and the Internet is also a rich source of information, with numerous articles and items of hardware and software available. Also, almost every technical college will have some information on their homepage about the PIC microcontroller: just type 'PIC16F84' into your favourite search engine.

You will need some programming tools to develop assembler programs. The Windows program MPLAB.EXE (editor, assembler and simulator) and the DOS programs MPASM.EXE (assembler) and PSIM.EXE (simulator) produced by Microchip are recommended. These are freeware and can be freely downloaded from the Internet. Datasheets for the microcontroller and numerous example programs are also available.

One example of free programming software is **NTPicprog**, which can be found at <http://home.swipnet.se/~w24528/NTPicprog> (Figure 3). ICPROG.EXE, available from www.ic-prog.com also works well. When setting up the hardware, the PICee system appears as 'JDM Programmer' (Figure 4). The trustworthy **PIP02** software is also available for those who would rather work under DOS.

Once the source code for the program (*.asm) has been prepared and successfully converted to a *.hex file using the assembler, this can be downloaded into the flash program memory in the microcontroller.

A large number of example programs can be found on the project software disk ref. **010062-1**.

(010062-1)

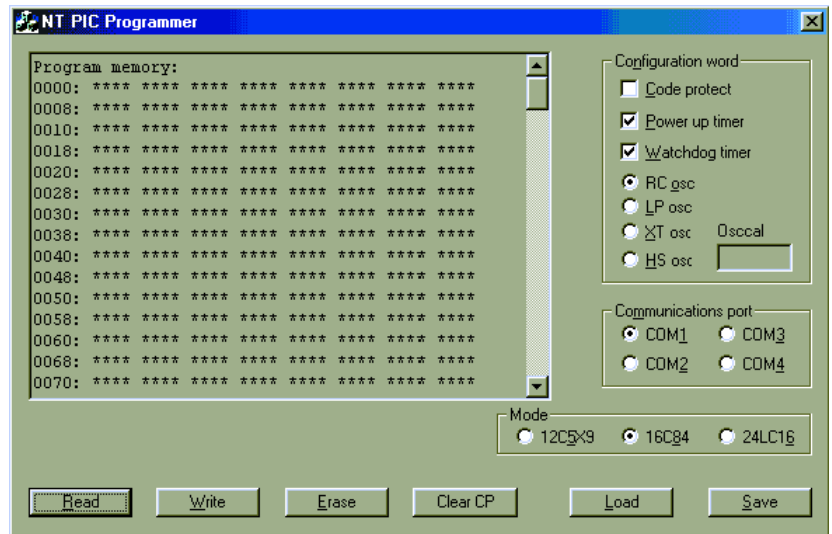


Figure 3. The NTPicprog programming software.

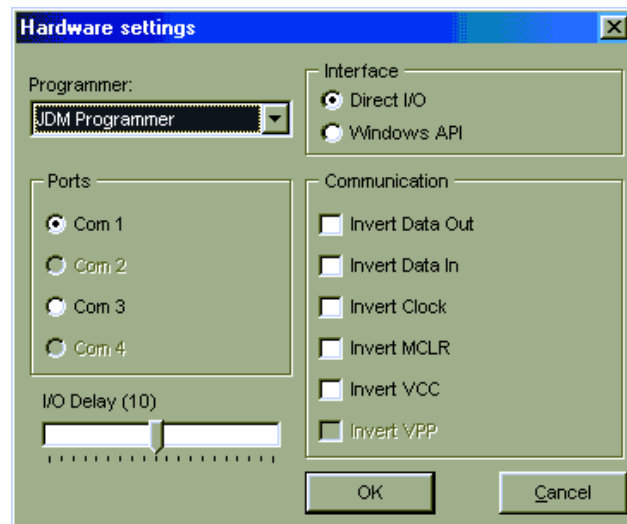


Figure 4. Hardware settings using ICPROG.

Relevant Internet sites:

<http://www.microchip.com>

<http://www.wolfgang-kynast.de/pic.htm>

<http://www.ludwig-geissler-schule.de/docs/picee/picee.html>

References:

David Benson
Easy PIC'n
Publisher: Square I

David Benson
PIC'n up the Pace
Publisher: Square I

F. Volpe
PICs in Practice
Publisher: Elektor Electronics (Publishing)

Back to school

This circuit was developed and tested at the Ludwig-Geissler school in Hanau, Germany with a particular view towards its use in teaching. The single-board computer has been used there very successfully for several years in various classes, both for training in the use of microcontrollers and in project work. For a component cost of only about thirty pounds, it also offers the radio amateur or electronics hobbyist an ideal platform for experimenting with and developing ideas.

Microcontroller Basics Course (2)

part 3: port characteristics and port accesses

By B. Kainka

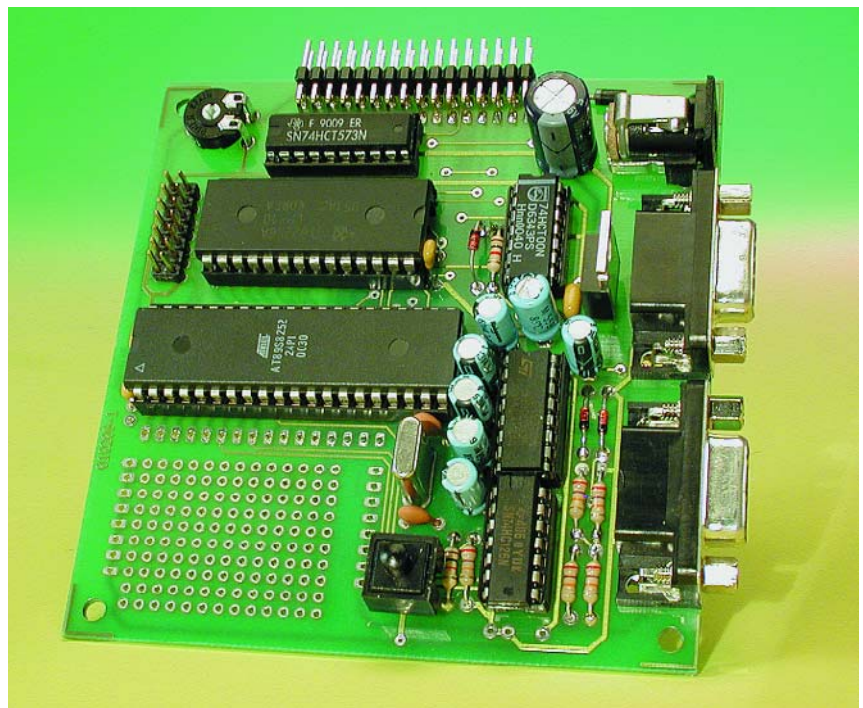
In the first instalment of this basics course, we introduced the assembler that we will be using. Now we come to the first practical applications using the processor ports. First, though, we have to take a look at the physical characteristics of the port connections.

Anyone who already has a bit of experience with digital electronics knows that there are many different types of outputs. Each type has quite specific characteristics, regardless of whether it is a TTL, CMOS, tri-state, open-collector or open-drain output, and if you want to connect something to the output you need to be aware of these characteristics.

In any case, the Port 1 outputs of an 8051 microcontroller do not fall in line with any of the known logic families, but instead employ a rather unique solution. These ports are what is known as 'quasi-bidirectional', which means that they can be used as inputs or outputs without having to be specially switched over. You should keep in mind that the ports of a microcontroller represent a sort of door to the outside world. Depending on the task to be performed, inputs or outputs are needed. Some microcontrollers use tri-state buffers that must be switched to the high-impedance state to allow them to be used as inputs. This naturally requires a special switching signal or special instructions to switch the data direction. This is not necessary with an 8051, since all ports can be used as both inputs and outputs without any switching.

The port in detail

A glance at the detailed circuit diagram of a port (**Figure 1**) shows how a quasi-bidirectional port is built. There is a single FET with a pull-up resistor located at the output. In the High state, the FET is cut off and the pull-up



resistor alone defines the internal resistance of the port. Consequently, it is certainly possible to connect any desired logic output here or change the signal level by means of a switch connected to ground. Even a logic input with high input impedance, such as that of a CMOS IC, will not have any difficulty recognising a High state.

The situation is quite different when the output is conducting and thus forces the signal level to be Low. In this case, the port impedance is relatively low. Anyone who attempts to force the output level of the port to a High state when it is in this state can only have bad intentions, since he or she is trying to force the microcontroller into the dig-

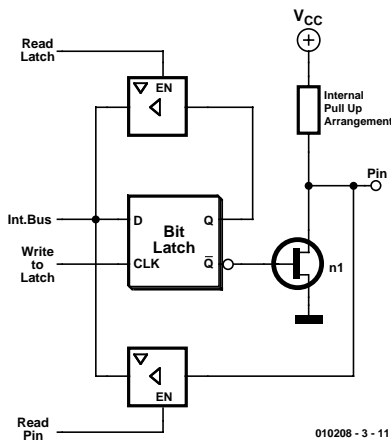


Figure 1. Internal structure of a port.

ital 'happy hunting ground'. We thus have the following rule: outputs must never be connected to other outputs. You must always remember that if you want to use a port pin as an input, it must be in the High state! This is always the case following a reset.

The internal circuit diagram of the port pin contains yet another simplification, since the pull-up resistor is in fact also a FET. Consequently, it acts like a constant-current source. This naturally raises the question of how large this constant current is. A measurement on the AT89S8252 shows that a short-circuit current of around 19 μA flows when the port is pulled to ground. This is not very much. It is far less than what is needed to drive an LED, for example. Is it actually enough to change the signal level on a lead reasonably quickly? Actually, it isn't, since every lead has an intrinsic capacitance that must first be charged or discharged. Here again there is a refinement in the internal circuit of the port. Strictly

speaking, the pull-up resistor consists of two (in fact, three) FETs. One of them provides the normal, very weak pull-up current. A second one provides a much stronger current, but it is active only for a short interval when the port is switched from the Low state to the High state (or when an address must be output). The port pin thus changes state very quickly, even with a certain amount of capacitive loading. However, it can still be used as an input and can be actively pulled to ground, since any possible short circuit lasts less than a microsecond.

A quasi-bidirectional port can also directly drive an LED, but only if series resistor for the LED is connected to V_{CC} rather than ground. **Figure 2** shows how an LED and a switch can be connected. For the first program of the previous instalment of our course, the LED must be connected to one of pins P1.4 through P1.7 for it to be illuminated, since only these pins are switched Low. In this case, the switch should be connected to one of pins P1.0 through P1.3, since they are in the High state and can thus act as inputs. By the way, here you can operate the switch as often as you like, but nothing will happen, since we must first write a program that polls the input and evaluates the result.

Our first program loop

After this introduction, it's time for something practical. What we want to do is to automatically switch the output levels on the port pins. To do so, we will modify the program from the first instalment of the course to obtain the program shown in **Listing 1**. This program first outputs the

Listing 1. A program loop with port outputs.

```
;flash2.asm, fast loop

#include 8051.H
        .org 0000H

main    mov    a,#0Fh      ;1 a = 15
        mov    P1,a       ;1 P1 = a
        mov    a,#0F0h   ;1 a = 240
        mov    P1,a      ;1 P1 = a
        sjmp   main      ;2
        .end
```

port value 0Fh and then outputs the port value F0h. When hexadecimal notation is used, the first character must always be a numeral. This is why the listing shows a '0' in front of the second value, which is thus '0F0h' instead of 'F0h'.

Another feature of this listing is that the register p1 is no longer defined in the text. Instead, we have added an 'include file' (8051.h) that contains all important definitions, including much more than just Port 1. Besides this, the starting address is explicitly specified using the .org statement. The microcontroller always starts its programs at address 0000h following a reset. Finally, the critical change is that the loop has been expanded. It now encompasses the entire program, which is executed over and over again.

This small program helps answer a quite important question, which is how fast such a microcontroller can actually run programs. All we have to do is to touch the input probe of an oscilloscope to one of the port pins. Here we will see a rectangular waveform with a period of around 150 kHz. This can also be demonstrated using a radio. A short piece of wire attached to the port pin can serve as an antenna. In the long-wave band, you will find the signal at around 150 kHz. The fifth harmonic can be received at roughly 750 kHz in the medium-wave band. This gives us more insight into the fundamental significance of the EMC directives. Whenever high frequencies and steep edges occur, special provisions must be taken to prevent the circuit in question from acting as a transmitter.

The number of instruction cycles is shown in the listing comments. An instruction cycle takes 12 oscillator clock cycles, which means it has a period of $(12 \div 11.059 \text{ MHz}) = 1.095 \mu\text{s}$. Most instructions have a duration of one instruction cycle, while the jump instruction takes two cycles. In total, the sum of the instruction times is six instruction cycles. This means that the loop takes $6.51 \mu\text{s}$, which gives a frequency of 153 kHz.

There is yet another interesting observa-

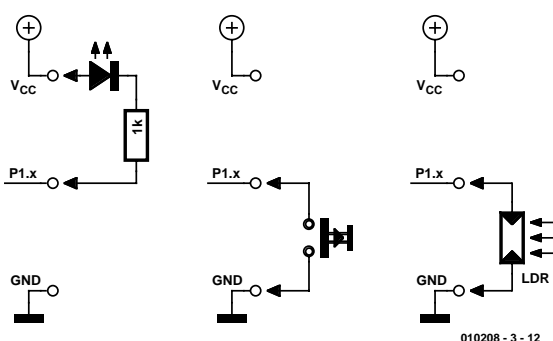


Figure 2. Port connections for inputs and outputs.

tion that we can make with this program. We can use it to watch the two pull-up FETs working 'live'. To do so, we connect a 33-kΩ resistor from one of the port pins to ground. With this resistance value, the smaller FET is no longer able to pull the signal level to V_{CC} , but the larger FET can still do so. In the oscillogram, we can now see how long (or better, how short) the current level is increased. The initial edge is very fast. This is followed by a high plateau with a duration of around 100 ns (one quarter of a clock period), and finally the voltage drops to a low level.

The port evidently has yet another characteristic that is not mentioned in most data sheets. The smaller FET is also divided into two and behaves differently when the port state is High than when it is Low. With a load resistance of 6.8 kΩ, we see a second kink in the curve at a voltage of roughly 1.5 V. Evidently, the current that flows in the region above this input voltage is greater than the current that flows below this level. Altogether, this acts like a form of current feedback and results in a certain amount of input hysteresis. It can be easily measured using a multimeter. At a voltage greater than 1.5 V, the port supplies up to 200 μA, but below this voltage the current is only 10 μA. Thanks to this port behaviour, connecting a simple resistance to an input always results in an unambiguous input state. It is even possible to connect a potentiometer or an LDR, which will then be read with well-defined hysteresis. In the original Intel and Philips data books, the division into three FETs can still be seen, but the data sheets for the more recent 8051 derivatives from Atmel neglect this detail.

Figure 4 shows a small circuit for our first experiment with input connections for a port. Here P1.0 can be set to zero either by means of a switch or by a shining a sufficiently bright light on the LDR. A matching program must read the port state and switch on the LED attached to port 1.1 only when a High signal is found at the input. Here we must take into account the fact that the LED is con-

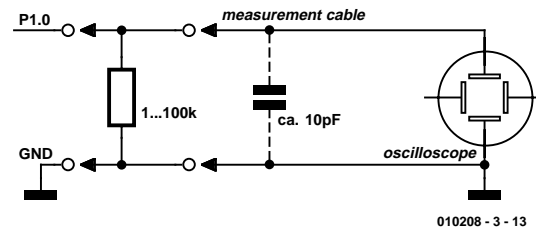


Figure 3. Measuring the port signal under load.

nected to V_{CC} , which means that it has an inverted function; it is 'on' when P1.1 is in the Low state. S2 allows us to provide feedback between the output and input of our test setup. What happens if we actuate S1 and S2 together and thereby short the P1.1 output to ground? Nothing serious, since a connection to ground is always allowed for a quasi-bidirectional port. The only thing is that in this case the LED is always on.

A conditional jump

The assembler program for **Listing 2** reads the port state at P1.0 and then executes a conditional jump. The instruction *jb* (jump if bit set) belongs to the group of special instructions for single-bit processing. Although most instructions work with byte values, these instructions evaluate, set or clear individual bits. Here 'P1.0' refers to a single port pin, while 'P1' in the first example refers to the complete port with all eight of its leads. The header file (8051.h) defines all necessary bit addresses, such as P1.0 and P1.1. You can view this ASCII file using a simple text editor.

In the program, if a '1' state is read at P1.0 in the first line, a jump to the destination 'ON' is executed. At this location, port P1.1 is placed into the Low state by means of *clr* P1.1 ('clr' = 'clear'), which switches on the LED. On the other hand, if a Low state is read at P1.0 in the first line of the program, the jump is not executed. Instead, the program quite properly executes the next following instruction. Here we find *setb* P1.1 ('setb' = 'set bit'). This is an instruction to enable port P1.1, which effectively means that the LED is switched off. In order to prevent this from being counteracted by

the lines with the label 'ON', it is necessary to jump over this part of the program. An additional jump instruction thus leads to the destination 'OFF', where the loop is closed by jumping back to the very beginning of the program.

This small example shows several interesting results. For one thing, we have practically built a logical inverter. A Low state on the input results in a High state on the output, and vice versa. Since the LED (Low = on) and the input switch (on = Low) have inverting functions, the overall operation is inverting. This means that if you press S1, the LED goes off, while if you release the switch the LED goes on again.

This experiment will show the input hysteresis of the port if a resis-

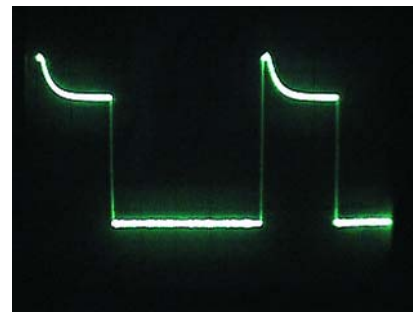


Figure 4. Port loaded with 33 kΩ.

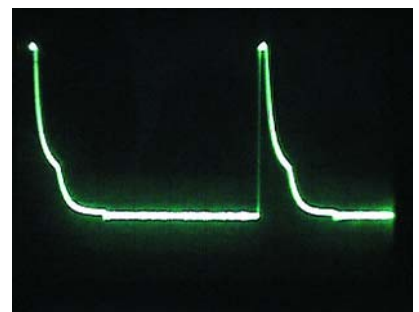


Figure 5. Port loaded with 6.8 kΩ.

Listing 2. Responding to an input state

```

;flash3.asm, input/output

#include 8051.H
        .org 0000H

main    jb     P1.0,ON  ;P1.0 = ?
        setb  P1.1     ;P1.1 = 1
        sjmp  OFF
ON      clr   P1.1     ;P1.1 = 0
OFF     sjmp  main
        .end
    
```

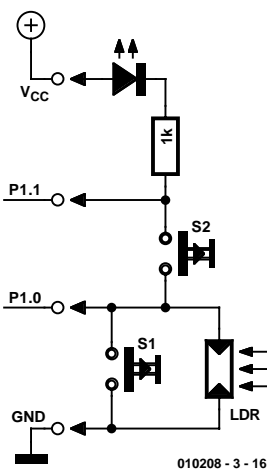



Figure 6. A circuit with an input and an output.

tor potentiometer or LDR is connected to the input. There is a definite gap between the switch-on and switch-off brightness of the light falling on the LDR.

If S2 is closed, feedback gives rise to a rapidly alternating succession of signal levels. The only visible effect of this is that the brightness of the LED is reduced, but an oscilloscope reveals the true situation and shows a sequence of fast rectangular signals. Naturally, this is exactly what should be expected, since the input state is only polled once at the beginning of the program. With feedback, the microcontroller will always find the output state resulting from the previous pass through the loop at this location, and that is exactly the opposite of the previous input state. The microcontroller thus has no other choice than to repeatedly switch the port state back and forth.

If we attempt something like this using a logic gate, in this case an inverter, the result can look much different. Usually the signal assumes an average voltage level, by means of which the gate reveals its analogue roots, since it actually an inverting dc amplifier. A sequential logic circuit (such as a microcontroller), by contrast, allows no possibility of analogue behaviour. Unambiguous yes or no decisions are always taken.

Counting loops

The final program for this instalment uses only outputs. In this case, we want to have all eight pins of port 1 output symmetrical square waves at different frequencies. Our model for this (in digital electronics) is an 8-stage binary counter. A clock signal is applied to the input, and its frequency is divided by exactly 2 for each stage of the counter. The microcontroller generates the clock signal itself using its program. The divider chain can be obtained very simply by incrementing a binary number. This can be done either by using an addition instruction or an inc (increment) instruction. The instruction inc a increases the value stored in the accumulator by 1 each time. The program must execute this instruction repeatedly in a loop and repeatedly output the value of the accumulator to the port.

The program shown in Listing 3 first loads an initial value of zero into the accumulator. Following this, the accumulator value is output to the port, the value in the accumulator is incremented, a jump is made to next for the next port output and so

Listing 3. A counting loop.

```

;flash4.asm port outputs
#include 8051.H
        .org 0000H

main    mov     a,#00
next    mov     P1,a      ;1
        mov     r1,#255  ;1
loop    djnz   r3,loop   ;2 * 255
        inc     a        ;1
        sjmp   next     ;2
        .end
    
```

on. The program also contains a second counting loop to cause everything to run a bit slower. The objective is to allow at least the lowest-frequency signal on port P1.7 to be directly observed using an LED.

The counting loop uses a register (r1). There are eight registers in total (r0 through r7). It does not matter here exactly which register is used. The register is loaded with a value of 255, which means that in this case a decimal number is used. We could have just as easily written this as 0FFh. The actual counting loop employs the complex assembler instruction djnz (decrement and jump if not zero). During the first pass through the loop, the value in r1 is reduced to 254. Since this is greater than zero, a jump is made back to loop, where the same instruction is again executed. During this pass, the value is reduced to 253. After a total of 255 passes through the loop, the value of zero is reached. Now the jump is no longer executed, but instead, the program continues with the next following line. In this manner a total of 255 loop cycles, amounting to 510 instruction cycles or 558 μs, are consumed. If we add the remaining instructions, we have a total period of approximately 564 μs, or a frequency of 1.77 kHz. This is thus the clock frequency at the input to the counter chain. Consequently, on P1.0 we find a square-wave signal with a frequency of 885 kHz, on P1.1 443 kHz and so on, down to around 7 Hz on P1.7. This frequency can be observed using an LED. It is worthwhile to perform a number of additional experiments using different loop parameters. The highest frequency results when the starting value is 1. By the way, the greatest reduction in the clock rate can be achieved with a value of 0, rather than 255, since when 0 is decremented it rolls over to a value of 255, resulting in 256 passes through the loop.

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This completes our introduction to assembler. The next instalment will start with BASIC-52.

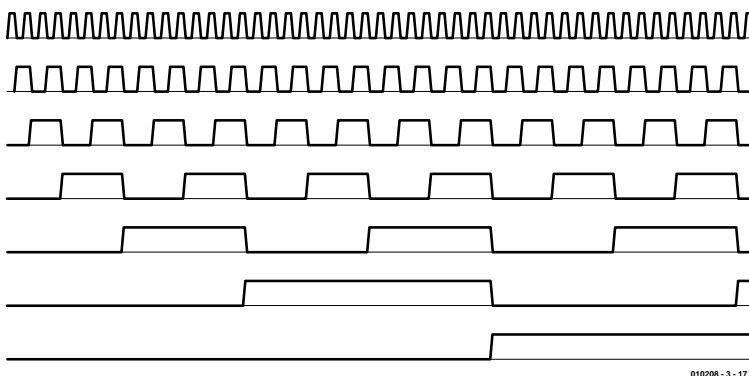


Figure 7. Output signals on P1.0 through P1.7.

Level Shifting

for digital signals

By K. Walraven

A surprisingly high number of questions from readers that arrive on our desks deal with the subject of the interconnection of two separate (sub) circuits that operate from different power supply voltages, or have dissimilar switching levels. How do you deal with that?

There was a time once, when all digital circuits always operated from 5 V. Give or take the odd exception, there were never any interface related problems and everything could simply be tied together. Those were the days!

Times change. It is now a frequent occurrence that an interconnection is required between systems operating from entirely different power supply voltages. A DCF time standard receiver module, for example, runs from a single 1.5-V penlight battery and has to be connected to a microprocessor system that runs from 5 V. Or consider a 12-V relay that needs to be connected to the same microprocessor system. The reverse may also occur: a sensor circuit that runs from 15 V has

to be connected to a system that operates from 5 V. In all these cases, and many more, some kind of level shifting will be required.

Open-collector

When the gap has to be bridged between different signal levels, it is by far the most convenient to use components with a so-called 'open-collector output'. These components have at their output a transistor (or FET), which has the emitter (source) connected to the common connection (0 V or ground). The collector or drain is connected via an external

resistor to the positive power supply. The above-mentioned DCF module also possesses such an open-collector output, as is shown in **Figure 1a**.

If you wish to connect such a component to a circuit that operates from a different power supply voltage, then it suffices to connect the collector resistor to the positive power supply rail of the **other** circuit and to simply join to common points. Figure 1b illustrates what is meant here. In this way, the amplitude of the output signal from the open-collector-stage is always equal to the power supply voltage of the second circuit — this is exactly what we need!

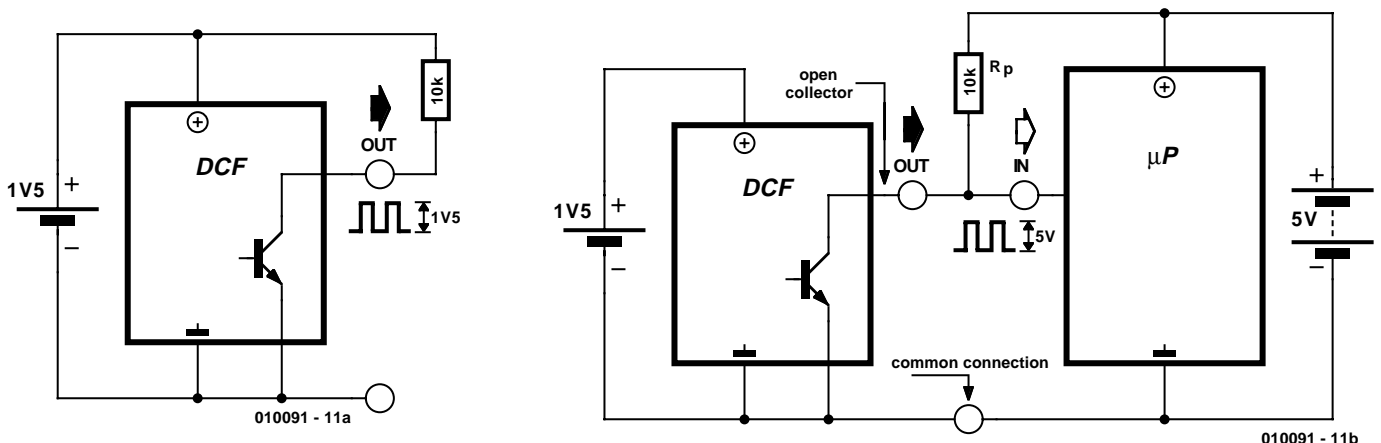


Figure 1. An open-collector-output makes it possible to shift the signal level by connecting the collector resistor to the power supply of the second circuit.

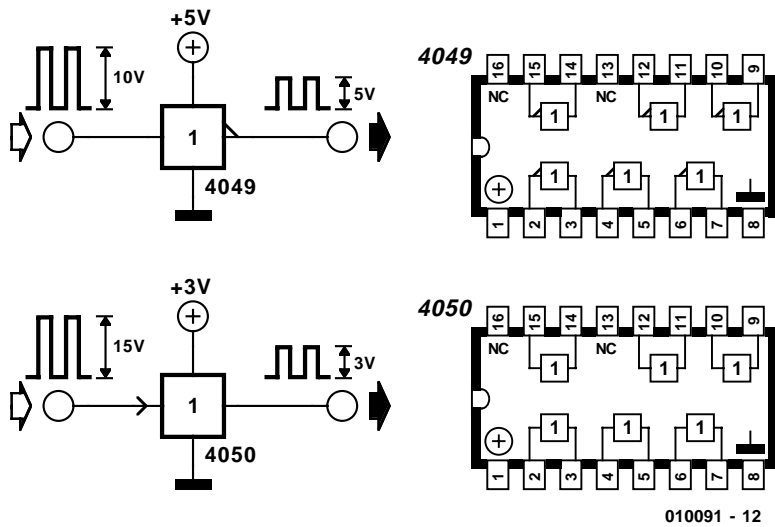


Figure 2. CMOS-buffers such as the 4049 en 4050 are ideal to convert a high voltage to a lower one.

High/low buffer

In many cases where a conversion from a higher to a lower voltage is required, another very simple solution is available. The CMOS-buffers 4049 and 4050 (inverting and non-inverting respectively) are specifically designed for this purpose. They tolerate input voltages of up to a maximum of 18 V, even if their power supply is only 3 V or 5 V. This means that they automatically convert the input voltage to (approximately) the level of their own power supply rail, as is shown in **Figure 2**.

Extra transistor

How to solve the problem of matching a lower voltage to a higher one when an open-collector output is not available? In this case we can easily create an open collector by simply adding a discrete transistor to the output! However, a practical problem rears its head: selecting the value of the base resistor. Remember that a transistor will start to conduct at about 0.7 V, and when converting from, for example, 5 V to 15 V this is much too early. It is much better to position the switching threshold at about a third, or even half, of the power supply voltage. To achieve this, we first make a potential divider from two resistors and connect the transistor 'behind' it (**Figure 3**). A rule of thumb is to select the values

of the resistors such that a current of about 0.5 mA flows through the potential divider. For example, resistor R1 becomes 1k5, while R2 gets the value of 1k8, for a threshold of one third of 5 V, or 1.67 V.

Instead of using a discrete transistor, it is also possible to use an IC from the ULN280x series. These are multiple darlington transistors in an IC package that is specifically intended to realise this kind of inter-

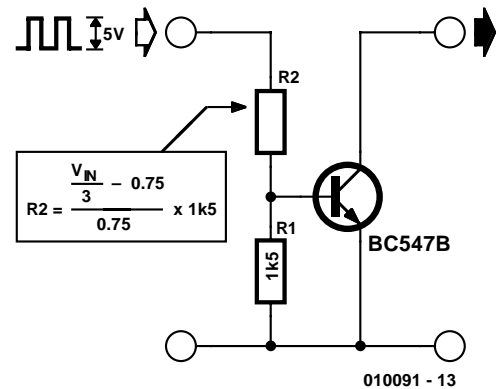


Figure 3. When an open-collector-output is lacking, a discrete output transistor can be added.

face. These ICs are dirt cheap and perfectly capable of switching loads of up to 500 mA. The ULN2803 is suitable for input voltages of 5 V, while the ULN2804 may be used for applications presenting input voltages between 6 V and 15 V. Two examples are shown in **Figure 4**. The resistors drawn are also contained in the IC.

An elegant as well as simple alternative is to not use a transistor at the output of the component under consideration. Instead, we add a FET as shown in **Figure 5**. The popular

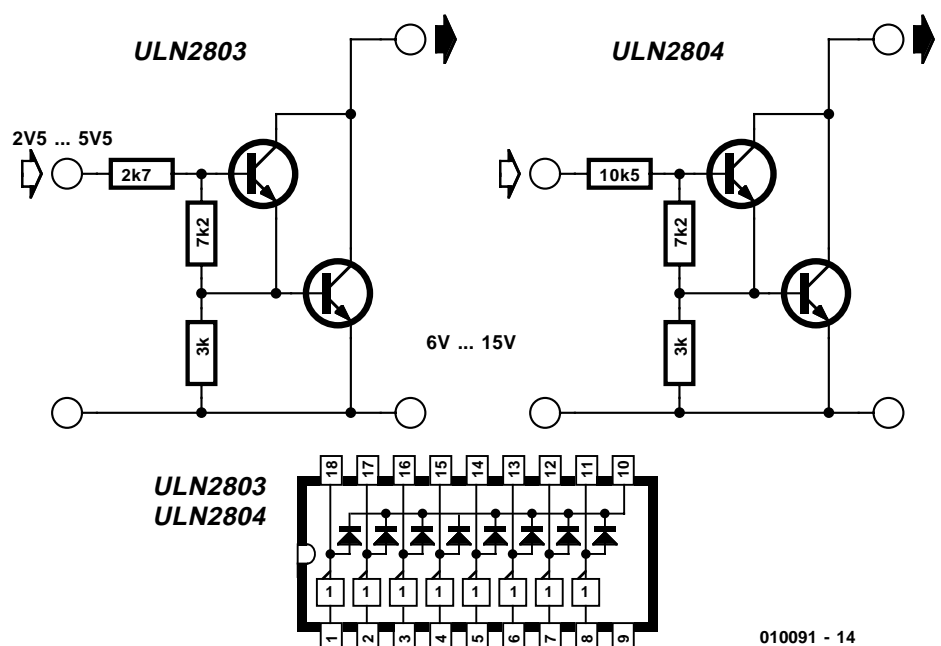


Figure 4. Darlington driver ICs type ULN2803 and 2804 are eminently suitable as open-collector outputs and can even switch up to 500 mA.

type BS170, as an example, switches on at about 2.5 V. An added bonus is that a gate resistor is not required, provided the gate voltage does not exceed 15 V.

No matter if you add a transistor, Darlington or FET, you have to keep in mind that all these components cause the logic level to be inverted.

Watch the current!

In most cases, the addition of the 'external open-collector transistor' will perform admirably. It will be clear, however, that a minimum of 0.5 mA is required to drive the transistor. While this is a very modest current, there are nevertheless outputs where this may be too much. For example, difficulties may arise with quasi-bidirectional outputs in microprocessors such those from the 8051 series, as well as some I²C ICs such as the well known PCF8574. These outputs can often sink several mA to ground but can only source 50 to 100 µA at the most.

All too often, these outputs are abused by driving an LED directly, for example, with the argument: it works, so why bother? However, this is definitely not the way it should be done and could easily lead to internal problems for the driving IC.

To avoid these risks it is best to use the FET solution of Figure 5. The threshold level is automatically correct and the input current to the BS170 is practically zero. Even the Darlington drivers inside the ULN2803 and ULN2804 draw too much current for these kinds of outputs. If you would still like to use one of these, because of their switching capability perhaps, then it is possible to insert a 4050 buffer between the output and the darlington. Because of their FET technology these types of CMOS ICs hardly draw any base current either.

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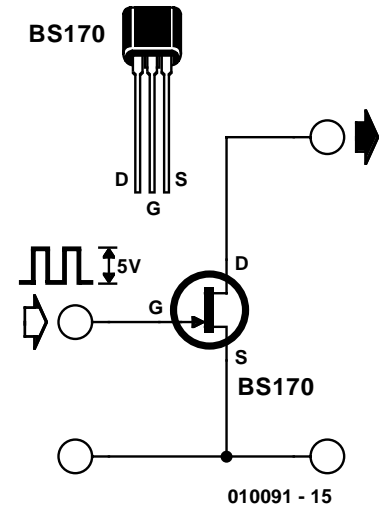


Figure 5. If a FET is used instead of a transistor, a base resistor is not required and the output load is negligible.

Handy S/PDIF Checker

a Digital Audio test unit

Design by G. Kleine

Audio equipment with digital signal interfaces offer high quality sound but trying to listen to what is happening at the interface is not easy without specialised test equipment. A new S/PDIF decoder chip with built-in D/A converters forms the heart of this simple but useful piece of test kit.

The S/PDIF audio digital interface standard has been around for a few years now and is increasingly being adopted by the latest models of CD players, DAT recorders and mini disc systems. Trouble-shooting digital systems invariably calls for the use of expensive test gear to analyse the signals. This circuit idea offers a simple low-cost method of listening-in to the digital interface.

This design uses just one 28-pin SMD chip that together with a voltage regulator and a few passive components produces a useful S/PDIF interface tester. The IC in question is the recently introduced IEC-958 Audio DAC type UDA1350ATS or UDA1351TS from Philips. The block diagram of this IC is shown in **Figure 1** it contains an IEC958 decoder and integrated stereo D/A converters to generate stereo analogue audio output signals from the S/PDIF digital data stream. The UDA1350ATS is an extremely versatile device and to use all the available functions an external microcontroller can be connected via the serial L3 interface, alternatively the chip can be operated in static stand-alone mode by pulling the SELSTATIC input high. In this mode it is not possible to use all the features of the chip such as volume, bass and treble boost, AF filter selection, soft muting or external de-emphasis control but in our application here we are not too concerned about using all the available bells and whistles so the chip is configured in its stand-alone mode.

Signal flow in the UDA1350/1

The digital data stream enters the chip at the

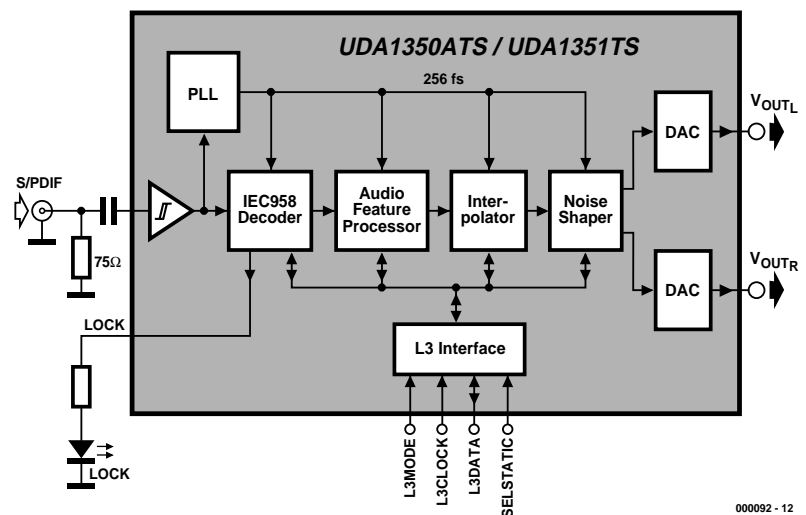


Figure 1. The UDA1350/I block diagram

Table 1.

Technical data of the UDA 1350 ATS and UDA 1351 TS

Parameter	Value
Operating voltage	+2.7 to +3.6 V
Supply current	80 mW @ 48 kHz 110 mW @ 96 kHz
Vref	0.45 - 0.55 V _{DDA}
Input signal level	0.2 - 3.3 V _{PP}
Input hysteresis	40 mV
Sample rate UDA 1350 ATS	28 kHz - 54 kHz
Sample rate UDA 1351 TS	28 kHz - 100 kHz
Output signal	900 mV _{eff}
Signal/Noise ratio	100 dB typ.
Channel separation	96 dB typ.
Output signal difference	0.1 dB typ.

The S/PDIF signal format

Historical perspective

When audio equipment began using digital techniques to store analogue signals (CD players and DAT recorders) the output from the equipment were standard analogue signals. It wasn't long before it was realised that there would be many benefits if the digital audio information were sent between equipment rather than converting to analogue, especially in the professional (studio) environment. The Audio Engineering Society together with The European Broadcasting Union collaborated on a paper outlining a standard for such a digital interface. The EBU-Document Tech. 3250 from November 1985 defined an interface with a 48 kHz (or 32 kHz) sampling frequency and an audio format of 24 bits per channel. Not long after this the electronics companies Sony and Philips jointly specified a consumer version of this interface standard called the S/PDIF (Sony/Philips Digital Interface Standard). The two standards, the professional AES/EBU and the Consumer S/PDIF interface were later combined by the IEC (International Electrotechnical Commission) to produce the IEC 958 standard.

The essential difference between the professional and the S/PDIF standard is not in the coding method of the analogue signal but in the format of additional data sent in the channel status block (see later). The frame structure and audio data coding are identical in both standards.

Subframes

The signal format of an IEC-958-interface consists of subframes, frames and blocks. Each sample of the audio signal is transmitted in a 32-bit subframe. The first four bits of the subframe form the preamble. Three types of preamble are possible, type B indicates that the sampled value in the subframe is for channel A (left) and is the first frame of a new 192 frame data block. Type M indicates left channel data also but this time it is not the start of the block. Type W indicates that the sampled value is for channel B (right).

The next 24 bits contains the digital code representing the sampled audio signal. The sampled value can be 24 bits long or less. CD players use 16 bit samples so the unused preceding bits will always be filled with zeroes. After this data sample a single validity bit is sent, if this bit is set it indicates that a sampling error was detected and the sampled value should be discarded. The next bit is 'user data' and conveys information (together with the other 'user data' bits in each of the subframes in a block). The information conveyed here could be for example text. Next comes the 'channel status' bit. Again each subframe contains a single channel status bit and these bits are used together in each block. These channel status bits contain information on the data channel and would include sampling rate, audio or data mode and professional or consumer mode. A parity bit is included as the last bit in the subframe and allows single bit transmission errors to be detected. Interpolation enables corrupted subframes to be simply discarded.

Frames and Blocks

Each frame contains as many subframes as there are audio channels. A standard stereo signal has one frame containing two subframes, one for the left and one for the right channel. The frames

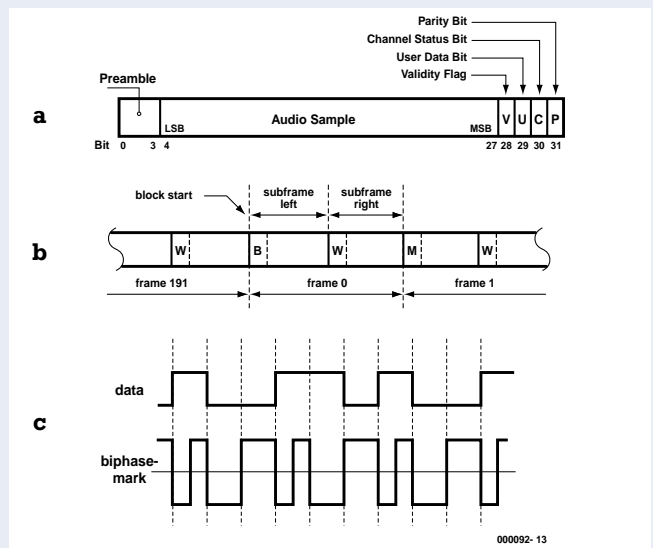


Figure 1. Data format of the S/PDIF signal.

- A subframe
- Frame and block format.
- Biphase mark encoding.

are sent at the sampling rate so that for a CD player using a 44.1 kHz sample rate the data rate will be:

$$2 \text{ channels} \times 32 \text{ bits per subframe} \times 44.1 \text{ kHz} = 2.8224 \text{ Mbit/s}$$

192 frames make up a single block. The block structure allows 384 'user data' bits to be sent in one frame (one 'user bit' per subframe). In practice, the user bits in the left and right subframes are identical so that there is only 192 bits of information sent per block. The same is true for the 'channel status' bits.

Sending the B preamble in the first subframe ensures that the start of each block can be easily detected and the complete block can be correctly decoded. As we already mentioned, the structure of the channel status information differs for the professional and S/PDIF signals. The first bit indicates which of the two standards is used.

Biphase signals

At the physical level, the S/PDIF-interface uses a 75 Ω coaxial cable to convey the digital signal between equipment with Cinch connectors. The signal source equipment generates a 500 mV_{ss} signal level while at the receiving equipment a minimum sensitivity 200 mV_{ss} is necessary. Cable lengths in excess of 10 m are possible with this interface. The signal coding used is Biphase Mark Code (BMC) which is a sort of phase modulation. For every '1' in the data stream two zero crossings of the signal occur and for every '0' there is only a single zero crossing. The signal has no dc component so that ac coupling is allowed.

The professional IEC 958 interface has a signal level ten times higher than S/PDIF (3 to 10 V_{pp}), uses a balanced 110 Ω cable (twisted pair) and of course a different connector. Some form of coupling transformer is generally employed to reduce any earth loop problems.

input pin SPDIF of the UDA1350/1. The signal is amplified to CMOS levels and fed to the IEC 60958 decoder where a Phase Locked Loop (PLL) locks to the data and all the 24 bits of data for both left and right chan-

nels along with some key channel status bits are extracted from the bit stream. The signal now passes to the *Audio Feature Processor*, where in static mode de-emphasis for the IEC 60958 data stream is inserted.

De-emphasis reduces the signal levels at the high frequency end of the signal spectrum which has the effect of also suppressing noise and so improving the signal to noise ratio. The de-emphasis simply compensates for the pre-emphasis that was added by the preced-

ing equipment (CD, DAT source etc). The net effect on the signal should be zero but with an improved S/N ratio. Next the *interpolator* converts the incoming data stream from $1f_s$ to $128f_s$ (where f_s is the sampling frequency) by cascading a recursive filter and a FIR filter. These filters introduce 50 dB attenuation to all of the signal components above about half of the sampling frequency. The *Noise Shaper* operates at $128f_s$. It shifts all of the in-band quantization noise up to frequencies beyond the audio band (10 Hz to 25 kHz). This ensures that a high signal to noise ratio is achieved. The noise shaper output signal is now converted into analogue by Filter Stream Digital to Analogue Converters (FSDAC). These are basically semi-digital reconstruction filters that convert the digital 1-bit data stream into analogue output signals. No additional external filters are necessary and the output signal swing is sufficient to drive a standard line input of an amplifier or a stereo headphone set.

The IEC 958 decoder strips off the left and right 24 bit long audio samples from the incoming data stream and also reads the channel status bits. These bits contain information on the pre-emphasis setting, the audio sampling frequency, the type of two channel Pulse Code Modulation (PCM) coding and the clock accuracy detection. An internal phase locked loop (PLL) enables the system to lock

Table 2.

Sample rate and corresponding data rate.

UDA1351TS	UDA1350ATS	Sample rate	Data rate
X	X	32.0 kHz	2.048 Mbit/s
X	X	44.1 kHz	2.8224 Mbit/s
X	X	48.0 kHz	3.072 Mbit/s
X		64.0 kHz	4.096 Mbit/s
X		88.2 kHz	5.6448 Mbit/s
X		96.0 kHz	6.144 Mbit/s

onto signals with the sample rates of between 28 kHz and 54 kHz. This range includes the most common sample rates of 32 kHz, 44.1 kHz and 48 kHz. Swapping the UDA1350ATS with the pin compatible UDA1351TS will enable the circuit to use sample rates up to 100 kHz.

LED D1 indicates that the IEC 958 decoder has locked onto and recognised the input data stream. When the code is not recognised D1 will be off and the audio output is muted. **Table 2** shows the common sampling rates together with the corresponding data rates and indicates the suitability of the UDA1350ATS or UDA1351TS.

The circuit

The complete circuit shown in **Figure 2** consumes approximately 10 mA quiescent current (with no digital input signal) and less than 30 mA operational. An LM317 voltage regulator (IC2) is used to produce the 3.0 V supply from a 9 V battery.

As we saw earlier IC1 contains several circuit blocks that together perform all the functions of the chip. All of these analogue and digital circuits together on the same chip can give rise to interference and crosstalk especially on the supply voltage. To reduce the possibility of this the IC is manufactured with separate pins for supply voltage to the different

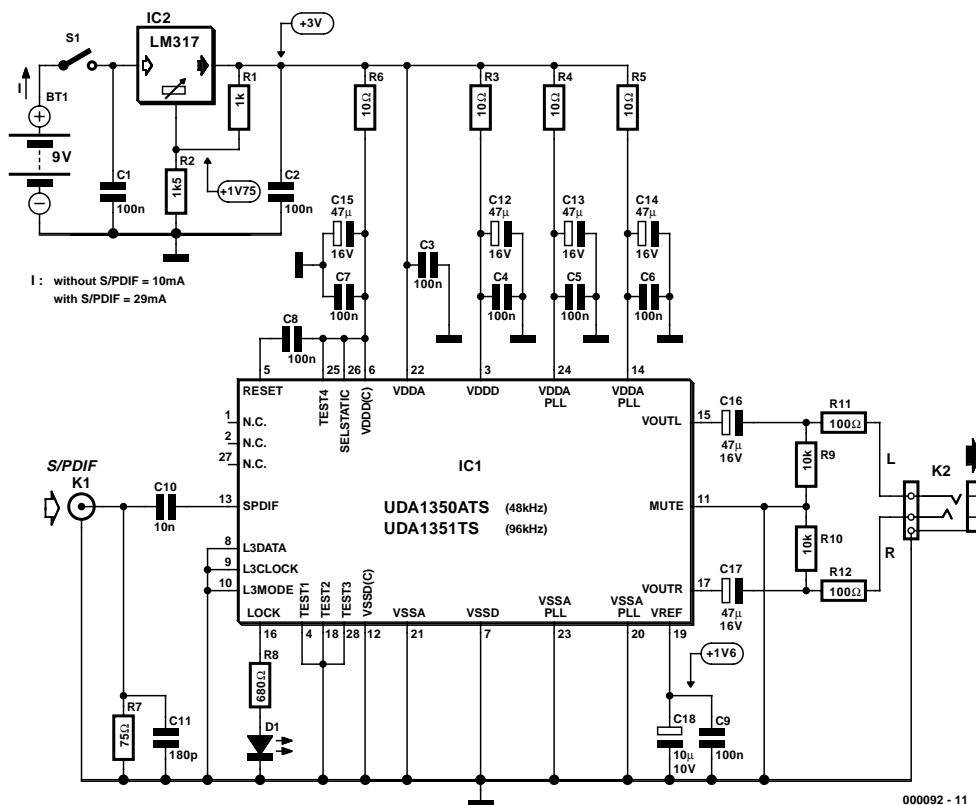


Figure 2. Circuit diagram of the S/PDIF checker

stages. A 10 Ω resistor together with a 47 μF and 100 nF capacitor form a network at each of these supply voltage input pins to ensure minimum interference. Capacitor C8 generates an active high RESET signal when the chip is powered up. The internally generated reference voltage level used by the D/A converters is half of the supply voltage level and is brought out to pin 19 where capacitors C18 and C9 are used to store and smooth VREF.

The COAX cable connecting the digital signal to the input of the test unit has an impedance of 75 Ω . Resistor R7 is used to match the cable impedance with the unit's input impedance and reduce any reflections that would otherwise occur. The level of the digital input signal should be +0.2 V to +3.3 V peak-to-peak and is ac-coupled to the input of IC1 by capacitor C10.

The LOCK output signal from IC1 is generated by the in-built IEC958 decoder and will only be high when a valid PCM audio signal is detected. LED D1 will therefore give a good indication that the input signal is in

order. The active-high MUTE input (pin 11) is not used in this application so it is tied low to ensure the output will not be muted. Any valid digital input signal will always be available as analogue signals at the outputs unless the input signal is corrupted or of the incorrect format in this case an internal circuit will mute the output to prevent the noise burst that would otherwise be audible.

Both audio outputs from IC1 are ac-coupled with 47 μF capacitors (C16 and C17) to the output socket K2. The 10 k Ω resistors ensure that the output signals have a load when there are no headphones connected. The 100 Ω series resistors provide output short-circuit protection.

Building and testing

Both the UDA1350ATS and the UDA1351TS are supplied in an SSOP28 package (Shrink Small Outline Package with 28 pins). There is no ready-made PCB available for the design but in this case it is not too much of a disadvantage. Apart from the main chip there is very little

external circuitry so the entire circuit can be contained on a small piece of SMD prototyping board. Begin building the circuit by fitting the voltage regulator chip IC2 together with R1, R2, C1 and C2. Before any other components are fitted check that 3.0 V is available at its output when a 9 V battery (or better still for test purposes a power supply with current limit set to 50 to 100 mA) is connected to the battery connector. If the voltage level is correct you can now turn off the power and solder IC1 in position along with all the remaining components. Next, using an eyeglass, check all the solder joints and especially the SMD connections for any unintentional solder bridges that you may have made. If you are confident that all is in order, power the circuit up, plug in some headphones to connector K2 and connect a digital audio signal to connector K1. If there is no audio output check the supply voltage again at all the points where it enters IC1 and also check that VREF is at 1.6 V (pin 19 on IC1). Any level that is less than it should be indicates that there is probably a solder bridge causing a short-circuit somewhere so power down, take up the eyeglass and look again. If all the voltages are OK and still nothing can be heard try connecting a different digital signal at the input.

(000092-1)

High Power Adjustable Load

with power limiter

Design by P. Hirschbrich

An adjustable load resistance is indispensable for the realistic testing of power output stages. This circuit provides an electronic alternative to the use of inconvenient high-wattage resistors.

Technical specifications

Input voltage:	0 to +100 V _{DC}
Sink current:	0 to 20 A
Load resistance:	< 1 Ω to > 100 kΩ
Maximum power dissipation:	approximately 100 W
Linearity error:	< 0.5 %

The circuit described here has two modes of operation. It can function as an adjustable current sink or as a variable load resistance. At a voltage of between 0 V and a maximum 100 V the sink current can be set in the range 0 A to 20 A. As a load resistance it can be set to values from less than 1 Ω to more than 100 kΩ. The maximum power dissipation is in the region of 100 W. The circuit is designed for operation with DC voltages, but with the addition of a rectifier can also be used with (low frequency) AC voltages. A carefully constructed and calibrated unit can deliver a linearity of better than 0.5 %.

Current Sink and Load Resistance

Let us first consider the circuit as a current sink (S2 in position 'CC': constant current) initially ignoring the part of the circuit consisting of the three op-amps IC1.B, IC1.C and

IC1.D and the surrounding components.

The load circuit consists of a power FET and resistor R2. An electrolytic capacitor is connected in parallel with these, smoothing the voltage across the FET and R2 and suppressing spikes. The FET is controlled by IC1.A. The op-amp attempts to keep the voltage at its inverting input (that is, the voltage across R2) the same as that on its non-inverting input. The latter is in turn equal to the voltage on the wiper of 10-turn potentiometer P1. We have therefore achieved what we sought to achieve: the current drawn is proportional to the wiper setting of P1. This means, of course, that the wiper voltage of the potentiometer must be kept constant, and this is guaranteed by the 2.5 V voltage reference D7. At the upper limit of the potentiometer's travel we have a voltage of 0.779 V (because of R3), assuming that S1 is open.

Only the first five turns of the ten-turn potentiometer can be used before the power limiter comes into effect. If the current sink is to be used within its power limits, a smaller range of voltages across R2 is more useful: this is the function of R12 and trimmer P2. When S1 is closed these are connected in paral-

lel with P1 and are adjusted so that the voltage is reduced by a factor of ten. This allows a current setting in the range 0 A to more than 5 A.

If S2 is switched to position 'CR' (constant resistance), a different voltage is applied across P1. The source is no longer provided by the voltage reference but by the input voltage. The remainder of the circuit works as before, meaning that the input current is proportional to the input voltage, and the constant of proportionality is equal to the constant resistance. With R4 being 475 kΩ load resistances from infinity to 1 Ω (at the extreme settings of P1) can be achieved.

Power Limiter

The remaining three op-amps in the LM348 form a power limiter to ensure that the power dissipation of the circuit remains under control. An analogue multiplier is not used in this circuit. R10 and R11 form a voltage divider across the input voltage, which operates linearly as long as diode D2 does not conduct. This does not happen abruptly, but rather follows the characteristic curve of the diode, providing a gradual transition as the input voltage rises. IC1.C buffers this voltage and drives

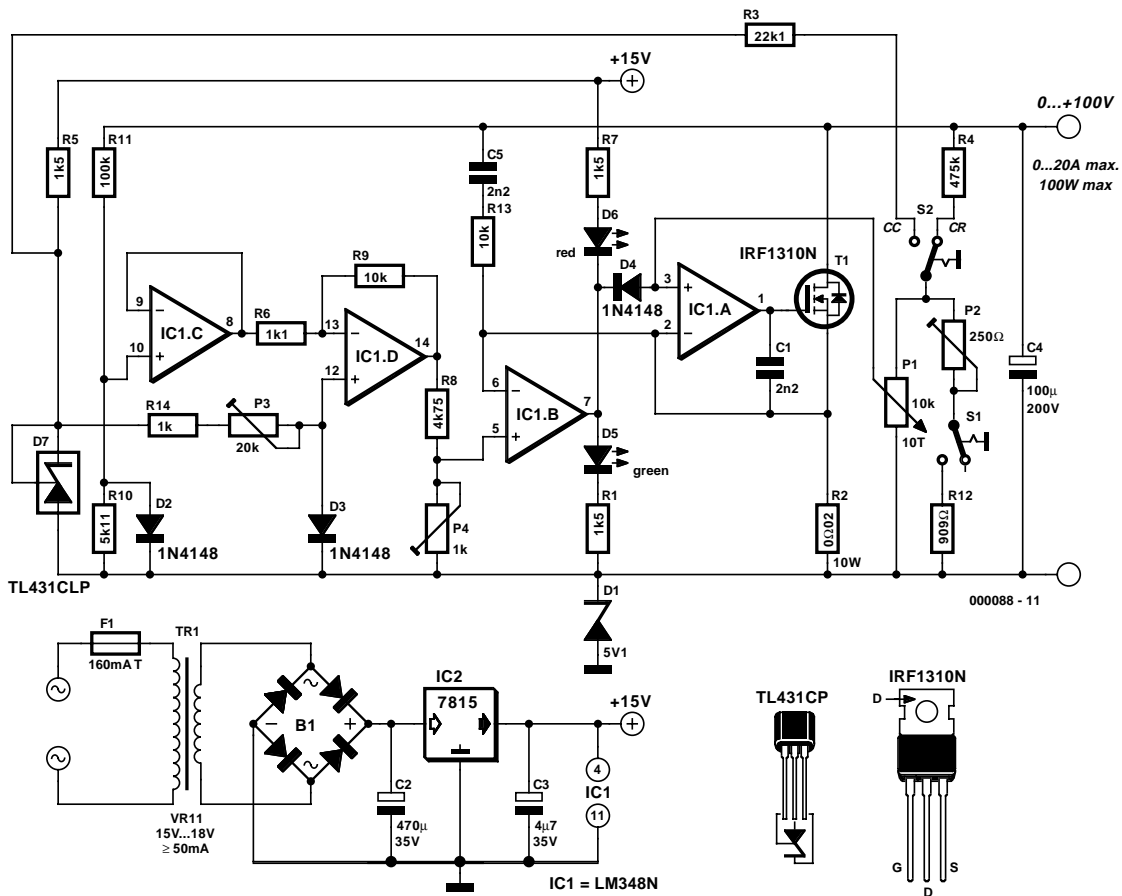


Figure 1. An op-amp, a resistor and a FET form the basis of this item of test equipment.

inverter IC1.D, which inverts the characteristic curve of the diode and amplifies it with a gain of ten. The output voltage of the op-amp is raised by an amount equal to the diode forward voltage via D3.

The operating point of D3 is determined by the reference voltage, R10 and the setting of trimmer P3. Finally, comparator IC1.B compares the output voltage of the inverter with the voltage across R2 and, if the power limit is exceeded, pulls down the control voltage to op-amp IC1.A, turns off the green 'OK' LED and lights the red 'warning' LED.

Note that this is only a crude protection circuit for limiting the power above about 100 W. It is highly dependent on device-to-device variations among diodes and on temperature, but is entirely adequate for protecting against excessive current draw from, for example, a 12 V car battery.

The circuit as a whole is powered from a mains power supply, which as usual consists of a mains trans-

former (15 V to 18 V, at least 50 mA), a bridge rectifier and a 15 V fixed voltage regulator.

C2 smoothes the rectified DC voltage and C3 suppresses transients. Since the op-amps are operating near to the lower input voltage, a negative supply is required for them (unless rail-to-rail types are resorted to). To this end Zener diode D1 provides a negative supply voltage about 5 V below the lower input voltage.

Construction and Calibration

Since we have not shown a printed circuit board layout for this circuit, a few words on construction are in order. The circuit presents no enormous technical difficulties, and simple prototyping board will suffice for construction, but a suitably heavy-duty enclosure must be found. The FET must be able to dissipate up to about 85 W under peak conditions, and the heat must be carried away

using a large heat sink. The data sheet for the FET indicates that a heatsink rated at 0.9 K/W or better is required. Alternatively, the FET can be fitted with a modern CPU fan. The circuit is intended for hobby use or for short laboratory tests, and not for continuous use (for example 24 hour soak tests). If additional thermal protection for the transistor is desired, a thermal relay (closing at 105 °C) can be fitted. The relay is glued to the transistor using two-part adhesive, and wired so as to short pin 3 of IC1.A to the lower input potential.

If you do not wish to fit a fan, but nevertheless want to operate either at high loads or continuously, then you can connect up to five FETs in parallel (for example type BUZ344). The circuit also works well with 150 W to 200 W power Darlingtontons such as the MJ11016 in a TO3 package, but not in a parallel arrangement and only with input voltages above 1 V.

R2 also plays its part in dissipating power. Either a 15 W power resistor should be fitted, a few millimetres above the surface of the circuit board, or a 10 W power resistor in a metal housing with heatsink can be used.

The FET, the power resistor R2 and elec-

trolytic C4 should be placed next to one another centrally on the rear side of the (horizontally-mounted) circuit board, with the two high-current terminals (a suitable type is available from AMP) on either side. The components should be connected with thick wire, taking care not to overheat the components when soldering.

The control electronics should be placed immediately next to the FET, and in particular the connection between the output of the control op-amp IC1.A and the gate of the FET should be short.

Since the electronics are fitted at the rear of the enclosure, the components fitted to the front panel (S1, S2, P1, the LEDs and the two banana sockets) must be wired to the circuit board. For the input we have already suggested AMP connectors, while for the remaining — low power — connections ordinary sig-

nal wire will do.

The power supply can be fitted in a spare corner of the circuit board. The mains enters via a switched and fused input socket, which avoids having the mains switch on the front panel. LED D5 serves as a power indicator.

In the interests of safety, the 'thin' wiring should be kept well separated from the power supply and from the thick input wiring.

Check over the circuit once more, fit it into the enclosure and check the wiring and that the heatsink is isolated from the FET (!), and switch on at the mains. Assuming that no smoke appears, and that the supply and reference voltages are correct, the unit can be switched off again

and the LM348 fitted in its socket. The control circuit can now be calibrated.

Turn the unit on again and apply a voltage of 10 V to the input. The voltage at pin 13 of the IC should be the same as that on pin 12 (P3). At the output of IC1.D a voltage of 0.95 V should be measured, and we want the current through the FET to be limited to about 10 A. Adjust P4 to set the voltage on pin 5 to 200 mV. This corresponds to a current of 10 A at 10 V (or 100 W). You will find that a little patience is required in adjusting P3 and P4. The adjustment of P2 has been described above. The calibration is now done: fit the lid to the enclosure and your unit is complete.

(000088-1)

Versatile Final Amplifier

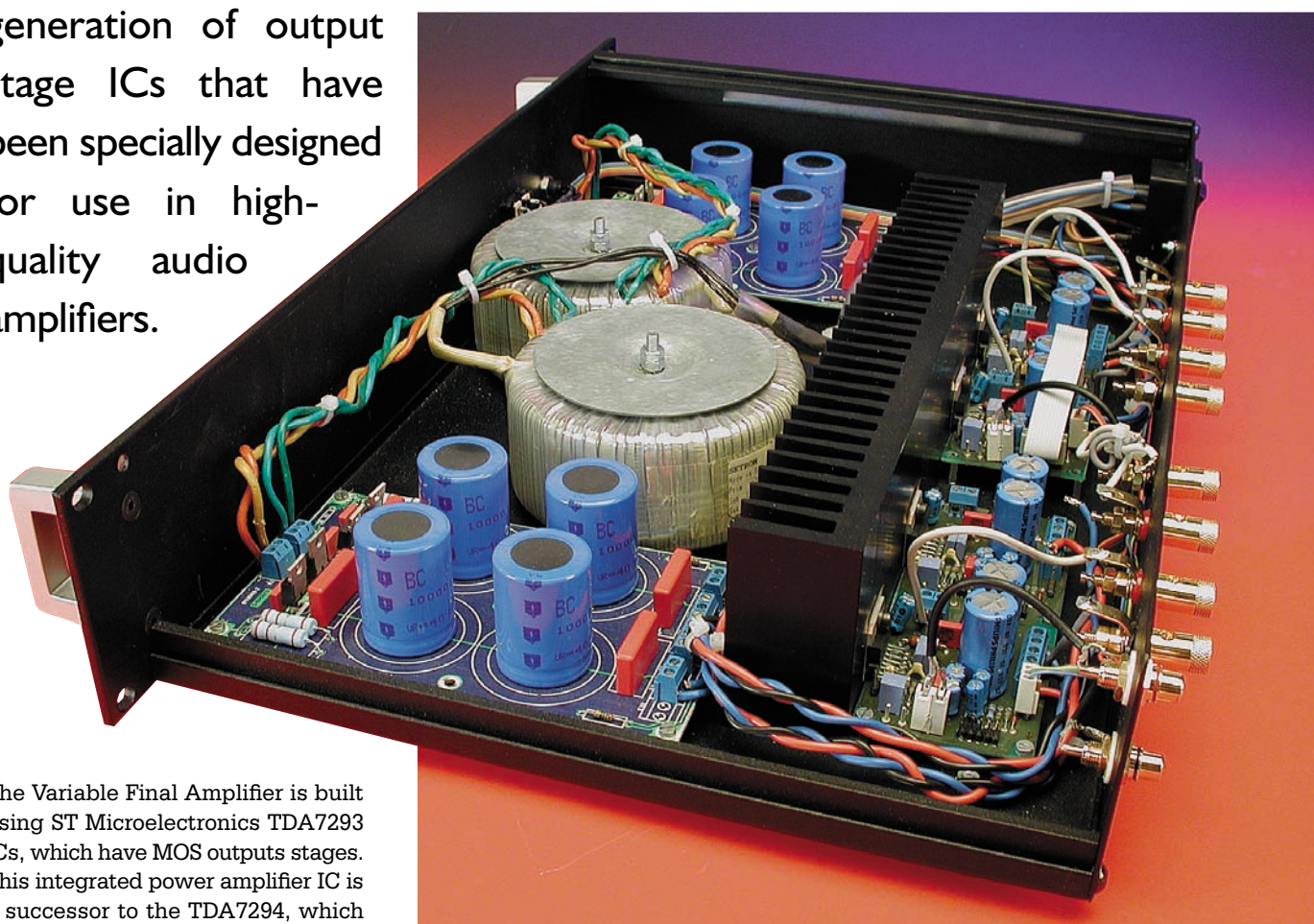
audio power with many features

By U. Böhmke

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Only a few years ago, many music lovers turned up their noses on seeing a hybrid circuit or integrated circuit in a final amplifier, but now there is a new generation of output stage ICs that have been specially designed for use in high-quality audio amplifiers.



The Variable Final Amplifier is built using ST Microelectronics TDA7293 ICs, which have MOS outputs stages. This integrated power amplifier IC is a successor to the TDA7294, which was described a long time ago in an *Elektor Electronics* data sheet (May 1993) based on an SGS Thomson product announcement. Evidently SGS Thomson experienced difficulties in the development of this IC, since it took three full years until the IC

appeared on the market. However, at the end of 1996 it was ready, and it was used in the 50-W A.F. Amplifier (November 1996 issue), which has been built by many hobbyists.

If we compare these two power amplifier ICs, the first thing that strikes us is the similarity of their internal circuitry and external wiring (see **Figure 1**). Two small but important differences allow the new IC amplifier to be

used much more flexibly: first, the connection between the input and output stages can be opened to allow the output stage to be driven externally, and secondly there is a special charging output for the bootstrap capacitor. We will see what these features allow us to do.

In addition to having good specifications and sounding good, the TDA7293 is distinguished by high reliability. Its quiescent-current stability is very good, the output is short-circuit proof and integrated thermal-shutdown circuitry prevents overheating. Integrated switch click suppression in the form of a mute/standby function makes an output relay unnecessary, which helps the damping factor. And if in spite of everything the preamplifier overdrives the final amplifier into clipping, the CLIP DET output announces the fact. The complete data sheet of the TDA7293 is available at the following address:

<http://eu.st.com/stonline/books/pdf/docs/6744.pdf>

Basic concept

A compact final amplifier is particularly interesting for anyone who wants more than simple stereo operation — in other words, anyone who wants to configure a system with biampification, active speakers or multiple channels. Such systems sound best when all amplifiers have the same construction.

Monoblock

In the simplest case, a complete final amplifier and power supply are built into a single enclosure, resulting in a small, ready-to-use monoblock.

Full amplifier

If we place a selector switch and a potentiometer ahead of the inputs of a stereo version of our final amplifier, we have a miniature version of a full amplifier. The gain of the compact final amplifier is fully adequate for a high-level signal source, such as a CD player. If more gain is needed, a small opamp preamplifier stage can be added to the configuration.

Biamping

Cognoscenti have long appreciated 'biamping' as the logical extension of 'biwiring', but in Germany this idea has become known to a relatively large group of listeners only within the last few years. This is no doubt due to the fact that biamping requires two stereo final amplifiers, which naturally means twice the expense if they are bought ready-made. However, with DIY construction the additional cost is not as great, since all that is necessary is to fit an extra final amplifier into the enclosure. Our compact final amplifier is an excellent choice for such use, due to its small size. In such applications, its relatively small output power (compared with large final amplifiers) is by no means a disadvantage. Besides better utilisation of amplifier capacity, biamping can produce an audible improvement in spaciousness and resolution.

In purely practical terms, two final amplifiers are need for each stereo channel, with their inputs connected in parallel. The output of

one amplifier is connected to the bass/midrange driver, while the output of the other amplifier is connected to the treble driver. In case of a three-way loudspeaker, one output is connected to the bass driver and other one is connected to the midrange/tweeter unit (true fanatics even go for 'triwiring' and dedicate a separate amplifier to each of the three drivers). Naturally, the passive crossover network in the loudspeaker must be designed for biwiring, which means that the high-pass and low-pass filters must be built as independent assemblies.

Active loudspeakers

The principal advantage of an active-loudspeaker system is that the only thing between the amplifier output and the loudspeaker driver is a length of wire. This eliminates both the complex load on the amplifier and the reduction in the damping factor that result from using a crossover network. As regards amplifier power, the same considerations apply as for biamping.

An active speaker solution is always an option when developing a new speaker design. Generally speaking, existing well-balanced speakers cannot easily be converted into active loudspeakers. Attempts to do so often result in a 'negative improvement' in the sound quality. Biamping is a more effective approach for improving a passive system.

Multichannel systems

Due to its small size, the compact final amplifier is naturally also especially suitable for use in multichannel systems. Separate amplifiers are preferable for the front, centre and rear channels. For the subwoofer, a parallel or bridge circuit is an ideal solution.

Amplifier circuit board with options

The schematic diagram of the compact final amplifier (Figure 2) represents a standard application circuit for the TDA7293, although it has a few unique features. The input connections are duplicated to allow the music signal to be 'daisy-chained' to a following final amplifier. After the input we find the usual filters. C1

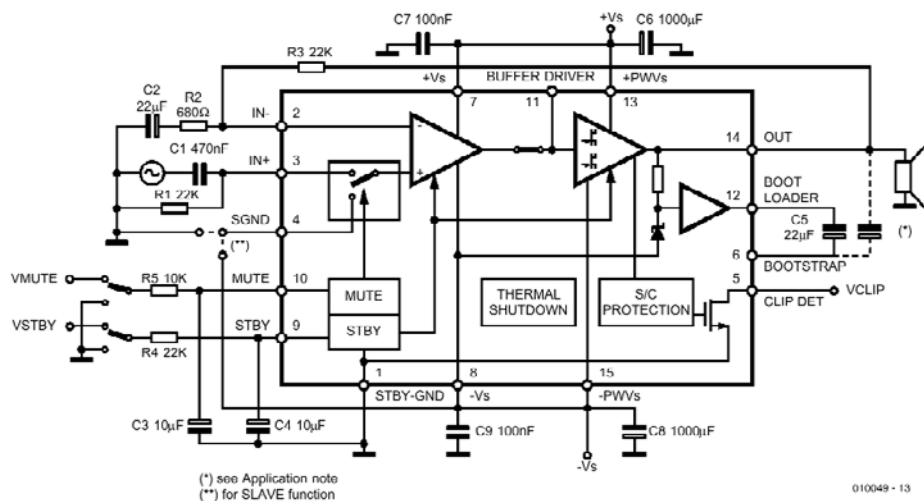


Figure 1. Internal circuitry of the TDA7293V.

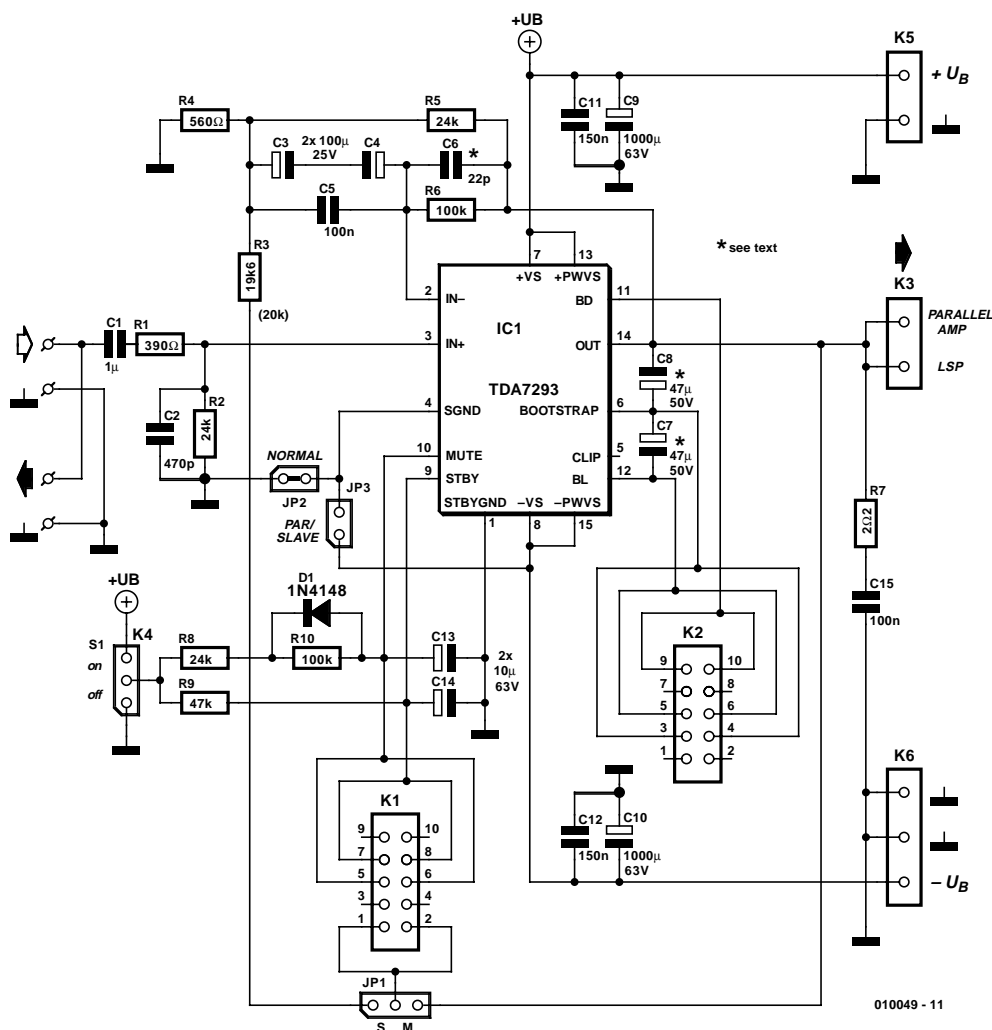


Figure 2. The circuit diagram of the amplifier corresponds to the standard application but has a number of special configuration options.

and R3 form an input high-pass filter that isolates the input from any dc voltage present at the output of the previous stage.

The TDA7293 can be seen as a non-inverting operational amplifier. The gain is set to around 35 by the negative feedback network. This amount of gain results in the best balance of speed, bandwidth and stability. In order to avoid amplifying the input offset voltage, the amplifier is ac coupled. Capacitor C6 improves the square-wave response.

Good-quality components must be used for the Boucherot network (R7 / C15). R7 must be a low-inductance type, while a foil capacitor must unconditionally be used for C15.

Although it is possible to obtain more output power by operating two modules in parallel (for a 2-Ω or 4-Ω

load) or in a bridge configuration (for an 8-Ω load), the TDA7293 gives the best results (in terms of both measurements and listening tests) when used alone. Consequently, parallel and bridge configurations should be used only for subwoofers.

If only a single amplifier module is used, the load impedance should not be less than 4 Ω. Since the protective circuitry cannot cope with extremely low-impedance or complex loads (such as the Infinity Kappa), the amplifier will be destroyed!

However, two amplifiers can easily be connected in parallel. The power dissipation is then divided over two packages and can thus be greater than with a single module. The internal resistance drops in proportion to the number of modules used. This yields certain advan-

tages, particularly with load impedances less than 8 Ω, and is the only manner in which a 2-Ω load can be driven. With a 4-Ω load, the supply voltage can be raised to the 8-Ω level, with the result that the output power can be increased to more than 100 W.

In bridge operation, one amplifier works against the output of a second, inverting amplifier instead of against ground. Theoretically, this doubling of the output voltage swing results in quadrupling the output power into a 4-Ω load. However, the loudspeaker must have an impedance of at least 8 Ω, due to thermal considerations, so only half of this theoretical increase can actually be realised, but that is still good for up to 150 W (depending on the quality of the power supply). The damping factor is reduced by a factor of 2 relative to a single amplifier driving an 8-Ω load.

The component connected to pins 1, 9 and 10 provide switch click suppression. The

COMPONENTS LIST
Amplifier Circuit Board

Resistors:

- R1 = 390Ω
- R2,R5,R8 = 24kΩ
- R3 = 19kΩ26 (20kΩ)
- R4 = 560Ω
- R6,R10 = 100kΩ
- R7 = 2Ω2 2W
- R9 = 47kΩ

Capacitors:

- C1 = 1μF MKT (lead pitch 5 or 7.5mm)
- C2 = 470pF
- C3,C4 = 100μF 25V radial
- C5 = 100nF
- C15 = 100nF (lead pitch 7.5mm)
- C6 = 22pF
- C7 = 47μF 50V radial
- C9,C10 = 1000μF 63V radial
(max. dia. 17 mm)
- C11,C12 = 150nF (lead pitch 7.5mm)
- C13,C14 = 10μF 63V radial

Semiconductors:

- D1 = 1N4148
- IC1 = TDA7293V (STMicroelectronics)

Miscellaneous:

- JP1 = 3-way pinheader *
- JP2 = 2-way pinheader with jumper *
- JP3 = 2-way pinheader *
- K1,K2 = 10-way boxheader *
- K3,K5 = 2-way PCB terminal block (lead pitch 5mm)
- K4 = 3- way pinheader
- K6 = 3- way PCB terminal block (lead pitch 5mm)
- Heatsink *
- Enclosure *
- PCB, order code **010049-I**
(see Readers Services section)

* see text

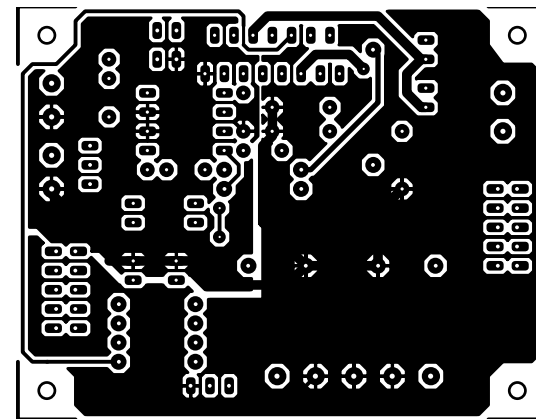
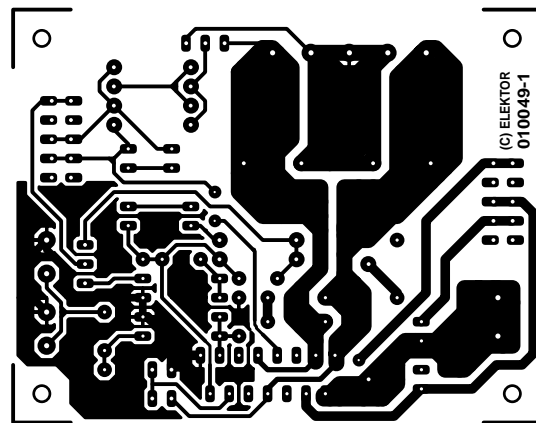
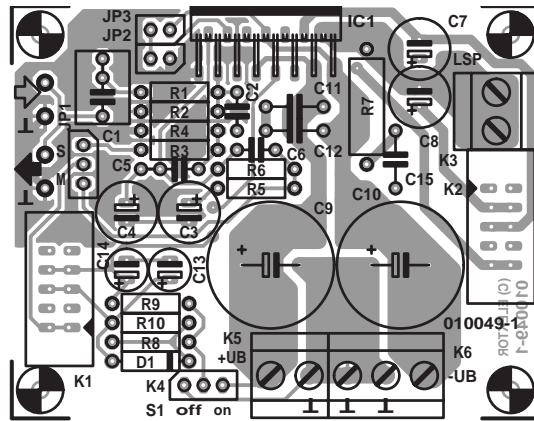
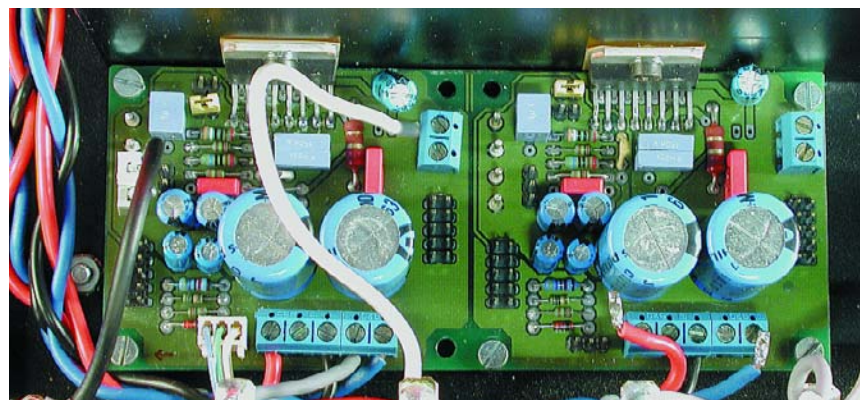


Figure 3. Small and double-sided — the printed circuit board for the Versatile Final Amplifier.

TDA 7294

The TDA 7294, which is the predecessor to the TDA 7293 and well known to readers of Elektor Electronics, can also be used with the circuit board for the Versatile Final Amplifier if the following considerations are taken into account:

- The maximum supply voltage must not exceed ±40 Volts.
- Parallel operation is not allowed.
- Bootstrap capacitor C8 must be fitted, with C7 being omitted.



	Normal	Bridge mode		Parallel mode	
		Master	Slave	Master	Slave
JP1	open	to M	to S	open	open
JP2	installed	installed	installed	installed	open
JP3	open	open	open	open	installed
PCI,2	input	input	bridge	input	open

Table 1: Jumper settings

selector switch (S1) is connected to connector K4. When S1 is switched to the supply voltage, the TDA7293 awakens from the standby mode after a brief delay, and shortly there-

after the mute circuit activates the output. If S1 is switched to ground, the output is first muted and then the IC goes into the standby mode, in which its current consumption is

only 0.5 mA.

Naturally, if several amplifiers are connected to a common power supply (for biamping, active loudspeakers, parallel operation or bridge operation), this function should be controlled using a single common switch. This can be achieved by using 10-way flat cable to interconnect the K1 connectors of the individual amplifiers, which is already the case for parallel and bridge configurations.

Caution:

This must only be done with final amplifiers that are powered from a single **common** power supply!

COMPONENTS LIST

Power Supply

(depending on number of final amplifiers)

Resistors:

- R1-R4 = 0Ω15 5W
- R5,R6 = 4kΩ7
- R7 = 12kΩ

Capacitors:

- C1-C4 = 47nF ceramic

- C5,C6,C11,C12 = 3μF3 250VDC / 160VAC MKT (size 11x21x31.5mm) (e.g., Epcos B32524-Q3335-K, Farnell # 331-3311)
- C7-C10 = 10,000 μF 63V radial, lead pitch 10mm, max. dia. 45mm), PCB mount

Semiconductors:

- D1-D4 = BYV29-200
- D5 = high efficiency LED

Miscellaneous:

- K1-K10 = 2-way PCB terminal block (lead pitch 5mm)
- Mains transformer, 2 x 22V at 225VA PCB, order code **010049-2**

(see Readers Services section)

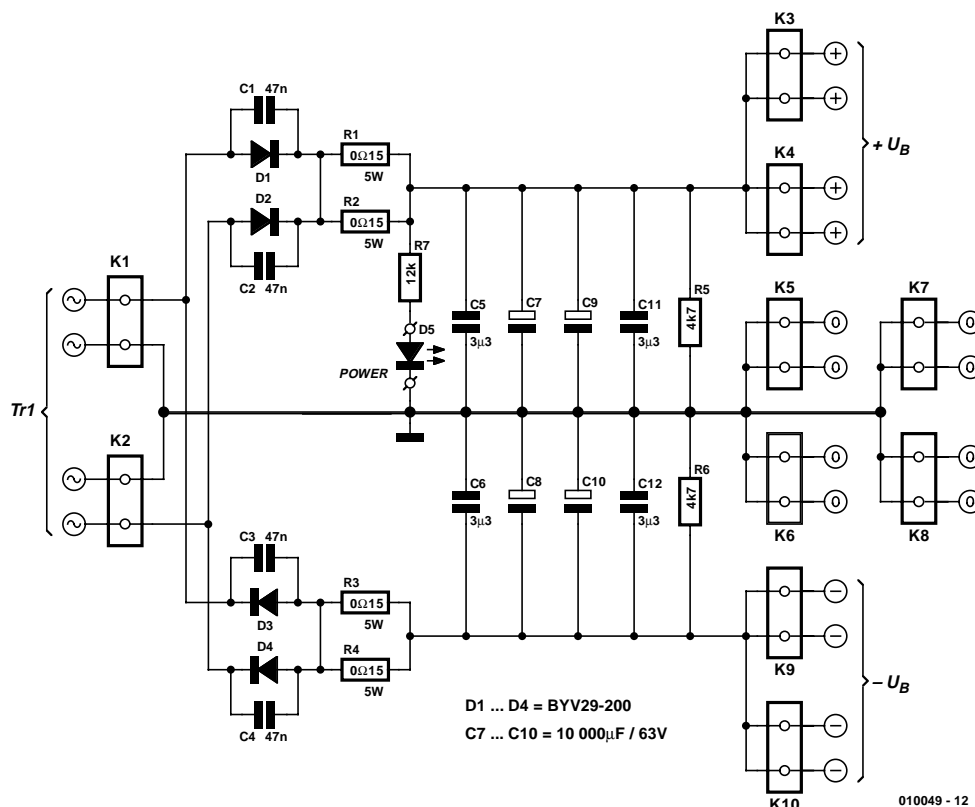


Figure 4. A classic design, but with fast rectifier diodes instead of a bridge module — the power supply for the Versatile Final Amplifier.

Operating mode settings

There is nothing particularly exciting to say about populating the double-sided circuit board shown in **Figure 3**. Build as many boards as you need, but do not fit the amplifier ICs right away. Once the circuit boards and heat sinks have been firmly attached to the

enclosure, insert the leads of the amplifier IC into the holes in the board, screw the IC tight to the heat sink and then solder the leads from the bottom side of the board. This is because it is extremely important that the IC lies absolutely flat on the heat sink, since otherwise the

overtemperature protection will become active after only a few moments.

Several jumpers and wire links must be set or fitted according to the desired operating mode (see **Table 1**). There are also a number of other special considerations, to wit:

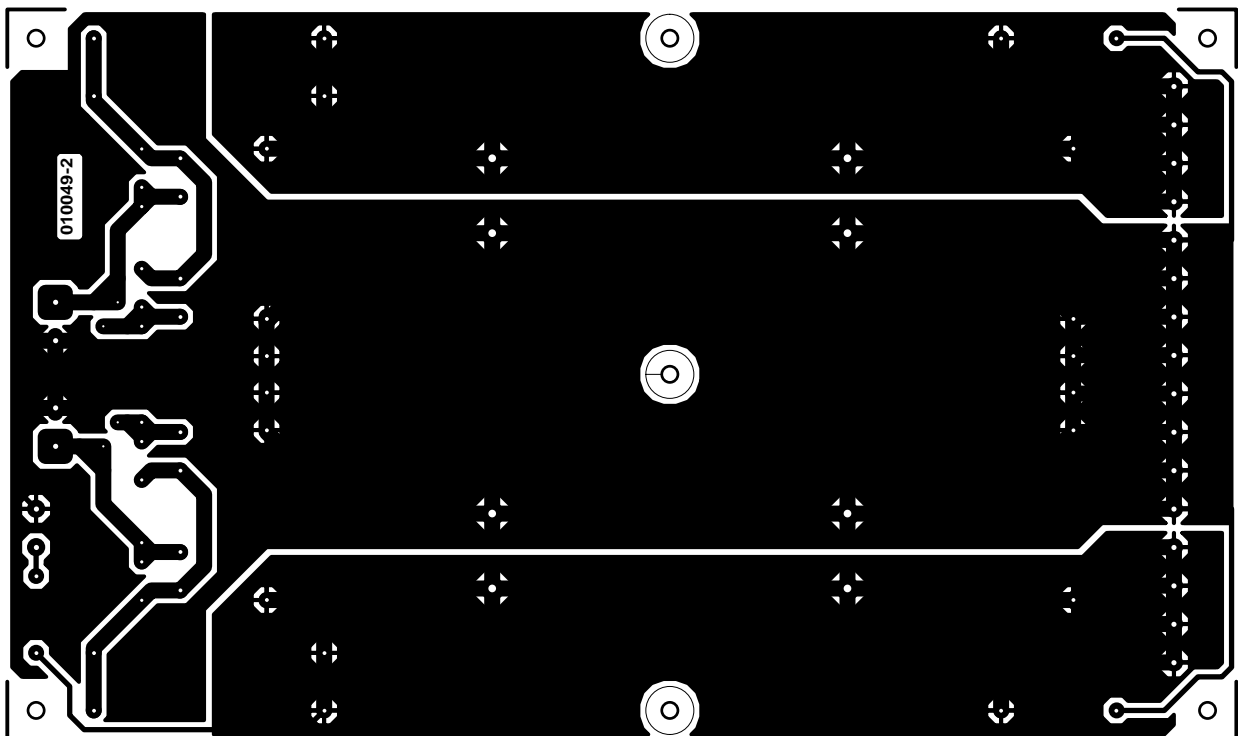
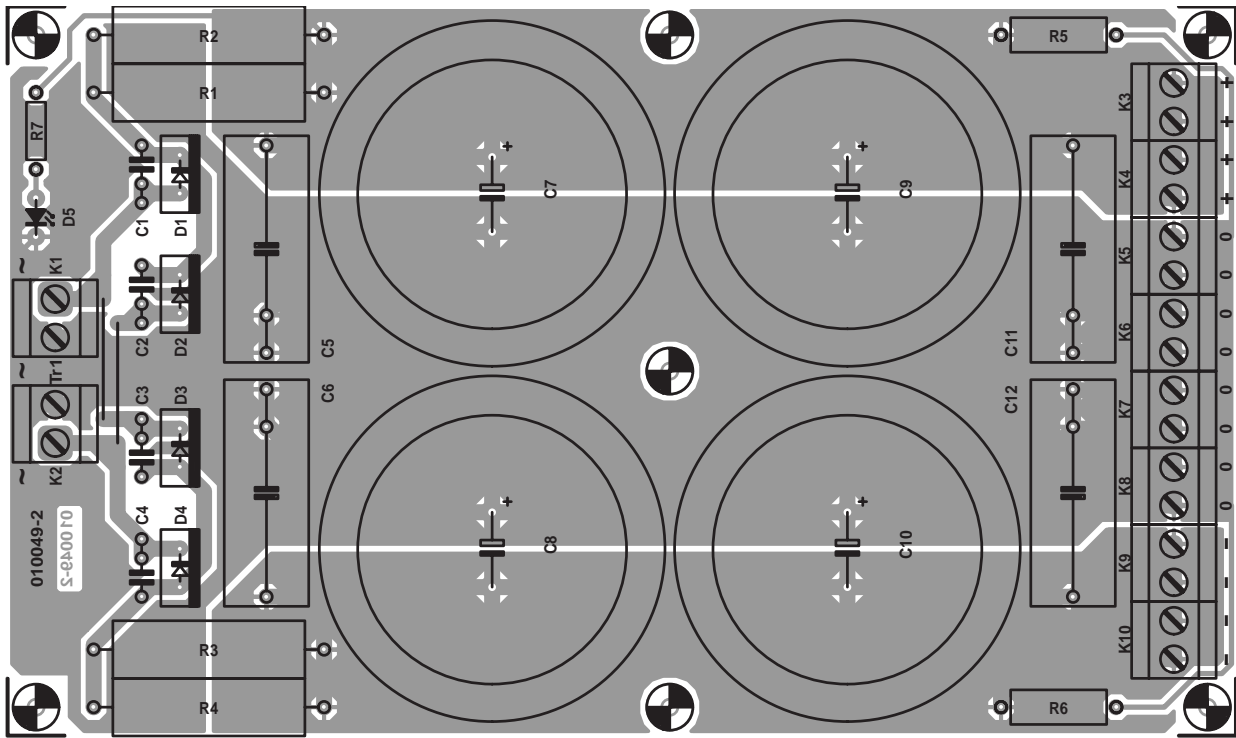


Figure 5. Up to four final amplifier circuit boards can be connected to the power supply board.

- In normal operation without the bridge configuration, jumper JP1 must never be installed!
- For parallel operation, a number of components must be altered on the slave amplifier board. R2, R4 and C5 are replaced by wire links, while R5, R6 and C6 are omitted. Connector K1 of the master amplifier is connected to K1 of the slave amplifier by a 1:1 10-way flat cable, and the same arrangement is used for connector K2. The screw terminals (K3) of both final amplifier boards must be interconnected using wire with a cross-sectional

area (c. s. a.) of at least 1 mm².

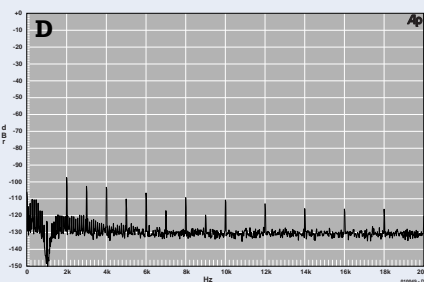
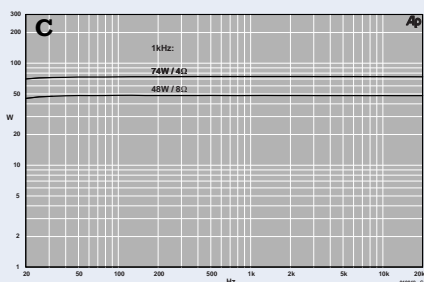
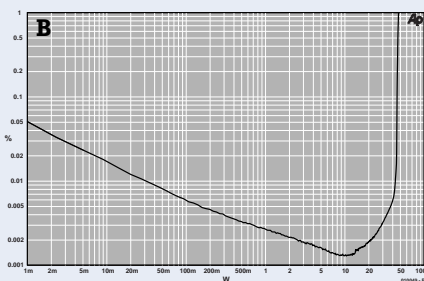
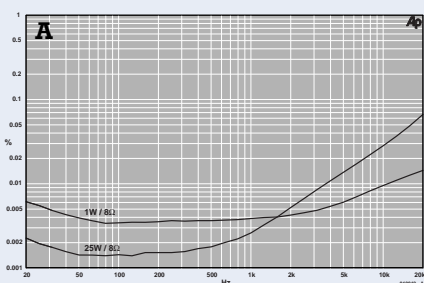
- In bridge operation, the K1 connectors of the two final amplifiers must be interconnected using a 1:1 10-way flat cable.
- The heat dissipated by the IC (up to 50 W!) must be transferred to a suitably dimensioned heat sink. Care must be taken to ensure good air circulation around the heat sink. The stereo biampifier shown in the picture at the beginning of this article uses a single common heat sink for the four final amplifier modules

(Fischer SK56, R_{TH} = 0.45 K/W).

- Glass-fibre reinforced silicone foil (Fischer WB) or a Kapton washer can be placed between the amplifier IC and the heat sink for electrical insulation. In either case, heat-sink paste is not necessary! An insulating shoulder washer must be used with the fastening screw.

Power supply

These compact final amplifiers deliver a lot of power relative to their small dimensions. The power supply (Figure 4) is therefore generously dimensioned. Up to four final amplifier



Measurement results

[single amplifier, power supply with 2 · 22 V / 225 VA transformer and 4 · 10,000 mF capacitance]

Parameter	Measurement conditions	Measured value	
Input sensitivity	47 W/8 Ω	560 mV	
Input impedance		24 kΩ	
Sine wave power	0.1 % THD	47 W/8 Ω 73 W/4 Ω	
Sine wave power, parallel mode	0.1 % THD	50 W/8 Ω 83 W/4 Ω 122 W/2 Ω	
Sine wave power, bridge mode	0.1 % THD	125 W/8 Ω	
Bandwidth	1 W/8 Ω	6.5 Hz – 200 kHz	
Slew rate		8.5 V/μs	
Signal-to-noise ratio	1 W/8 Ω B = 22 Hz – 22 kHz	>98 dB(A) >95 dB linear	
Total harmonic distortion + noise (bandwidth 80 kHz)	8 Ω	4 Ω	
	1 kHz	<0.004 % (1 W)	<0.006 % (1 W)
		<0.003 % (25 W)	≈ 0.003 % (50 W)
20 kHz	<0.07 % (25 W)	<0.08 % (50 W)	
Dynamic intermodulation distortion	3.1-kHz square wave and 15-kHz sine wave	0.006 % at 1 W/8 Ω 0.06 % at 20 W/8 Ω	
Damping factor at 8 Ω	1 kHz	> 1000	
	20 kHz	> 750	

modules can be connected to a single power supply circuit board. If at all possible, you should use a separate power supply for each stereo channel, as well as for the subwoofer.

The power supply uses modern, fast discrete diodes for rectification instead of a bridge module. The circuit board shown in **Figure 5** has room for four sturdy 10,000- μ F electrolytic reservoir capacitors. A total of up to 88,000 μ F can be fitted on the circuit board.

The ideal secondary voltage for the transformer is 2×22 V. With such a power supply, the compact final amplifier provides slightly less than 75 W into 8 Ω or around 47 W into 4 Ω . If you want to have 80 W into 8 Ω , you should use a transformer with a 2×30 V secondary, but in this case a 4- Ω load can only be driven using two amplifier modules connected in parallel. This is only recommended for use in the bass range. If only a relatively small amount of output power is needed, the secondary voltage can be reduced to 2×18 V, for which the output power is around 30 W into 8 Ω or 50 W into 4 Ω . In this case, the rated voltage of the capacitors can also be reduced to 35 V. For biamping or active-loudspeaker systems, this is often more than adequate!

Around 1.5 W of transformer capacity should be provided for each watt of amplifier

capacity. Only extremely high-quality encapsulated toroidal-core transformers should be used. The less expensive open types tend to be noisy.

The amplifiers are effectively switched on and off using switch S1. However, since S1 only switches a control voltage, a mains power switch (possibly illuminated) must always be placed in the primary circuit of the transformer for reasons of safety, so that the amplifier can be disconnected from the mains network when no one is present.

Tests and measurements

Assuming the use of an adequate power supply and good-quality components, the Versatile Final Amplifier is distinguished by a warm, vivacious sound image. In biamping or active-loudspeaker operation, it can compete with final amplifiers having much greater output power.

Figure A shows total harmonic distortion plus noise versus frequency (measured with a bandwidth of 80 kHz). The upper curve in the

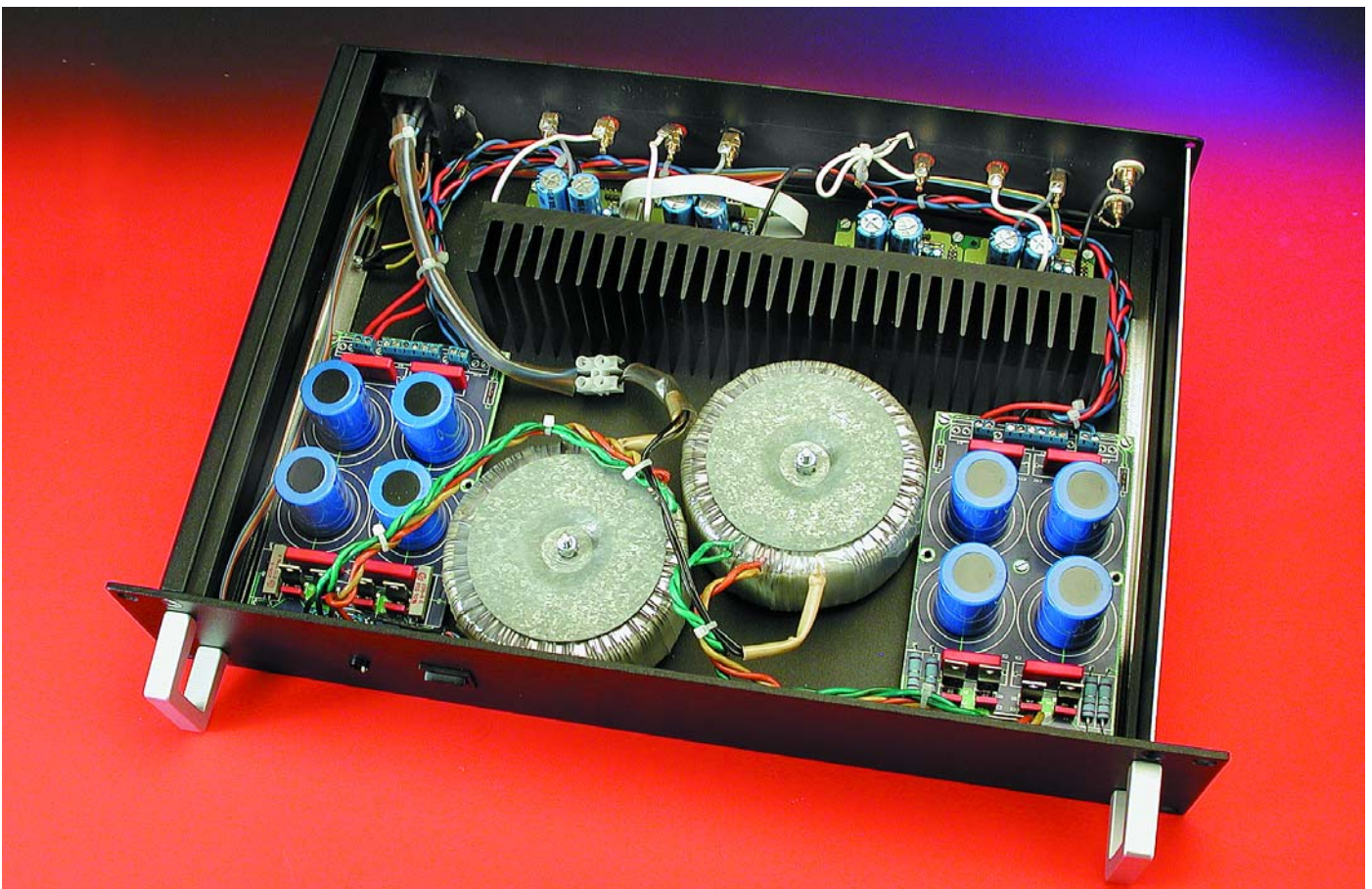
lower-frequency region is for 1 W into 8 Ω , while the other curve is for 25 W into 8 Ω . At the 1-W level, predominantly noise is measured up to around 2 kHz; at 25 W there is a clear increase in distortion above 500 Hz.

Figure B also shows total harmonic distortion plus noise, this time as a function of signal amplitude with an 8- Ω load. Here the bandwidth is limited to 22 kHz to make the effects of distortion at high amplitudes more visible. The distortion rises above the noise level for power levels greater than 10 W.

Figure C, which shows the maximum power into 4 ohms (74 W) and 8 ohms (48 W) at a bandwidth of 80 kHz and a THD of 1 %, is not particularly exciting. The slight droop below 20 Hz is hardly worth mentioning.

Finally, **Figure D** shows the Fourier spectrum of a 1-kHz signal at 1 W into 8 Ω . The second harmonic has a value of only -97.5 dB (THD+N = 0.0037 %), with the higher harmonics being even further down.

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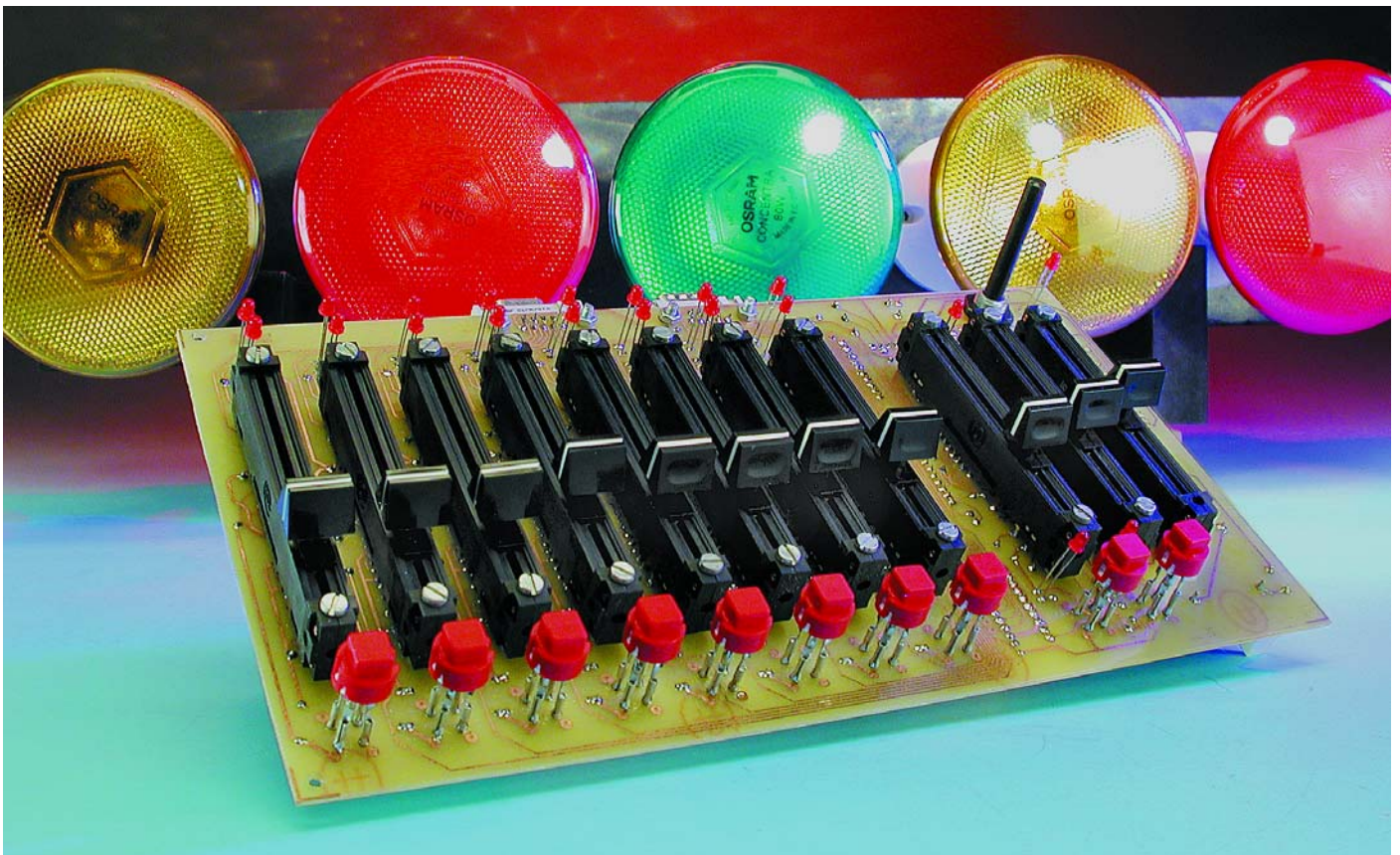


Light Mixer Panel

with eight channels

Design by J. Mack

This highly affordable eight-channel light mixer panel has all of the standard control features plus extras such as a switchable rhythm generator for the control inputs and optical output voltage indicators. Furthermore, the circuit is designed to accommodate extensions, in particular external control signals originating from a PC.



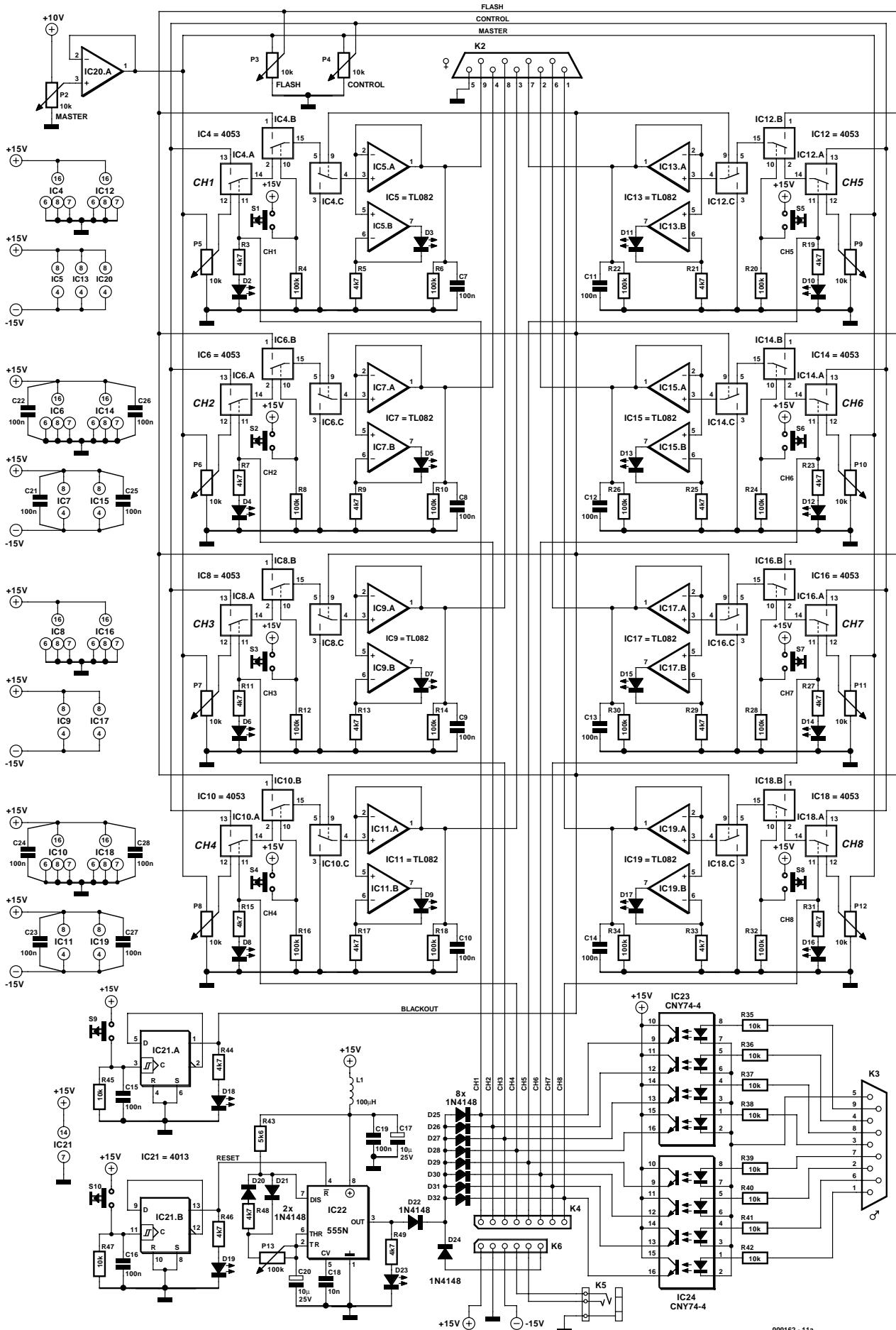
A light mixer panel forms the heart of every lighting installation in a theatre, concert hall or discotheque. The days when lighting control was limited to the legendary 'lights out, spot on' are far behind us. For some time

already, lighting has been regarded as an integral part of the performance. A light mixer panel is just as important for the lighting as an audio mixer panel is for sound. The job of the light mixer panel is to pro-

vide the control signals for the various dimmer/lamp combinations; it acts as a sort of 'user interface' between the responsible technician and the lighting installation.

There are many different models and types of light mixer panels, ranging from extremely

1a



000162 - 11a

small boxes to large digital fader panels that occupy a whole desktop. A basic model for small to medium-sized halls usually consists of four to six channels, each having a (slide) potentiometer, while the common extra features are generally limited to a master fader that governs the total brightness and a 'flash button' for each channel. The latter feature allows the light to be briefly switched on 'full blast', independent of the potentiometer setting.

Features

In the design of this (analogue) light mixer panel, the author has gone somewhat further than just a basic model.

To start with, a choice has been made for eight channels, since that makes the unit quite a bit more versatile than a model with only four or six control outputs. Each channel has its own slide potentiometer, which adjusts the dc voltage on the associated output over the standard range of 0–10 V. This is the input voltage used by voltage-controlled dimmers. Naturally, there is also a master fader to allow all of the channels to be adjusted at the same time.

Flash buttons have also been considered to be a necessary feature, but in order to avoid being bound to 100% brightness, the flash level has also been made adjustable by means of a slide potentiometer. In addition, there is a 'blackout' button. This is more or less the opposite of the flash function, since as the name suggests, all outputs are switched to 0 V when this button is pressed.

Finally, there is also a control input for each channel, which allows a music controller to be connected to the mixer panel. A set of optocouplers and a sub-D connector have also been added to allow a PC to be used as the source of the (control) signals if desired. The level of the 'control' signal can be adjusted using a separate slide potentiometer.

In order to allow something useful to be done with the control function even in the absence of a music signal, a simple rhythm generator has been added to the circuit. This can be enabled at the press of a button, and its rate can be adjusted using a (rotary) potentiometer. The lights connected to the mixer will then flash in sync with the rhythm generator.

What else is there? One important feature is the provision of clear function indicators for the mixer panel. Each channel has its own LED level indicator, with the brightness of the LED corresponding to the level set for the output voltage — not an essential feature, perhaps, but certainly a handy way to monitor what you're doing when working in the dark or with subdued lighting. An additional set of eight LEDs indicate what's happening on the control inputs, while a further pair of LEDs indicate whether the blackout function and the rhythm generator are active.

Circuit description

The schematic diagram shown in **Figure 1** may appear to be horribly complicated at first glance, but it is

actually a lot simpler than it looks, since it is made up of eight identical blocks forming the individual channel faders.

First we'll describe the schematic diagram in general terms from top to bottom, in order to get an idea of where the various parts of the circuit are located. After that we will turn our attention to the details.

At the top left we find the master fader (P2). It is actually located between the power supply and the control portion and allows a voltage in the range of 0–10 V to be set. Moving to the right, we come to the slide potentiometers for the 'flash' and 'control' levels (P3 & P4), with the output connector (K2) located next to them. Signals for all of the controlled channels are present at this connector.

If we shift our glance downwards, we encounter the eight channel faders, which together take up around two thirds of the schematic diagram. To the left of them we see the power supply connections for the ICs, together with the necessary decoupling capacitors.

Below the faders, on the left-hand side, we find the flip-flop for the blackout function (IC21a). This is switched on and off by pressing S9. LED D18 indicates the status of this function. Directly below this we see an identical flip-flop (IC21b), which allows the rhythm generator built around IC22 to be enabled by pressing S10. Here D19 provides the status indication. The frequency of the rhythm generator can be set using P13 and checked using LED 23.

The portion of the schematic diagram at the bottom middle and right brings us to the 'extras' department. With an eye to possible extensions, all control inputs have been made directly accessible via connector K4. Connector K6 has also been added to allow for extension with music control, for example. The ±15-V supply voltage is present on this connector, and a signal can be inserted (or tapped off) via audio jack K5. The 9-way sub-D connector (K9) is intended to be used for connection to the parallel port of a PC, among other things. It can be used for connecting external control signals, which are electrically isolated from the rest of the circuit by optocouplers (IC23 & IC24).

Operation

Now that we have an idea of how the schematic diagram is organised, it's time to take a detailed look at the operation of the circuit. Our attention here will primarily be focused on the channel faders, since the rest of the circuit has already been fairly well described.

Let's start again with the master poten-

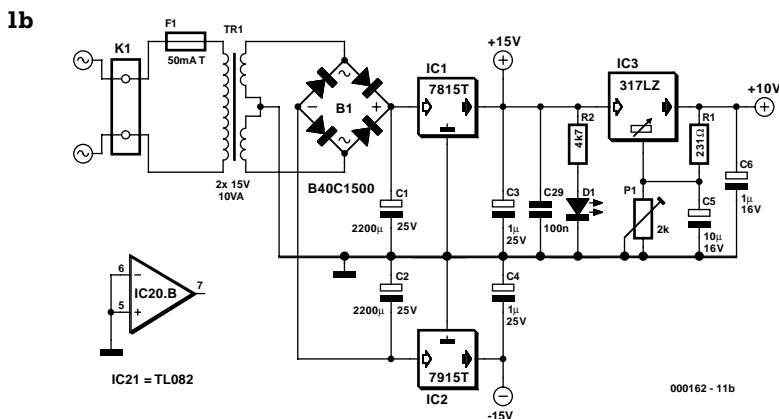
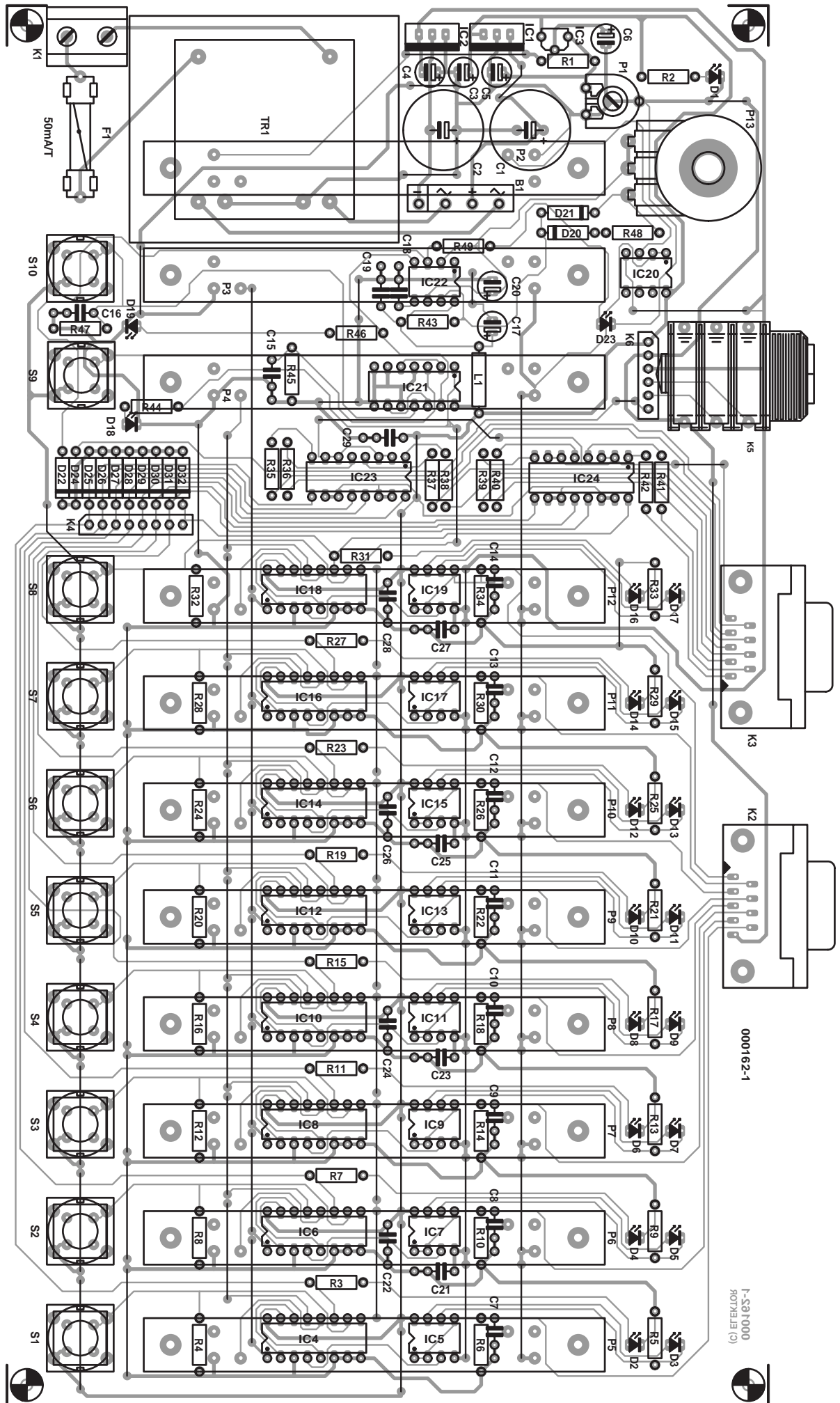


Figure 1. Although the schematic diagram is fairly extensive, it's actually not complicated. The power supply (1b) employs three integrated voltage regulators.



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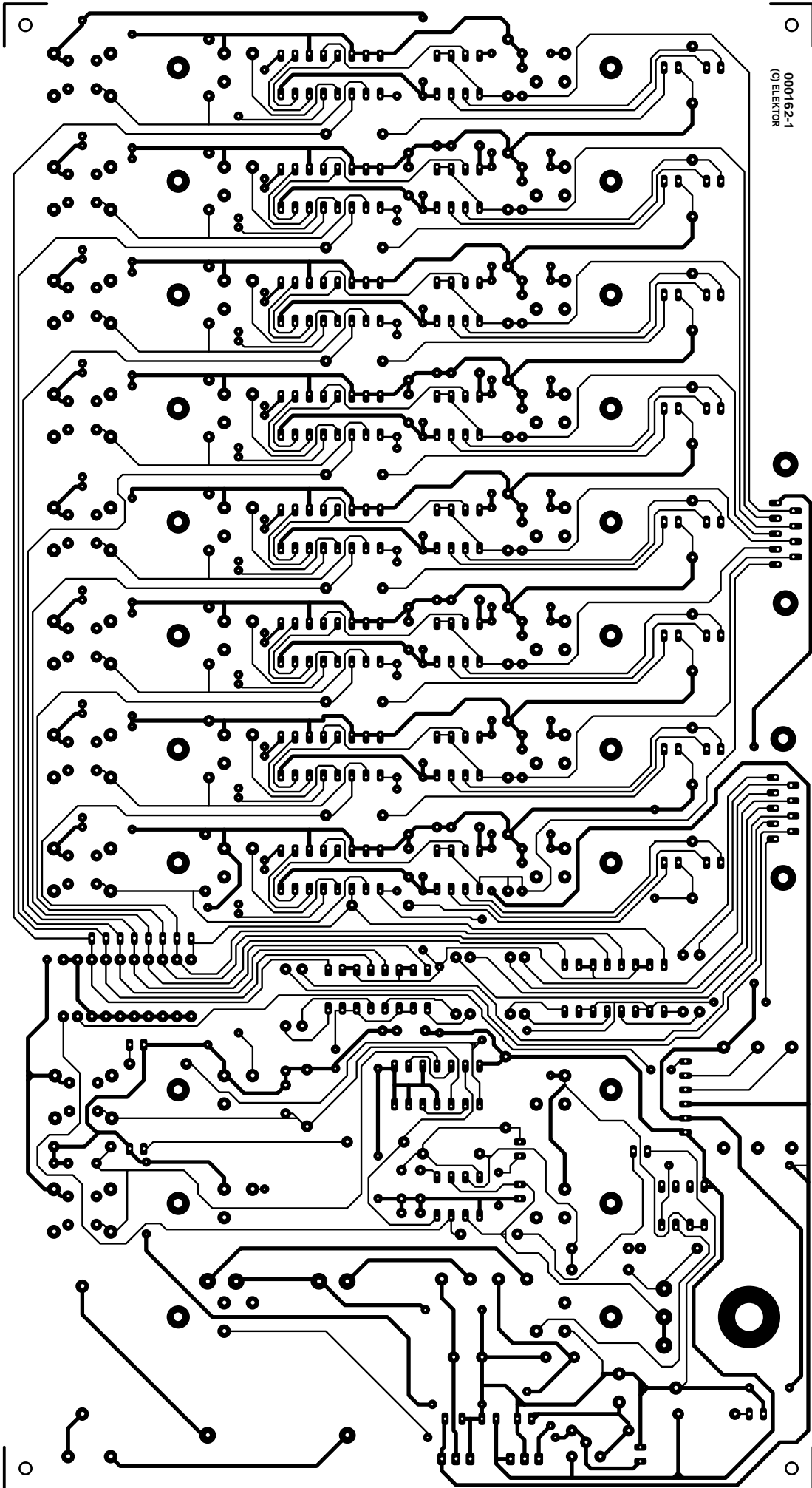


Figure 2. The printed circuit board (available from Readers Services) houses the complete circuit, including the various connectors and the power supply.

COMPONENTS LIST

Resistors:

R1 = 231Ω
 R2,R3,R5,R7,R9,R11,R13,R15,R17R19,R21,R23,R25,R27R29,R31R33,R44,R46,R48,R49 = 4kΩ7
 R4,R6,R8,R10,R12,R14,R16,R18,R20,R22,R24,R26,R28,R30,R32,R34 = 100kΩ
 R43 = 5kΩ6
 R35-R42,R45,R47 = 10kΩ
 P1 = 2kΩ5 preset
 P2..P11 = 10kΩ slide potentiometer, linear law, e.g., Radiohm (Conrad Electronics # 44 14 49-60)
 P13 = 100kΩ potentiometer

Capacitors:

C1,C2 = 2200μF 25V radial
 C3,C4 = 1μF 25V radial
 C5,C17,C20 = 10μF 25V radial
 C6 = 1μF 16V radial
 C7-C16,C19,C21-C29 = 100nF
 C18 = 10nF

Inductors:

L1 = 100μH

Semiconductors:

B1 = B40C1500 bridge rectifier (40V piv, 1500mA continuous)
 D1-D18 = LED, red, 3mm, low current
 D2-D17,D19,D23 = LED, green, 3mm, low current
 D20,D21,D22,D24-D32 = 1N4148
 IC1 = 7815
 IC2 = 7915
 IC3 = LM317LZ
 IC5,IC7,IC9,IC11,IC13,IC15,IC17,IC19,IC20 = TL082
 IC4,IC6,IC8,IC10,IC12,IC14,IC16,IC18 = 4053
 IC21 = 4013
 IC22 = 555
 IC23,IC24 = CNY17-4

Miscellaneous:

S1-S10 = pushbutton, one make contact, e.g., D6-R (ITT Schadow)
 F1 = fuse 50mAT (time lag) with PCB mount holder
 TR1 = mains transformer, PCB mount, 2x15V/10VA, e.g., ERA type EL48/16,8
 K1 = 2-way PCB terminal block, lead pitch 7.5 mm
 K2 = 9-way sub-D socket (female), PCB mount, angled pins
 K3 = 9-way sub-D socket (male), PCB mount, angled pins
 K4 = 8-way SIL pinheader
 K5 = stereo 6.35 mm jack socket, PCB mount
 K6 = 6-way SIL pinheader
 Enclosure: e.g. Teko 52 39 84-60 (310 × 170 mm)
 PCB, order code **000162-1** (see Readers Services pages)

tiometer P2. This is connected to a 10-V voltage provided by the power supply, so a voltage ranging from 0 V to 10 V is available on its wiper. In order to avoid undesired feedback, the wiper voltage is first buffered by IC20a, following which it serves as the input voltage for the 'flash' and 'control' adjustment potentiometers P3 and P4 (which act on all channels equally) and the individual channel potentiometers (P5–P12).

Since all eight-channel fader circuits are identical, we can limit our discussion to a description of the operation of the channel 1 fader, which consists of the circuitry around IC4 and IC5.

The desired level is set using P5. On its way to the output, the voltage from the wiper of the potentiometer passes through three electronic switches connected in series (IC4a, IC4b and IC4c). As long as no special function has been enabled, the signal simply passes through all three switches, following which the voltage is buffered by IC5a and fed to output connector K2. However, this situation changes if any of the three electronic switches is activated. Whenever a voltage is placed on the control input (either externally via K4 or by switching on the rhythm generator), switch IC4a is activated via the CH1 line. This breaks the connection to P5 and causes the 'control' signal to be passed through to the output, with the 'control' potentiometer P4 setting the output level. In this situation, LED D2 is illuminated.

If the 'flash' button (S1) is pressed, IC4b switches and the voltage level set by the 'flash' fader (P3) is passed through to the output.

The 'blackout' function acts on all channels simultaneously. Whenever this function is activated using pushbutton S9, IC4c switches. This breaks the through path and causes the input of buffer IC5a to be connected to ground. The output is thus always 0 V, regardless of any other settings.

An additional opamp (IC5b) has been added to indicate the output voltage. It acts as a voltage-to-current converter that changes the 0-10 V control voltage into a practically linear variation in the brightness of LED D3 (which, like all of the other

LEDs, must be a low-current type).

A low-pass filter in the form of C7 has been added to the output to eliminate possible cable effects. Resistor R6 acts as an open-circuit load.

Actually, we have now described the complete circuit. The only thing we still owe you is a few words of explanation regarding the rhythm generator, even though there's not really very much to say about it. As can be seen, it is built around a 555 timer (IC22) wired as a 'textbook' multivibrator, with the frequency being adjustable over the range of around 1-10 Hz using P13. Diodes D29 and D21 ensure that the duty cycle is 50%, and the combination of L1 and C17 prevents the +15-V line from being contaminated by noise pulses originating from the 555, since such pulses could have a detrimental effect on the analogue portion of the circuit. Via D22 (which prevents feedback), the signal output signal is applied to a diode network consisting of D25-D32, by means of which all control inputs can be driven simultaneously.

Power supply

The mixer panel needs three different power supply voltages: +15 V for the electronic switches, ±15 V (symmetrical) for the opamps and, last but not least, exactly +10 V for the input voltage to the faders.

As can be seen from the schematic diagram in **Figure 1b**, the power supply has been kept as simple as possible. The well-known combination of a transformer, a bridge rectifier module and a pair of electrolytic capacitors provides unregulated dc voltages of ±19–20 V. These are adequate input voltages for the 7815T and 7915T three-terminal regulators (IC1 and IC2), which deliver nicely stabilised voltages of +15 V and –15 V.

To generate a stable +10 V voltage, we have chosen an LM317LZ adjustable voltage regulator (IC3). In order to keep the dissipation of this regulator as low as possible, its input is taken from the +15-V line. The output voltage can be adjusted using P1. In order to compensate for possible losses, the voltage is best measured at the output connector

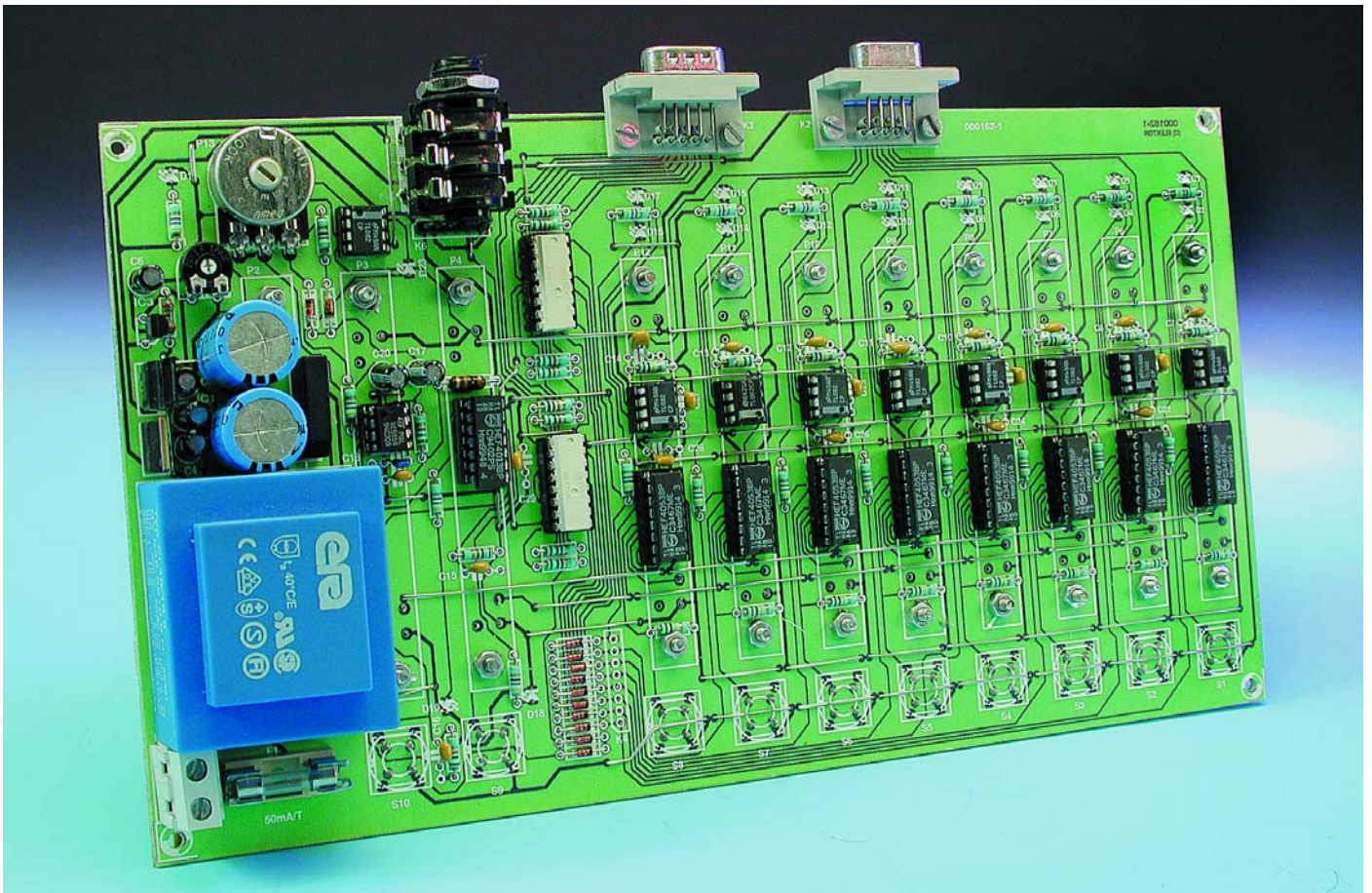


Figure 3. So, which is the top side and which is the bottom side? In any case, this is the component side.

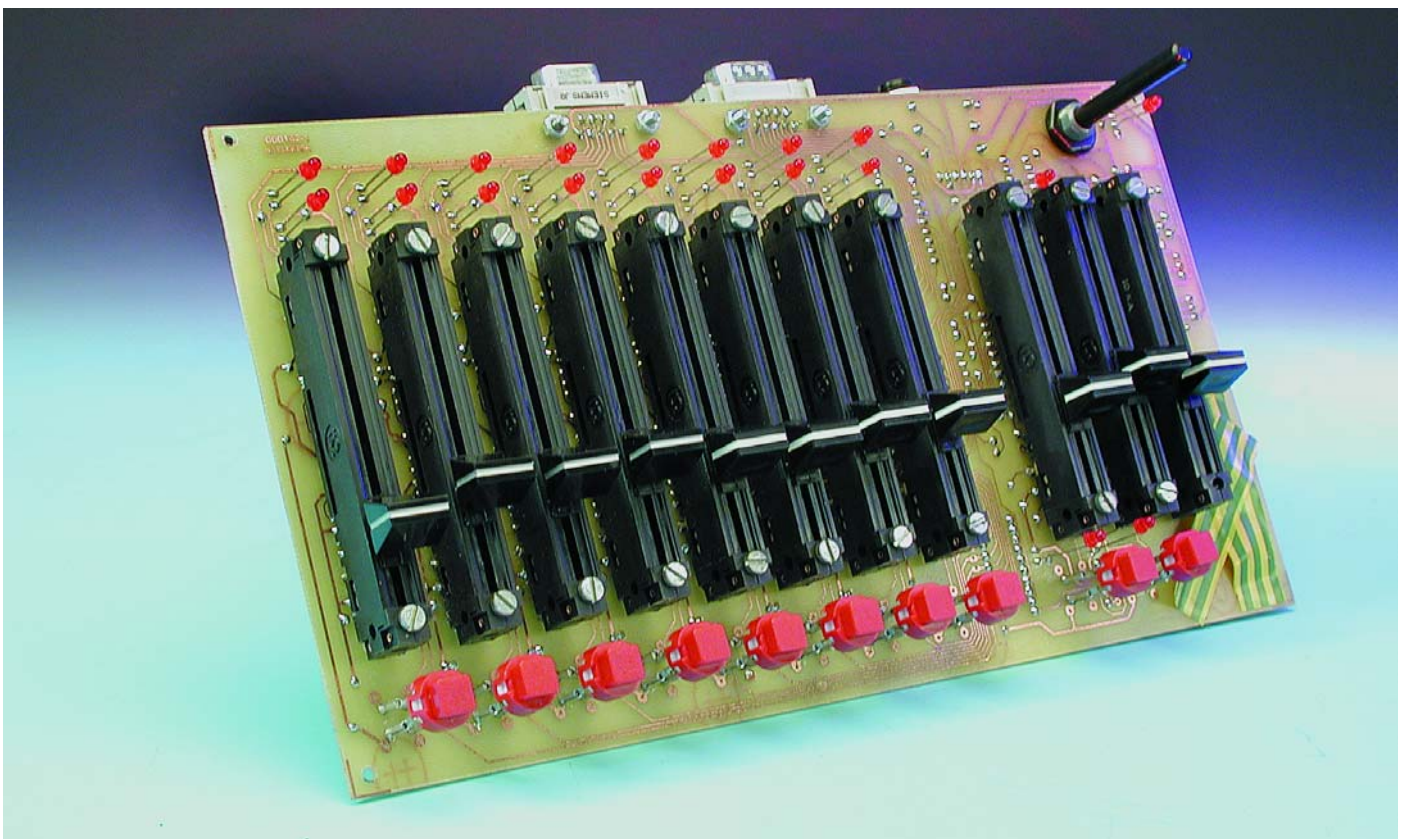


Figure 4. The switches, slide potentiometers and LEDs are fitted to the copper side.

(K2). For example, you can connect a voltmeter between pin 5 (ground) and pin 9 (channel 1 output), rotate P2 and P5 fully clockwise and then adjust P1 for a meter reading of exactly 10 V.

Now that's soldering!

The copper track and component layouts of the printed circuit board that has been designed for the mixer panel are shown in **Figure 2**. This circuit board houses the complete circuit, including the power supply, and since it has to provide space for eleven slide potentiometers, it is not exactly small.

Before saying anything about the construction of the circuit board, it's worth mentioning that a **single-sided** design has been used for the circuit board in order to reduce its cost. However, this means that quite a few wire bridges are required — no fewer than 79 in total! In combination with the fact that components must be fitted to both the 'normal' component side and the solder side of the board, this means that building this circuit board is not such a simple project. If you are not an experienced hobbyist, we thus recommend that you don't undertake this project unless you can be sure of help from an 'old hand'.

As noted, both sides of the circuit board must be 'assembled'. Everything is arranged such that all of the control elements, consisting of the slide potentiometers, the LEDs (including the on/off LED D1) and the pushbuttons, are fitted to the copper side and the remaining components are fitted to the regular component side.

You should start with the component side of the circuit board, and it is recommended to first fit all the wire bridges. Make a copy of Figure 2 and mark off each wire bridge as it is fitted, in order to avoid forgetting any of them. Pay careful attention, since a few of them are well hidden, such as those on either side of C2, the ones next to IC1 & IC2 and a very short one between K6 and P4. There are also two bridges that must be bent diagonally, next to IC21 and at S8.

After this, you can start to fit the other components, beginning as usual with the lowest components (resistors, diodes and IC sockets), followed by the higher components (electrolytic capacitors, voltage regulators, fuse holder and transformer).

The various connectors can also be fitted directly to the circuit board, with K2, K3 and K5 being intentionally located on the same side of the board, since this is much more practical when it comes time to fit the board into an enclosure.

Once all the components on the compo-

nent side have been fitted according to instructions, you can turn the board over and begin the difficult task of fitting the slide potentiometers one after another. Each potentiometer must be attached to the board with two M3 screws. Use distance rings (preferably plastic) to ensure that a clearance of a few millimetres is maintained between the bottom edge and the surface of the circuit board; otherwise you will not be able to reach the connection tabs with the soldering iron. For the same reason, you should preferably work from right to the left when fitting the slide potentiometers, which means first fitting P2, P3 and P4, followed by P12, P11 and so on.

In order to ensure that switches S1–S10 remain accessible after the board is mounted in an enclosure, they must be fitted a certain distance above the circuit board. With the prototype, this was achieved by soldering four pins to the circuit board for each switch and then soldering four pin sockets to the terminals of each switch. Naturally, other solutions are also possible.

The indicator LEDs must also be fitted with 'long legs' to ensure that they are visible after the board has been fitted into its enclosure. Fortunately, this is not particularly difficult, since LEDs come with long leads.

In the photos of the prototype circuit board shown in **Figures 3 & 4**, you can easily see the various construction details. It certainly won't do any harm to carefully compare your own board with the photos.

Since the light mixer panel does not include any alignment elements other than P1 in the power supply, strictly speaking there is no reason to expect it not to work right away. After the final component has been fitted to the board, there is thus a great temptation to immediately connect a mains cable to K1 and plug it in to an outlet. Still, there are two reasons why it is a good idea to exercise a bit of restraint. In the first place, due to the large number of wire bridges a very thorough visual inspection of the assembled board is in order. In the second place, you should firmly bear in mind that mains voltage is present on part of the circuit board, and it only needs a

moment of inattention to come in contact with this voltage. Be very careful what you do!

Fitting the enclosure

You are completely free in choosing a suitable enclosure, although naturally a console-type enclosure with a sloping top surface is the most suitable for this type of circuit.

For the prototype, we selected such an enclosure, namely the Teko model 52 39 84-60. With a width of 310 mm and a depth of 170 mm, this enclosure provides ample space for the circuit board, so you don't have to make the most of every millimetre. The circuit board can be attached to the inside of the sloping top face of the enclosure using a set of (plastic) standoffs. Suitable openings must be made for the potentiometer knobs, pushbuttons and LEDs. You should make a neat job of this, since an attractive result is naturally desirable.

Pay attention to the electrical safety of the overall construction. Provide a solid mains switch and mains entrance connector and a well-insulated connection between the mains entrance connector and connector K1. Furthermore, ensure that there is adequate clearance between the metallic parts of the enclosure and the fuse holder for F1 and the circuit board traces interconnecting K1, F1 and TR1! Finally, attach a copy of the type label accompanying this article to the rear of the enclosure, with a clear indication of the supply voltage and fuse value, in order to comply with safety regulations.

You can connect the various lights and other equipment to be controlled to connector K2. Some commercially available spotlights and other lighting equipment have built-in dimmers with 0–10 V control inputs. If you want to fit such dimmers to your own equipment, have a bit of patience — we will shortly be publishing a design for such a circuit in *Elektor Electronics*.

(000162-1)

Infrared Transceiver for the PC

part 2: RC5 transmitter and data link

Design by B. Kainka

<http://home.t-online.de/home/B.Kainka>

The circuit of the IR transceiver was introduced in the previous issue, along with sample applications for use as an RC5 receiver for the PC. In this, the second and last instalment, we examine the operation of the transmitter. In addition to PC remote control uses, we describe how it can be used for infrared data transmission between two PCs, with several examples to make everything clear.

It can be useful to not only receive RC5 signals but also generate them, for example to automatically start a video recorder or switch on the television set at a certain time. In principle, every control function of an RC5 IR remote control unit can be automatically executed from the PC.

RC5 transmitter

Normal remote control units use short pulses with duty cycles of less than 10% together with currents of up to around 1 A. As a result, the signal strength at the receiver is adequate, even if it only reaches the receiver via reflections from white walls. Such current levels cannot be attained using the serial port, but this is offset by the fact that there is enough voltage to allow several LEDs to be operated in series. The IR transmitter thus uses two transmitter diodes with a pulse current of around 200–300 mA, thereby generating short but powerful light pulses that ensure a range of several metres.

The first transmitter program (**Listing 1**) uses the procedure RC5tx to generate the control pulses on the TxD line. The three data blocks Ctr, Dtr and Dat are transferred on being called. The user can enter the device address and button code in the appropriate text windows. The control code is repeated a total of five times, which means that the data are sent for an interval of 0.5 s.

The procedure RC5tx works in the real-time mode. It first generates the start sequence and then processes the remaining control data. Each bit is transferred to the 0 position by a sort of shift operation (implemented by dividing by 2), which isolates it from the rest of the bits. Procedure RC5tx is driven by these individual bits.

The REALTIME DLL function in PORT.DLL causes the user program

to be assigned a high priority. As a result, the critical procedure cannot be interrupted by other Windows processes. This technique minimises the effects of the general problem that Windows programs have poor real-time capability. Interrupts by mouse events, for example, are suspended for the duration of an RC5 pulse. The line REALTIME false at the end of the procedure is also important, since it restores the normal state.

The RC5 transmitter can be used for automatic control. For this, you need to know the device address (see **Table 1**) and the button code (see **Table 2**). Given this information, the transmitter can generate a suitable command at a particular time or in response to some event.

For instance, suppose you want to switch your video recorder to 'record' at a particular time and stop

Editor's note:

Due to their size, Listings 4 and 5 could not be reproduced in this article. All listings mentioned in this article are on the Free Downloads page of the Elektor Electronics website, together with the program files, under item number **010052-2-11**.

it at some later time. The program VCtimer (**Listing 2**) allows you to enter the switching times. It also provides two switch buttons for direct (manual) operation.

Another example program is related to the 'zapper finger' syndrome. Given the large number of channels available nowadays, this dreaded ailment is as much a threat to the avid television viewer as tennis elbow is to sports adepts or mouse arm to computer fanatics. Using the small program shown in Listing 3, the RC5 transceiver can help remedy this problem.

Serial data transmission

Up to now, we have only discussed infrared signals that comply with the RC5 standard. Now we would like to consider transmitting serial data between PCs using modulated infrared light. For instance, the IR transceiver can be used in a computer room for 'chatting', which means sending text messages back and forth. A few preliminary experiments can help clarify what is

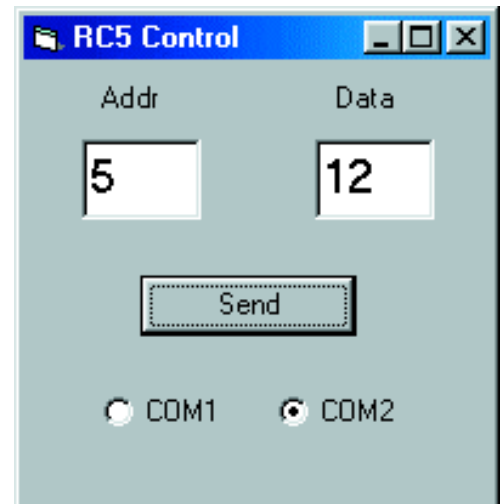


Figure 1. A test program for RC5 signals.

Listing 1

RC5 transmitter program RC5TxTast.vbp)

```

Dim Control
Private Sub Form_Load()
    i = OPENCOM("COM2,1200,N,8,1")
    If i = 0 Then
        i = OPENCOM("COM1,1200,N,8,1")
        Option1.Value = True
    End If
    If i = 0 Then MsgBox ("COM Interface Error")
    RTS 1
    DTR 1
    Control = 1
    Chan = 0
End Sub
Private Sub Form_Unload(Cancel As Integer)
    CLOSECOM
End Sub
Private Sub Option1_Click()
    i = OPENCOM("COM1,1200,N,8,1")
    If i = 0 Then MsgBox ("COM1 not available")
    RTS 1
    DTR 1
End Sub

Private Sub Option2_Click()
    i = OPENCOM("COM2,1200,N,8,1")
    If i = 0 Then MsgBox ("COM2 not available")
    RTS 1
    DTR 1
End Sub

Private Sub TxBit(ONOff!)
    If ONOff = 1 Then
        TXD 0
        DELAYUS 888
        TXD 1
        DELAYUS 888
    Else:
        TXD 1
        DELAYUS 888
        TXD 0
        DELAYUS 888
    End If
End Sub

Private Sub RC5tx(Ctr, Adr, Dat)
    BitTime = 888
    REALTIME True
    TIMEINITUS
    TXD 1
    While TIMEREADUS < (BitTime * 1)
    Wend
    TXD 0
    While TIMEREADUS < (BitTime * 2)
    Wend
    TXD 1
    While TIMEREADUS < (BitTime * 3)
    Wend
    TxBit (Ctr)
    BitVal = 16
    For n = 1 To 5
        b = Int((Adr And BitVal) / BitVal)
        TxBit (b)
        BitVal = BitVal / 2
    Next n
    BitVal = 32
    For n = 1 To 6
        b = Int((Dat And BitVal) / BitVal)
        TxBit (b)
        BitVal = BitVal / 2
    Next n
    TXD 0
    REALTIME False
End Sub

Private Sub Chan1()
    For n = 1 To 3
        RC5tx Control, 5, 1
        DELAY 100
    Next n
    If Control = 1 Then
        Control = 0
    Else
        Control = 1
    End If
    Chan = 1
End Sub

Private Sub Command1_Click()
    Adr = Val(Text1.Text)
    Dat = Val(Text2.Text)
    For n = 1 To 5
        RC5tx Control, Adr, Dat
        DELAY 100
    Next n
    If Control = 1 Then
        Control = 0
    Else
        Control = 1
    End If
End Sub

```

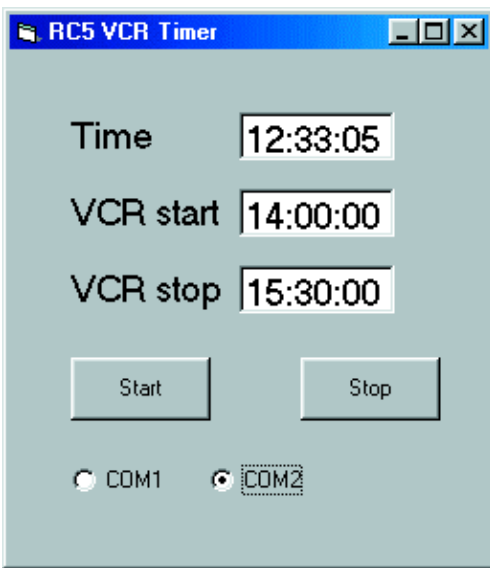


Figure 2. The video timer in operation.

involved in transmitting a serial data signal.

In the first of our experiments, we will use the serial interface rather differently than how its designers originally intended it to be used.

This interface was actually intended to be used to support data transmission via a modem. Signal names such as 'Ring Indicator' and 'Data Terminal Ready' still give evidence of this original purpose. In the course of time, the serial interface has repeatedly proven to be useful in other areas. The greatest extension to its range of use came when the mouse was connected to the RS232 port. When the RS232 standard was originally introduced, it's unlikely that anyone ever thought that some day devices having their own microcontrollers would work attached to the serial port and even draw their operating power from this port. On the other hand, (many) years later this idea was one of the



Figure 3. The automatic zapper program.

basic assumptions in the design of a new type of serial port called USB.

However, let's get back to our old friend, the RS232 port, which was designed to transfer data via two lines called RxD and TxD. For this purpose, the data are converted to a serial bit stream by the interface hardware and sent on their way. At the receiver, a second interface converts the serial data stream back into parallel data bytes that can be further processed by the PC. All of this represents a successful division of labour between hardware and software. Sending a character via the serial interface takes a lot of time in comparison with other operations; such a process is measured in terms of milliseconds, rather than the microseconds that we otherwise have to deal with. Consequently, the interface in the PC contains a special hardware component called a Universal Asynchronous Receiver / Transmitter (UART), which is a sort of telegraph office for the PC. Messages to be sent are 'handed over' to the UART, which transmits them on its own. In the opposite direction, the UART independently receives messages and deposits them in a 'cubby-hole', where they can be retrieved by a program in a single action.

A typical task for the serial interface is to transfer data from one PC to a second PC. This can be any desired kind of data, such as text, programs, images and so on. Ultimately, what is sent is individual bytes, which are blocks of eight bits. The interface can also be configured to have each character consist of only seven bits or even only five bits. The 5-bit character is a remnant of the days of the Teletype machine, which was a sort of long-distance typewriter that transmitted these characters serially over the telephone line in a manner that is in principle the same as that used by an RS232 interface, only much slower and a lot louder.

A zero-modem link

Our first experiment involves connecting two PCs so that they can exchange characters with each other. For this, we need a crossover connection for the TxD and RxD

Table 1

Device addresses

Device address	Device
0	television
2	teletext
5	video recorder
7	experimental
16	preamplifier
17	receiver/tuner
18	cassette recorder
19	experimental

Table 2

A few important button codes for the television set and VCR

Code	Function
0-9	numerical buttons 0-9
12	standby
13	mute
16	volume +
17	volume -
32	channel +
33	channel -
48	video pause
53	video play, television sound on
54	video stop
55	video record

lines. Such a connection is called a 'zero modem' (or 'null modem') cable, since no modem is used. Although modems and telephone lines are needed to send data over long distances, we can manage with just a normal cable for distances of up to a few metres (see **Figure 4b**).

For the serial transfer of data, PORT.DLL (which we already know from the first instalment) provides us with the procedures SENDBYTE and READBYTE. A further prerequisite for a successful data exchange is that the interfaces of both parties are initialised with the same parameters. The command

```
OPENCOM("COM2,1200,N,8,1")
```

causes both PCs to agree to use a transmission data rate of 1200 bits per second (1200 baud), no parity

Listing 2

The most important video timer procedures (VCRtimer.vbp)

```
Private Sub RecordOn()
  For n = 1 To 10
    RC5tx Control, 5, 55
    DELAY 100
  Next n
  If Control = 1 Then
    Control = 0
  Else
    Control = 1
  End If
  ONOff = 1
End Sub

Private Sub RecordOff()
  For n = 1 To 10
    RC5tx Control, 5, 54
    DELAY 100
  Next n
  If Control = 1 Then
    Control = 0
  Else
    Control = 1
  End If
  ONOff = 0
End Sub

Private Sub Timer1_Timer()
  Text1.Text = Time$
  If (Time$ > Text2.Text) And (Time$ < Text3.Text) And (ONOff = 0) Then RecordOn
  If (Time$ > Text3.Text) And (ONOff = 1) Then RecordOff
End Sub

Private Sub Command1_Click()
  RecordOn
End Sub

Private Sub Command2_Click()
  RecordOff
End Sub
```

bit, 8 bits per character and one stop bit. These parameters are discussed in more detail later on.

Figure 5 shows the oscillogram of a serial data signal for a '1' character sent at a data rate of 1200 baud. At this data rate, the time required to transmit each bit is $(1 \text{ s} \div 1200) = 833 \mu\text{s}$. At the left-hand side of the figure, you can see the start bit with a length of 0.833 ms. Only the first bit following the start bit, with a Low level, can be clearly distinguished from the rest of the data stream. You can see that a '1' is being sent if you know that the least significant bit is sent first and the bit levels are inverted. The first bit is followed by seven more High states, each of which is also 0.833 ms long. They represent the seven subsequent zero bits. The byte that is read is thus 00000001 = 1. Following the

last data bit there is a stop bit, even though this is not obvious in the figure. The stop bit has a Low level and is simply the minimum allowed interval before the start of any subsequent character. The actual inter-

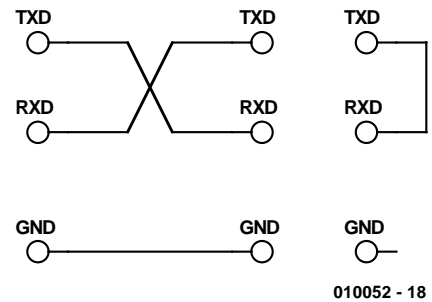


Figure 4. A null-modem connection (a) and a test using a single interface (b).

val can be as long as desired.

The process that is required for receiving a character in the UART follows directly from the structure of the serial data character. The interface IC must continuously watch its input line until a Low-to-High transition appears, which marks the beginning of a start bit. When this happens, the UART must start an internal timer and wait for the end of the start bit. After this, it must wait again for half of a bit period and then sample the signal level. This reads the first bit. Each following bit is read exactly one bit period later. This means that the reading process is theoretically completed half a bit period before reaching the stop bit.

As you can see, data frame synchronisation occurs only once, namely at the beginning of the start bit. All other time frames must be precisely known and must be maintained relatively exactly. If the transmitter operates at a different baud rate from the receiver, read errors are an unavoidable consequence. Modern UARTs, by the way, do not limit themselves to a single sampling of each data bit, but instead take three samples (for example) around the middle of each bit. Normally, all three samples should have the

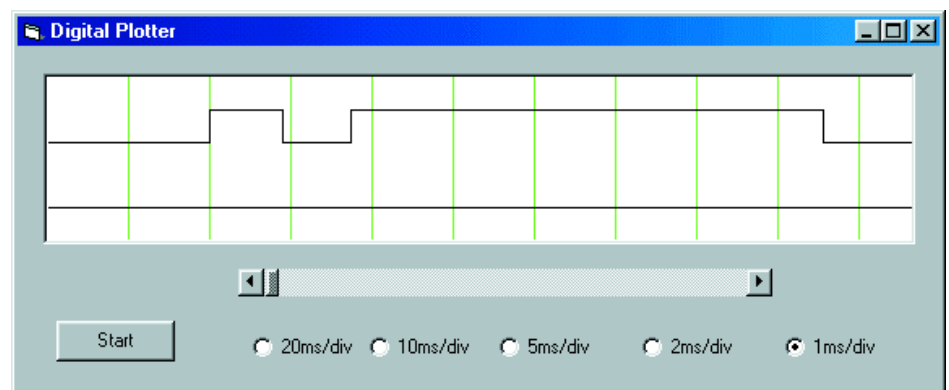


Figure 5. Transmitting a '1' byte at 1200 baud.

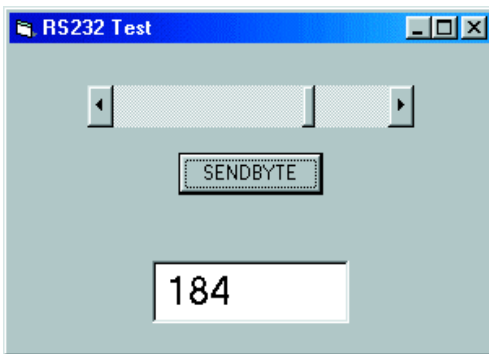


Figure 6. Transmitting and receiving individual bytes.

same value, but if they do not (as the result of disturbances), a majority decision is taken. This allows greater data security to be achieved. Our first example program for data transmission (**Listing 4**; see the Downloads box) transfers only one byte for each command. Here the position of a slider, expressed as a numerical value between 0 and 255, is sent to the other PC. In the other direction, each received byte is displayed in a text window (see **Figure 6**). With an additional program (**Listing 5**) and its description, which are contained in the download file, you have a universal terminal emulator program that is particularly suitable for use in your own experiments and with applications you have developed yourself.

Optical data transmission

For a simple experiment with optical data transmission, all you need is an IR transmitter diode and a phototransistor (see **Figure 7**). The transmitter diode can be connected directly to the TxD line, but the receiver needs an auxiliary supply voltage, which can for example be obtained from the DTR line if it is set High. As soon as light from the transmitter diode falls on the phototransistor, current flows and TxD is pulled High. In this way, the original signal applied to the TxD line is reproduced at the RxD line.

This experiment can be performed using the terminal emulator program from download file, for example. Here it makes no difference whether data are being exchanged between two PCs or the experiment is performed using only one PC feeding the data back to itself, similar to what is shown in **Figure 4b**. All you have to remember is to set the DTR line High for the receiver.

This simple data transfer arrangement has only a small range (a few millimetres). However, a fibre-optic cable can be used to extend

Listing 3

Automatic channel changer (Zapper.vbp)

```
Private Sub Chan1()
  For n = 1 To 3
    RC5tx Control, 5, 1
    DELAY 100
  Next n
  If Control = 1 Then
    Control = 0
  Else
    Control = 1
  End If
  Chan = 1
End Sub

Private Sub ChanPlus()
  For n = 1 To 3
    RC5tx Control, 5, 32
    DELAY 100
  Next n
  If Control = 1 Then
    Control = 0
  Else
    Control = 1
  End If
  Chan = Chan + 1
End Sub

Private Sub Timer1_Timer()
  If Chan = 30 Then Chan1
  If (Chan > 0) And (Chan < 30) Then ChanPlus
  If Chan = 0 Then Chan1
  Text1.Text = "Chan " + Str$(Chan)
End Sub

Private Sub Command1_Click()
  Timer1.Enabled = True
End Sub

Private Sub Command2_Click()
  Timer1.Enabled = False
End Sub
```

the range to several metres. There are special optical transmitters and receivers that can be interconnected using standard fibre-optic cables. One example is the combination of the SFH250 transmitter diode and SFH350 phototransistor, which have openings to receive fibre-optic cables. With such a fibre-optic link, data can be sent at any desired rate up to 115,200 baud.

If we want to achieve even greater range and also do without cables, we can use modulated signals, as is common with IR remote control units. In this case, we can use the IR transceiver described in the first instalment for both the transmitter and the receiver. The received signal is inverted by a transistor stage in the transceiver, so it has a Low level in the quiescent state. It must be noted that the opti-

cal receivers that have been specially developed for IR remote control (such as the SFH506 and TSOP1836) are only suitable for conveying RS232 signals if a few special features are taken into account. These features are based on the fact that the greatest possible range is required for the reception of IR remote control signals, even with high levels of ambient illumination. This is achieved by using a band-pass filter and automatic gain control (AGC) (see **Figure 8**), which we also find in radio and television receivers. Such a control circuit always has a time constant that must be optimised for the intended application.

On account of the value of this time constant, which has been dimensioned for IR remote control signals, the modulated transmitter

Downloads

At the *Elektor Electronics* website (www.elektor-electronics.co.uk), in the Free Downloads list for this month's issue you will find a zip file for this article under number **010052-2-11**. Once you have downloaded and unzipped this file, you will have a folder containing all the program files and a file containing all the program listings and a description of the terminal emulator program. These files can also be downloaded from the author's home page at <http://home.t-online.de/home/B.Kainka>.

pulses must be neither too short nor too long. These devices are thus not suitable for use with continuous signals, since they cause the automatic gain control to overreact.

The data sheet lists the following restrictions:

- A burst must consist of at least 6 pulses, which means that it must be at least 167 μ s long at 36 kHz.
- A pause of at least 9 pulse widths (250 μ s) is required following each burst.

A pause lasting at least 15 ms must occur after at most 90 ms.

Particularly if the last condition is to be satisfied, the required 15-ms pauses must be ensured by the stop bits of the serial data signal. The worst-case situation occurs when a continuous series of '0' bits is being transmitted, since the start bit and all eight data bits are then 'on' and only the stop bit forms a pause. Strictly speaking, this means that the maximum allowable baud rate

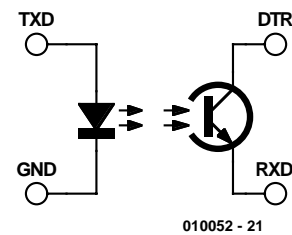


Figure 7. The simplest form of optical transmission path consists of nothing more than an LED and a phototransistor.

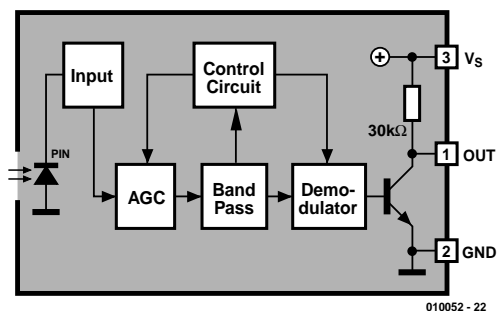


Figure 8. Block diagram of the TSOP18xx IR receiver.

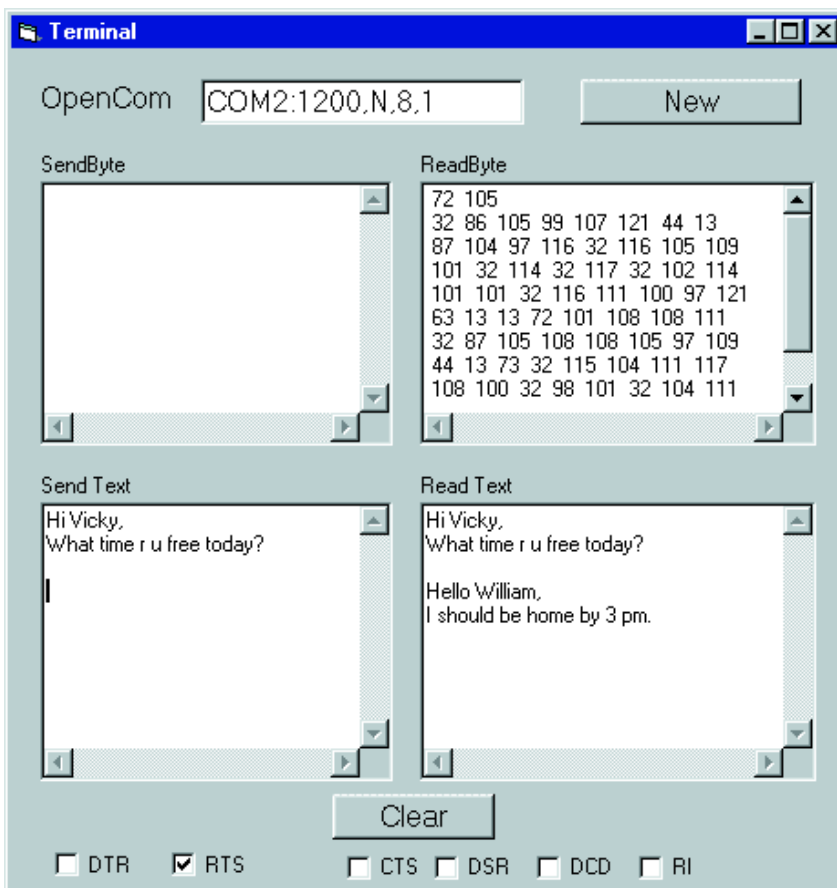


Figure 9. An infrared chatroom set up using the universal terminal emulator program.

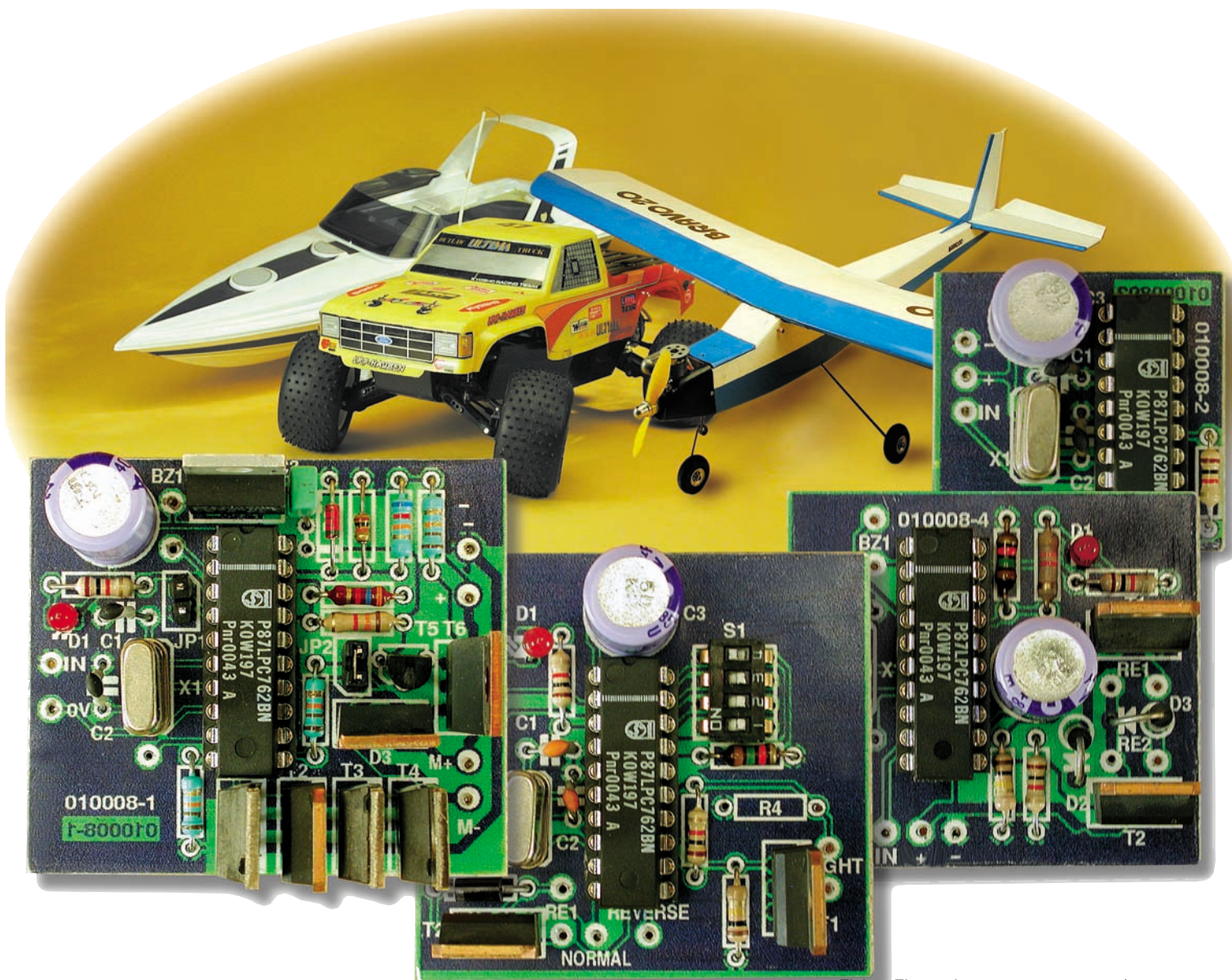
for a continuous data stream is 130 baud if two stop bits are used. The situation is more favourable if the data bytes appear sporadically (which is the case with data from a keyboard while someone is typing, for example), rather than in a continuous stream. On average, the random mixture of different bytes in an uninterrupted data stream also provides an adequate number of pauses. Using the terminal emulator program from the download file, you can easily experiment with the various transmission parameters (Figure 9). A thorough test using manually typed characters showed that all baud rates between 50 and 2400 baud resulted in error-free transmissions. At 2400 baud, the shortest interval for the pulses and pauses is 417 μ s, which falls within the receiver specifications. The maximum usable distance between the transmitter and the receiver depends primarily on the transmitter diodes used and their currents. The data sheet for the TSOP18xx specifies a typical range of 35 metres, which certainly isn't bad.

(010052-2)

Multi-purpose IC for modellers (I)

Fourteen functions using four PCBs

Here is something the modelling world has never seen before: a single chip that can be fitted to four different PCBs and which offers 14 functions which can be combined in various configurations as if it were a standard component. This unique device is a readily available microcontroller, and it derives its versatility from the software with which it is programmed.



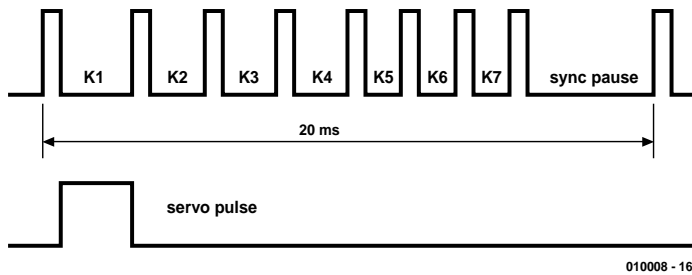


Figure 1. Typical PPM signal.

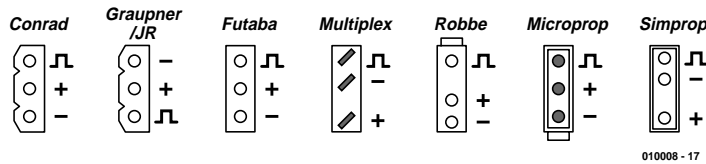


Figure 2. Pinouts of the commonest types of servo connectors.

Model builders continuously demand more and more functions from their systems, which can be implemented either by using expensive computerised transmitters or by adding auxiliary circuits at the receiver end. These circuits are almost always designed for one specific function, with the result that often more hardware is used than strictly necessary. In models where weight is critical this can mean that some of the additional functions have to be sacrificed. Also, there is always only a limited amount of space at the receiver end for circuit boards, and the less electronic hardware there is in the model, the more reliable the whole system. Moreover, by implementing as many different functions as possible in a microcontroller, the user can select just those required for his particular application.

Since it makes little sense to implement all the functions on a single circuit board, four different variants have been developed, which can be populated as the user wishes. A socket is recommended for the microcontroller, so that it can be moved as necessary from one board to another. The system employs a 'universal chip' which provides all the most frequently required functions. An overview of the individual functions of the multipurpose device (which is based on a Philips microcontroller type 87LPC762) appears in **Table 1**.

How these functions can be combined will be explained later.

SERVO CONTROL

Before we look at the functions and circuits in detail, a few words on the fundamentals of servo control in models.

Modern radio control systems use either PCM (pulse code modulation) or PPM (pulse position modulation). In a PCM system the position of each joystick and switch on the transmitter is converted into a value in the range 1 to 255.

These digital values are sent out from the transmitter one after another as 8 bit quantities. Along with the control information so-called 'check bits' are included in the PCM signal. With the aid of these check bits the microcontroller in the receiver can verify the correctness of the incoming data stream and so ensure that only reasonable commands are sent to the servos.

If an erroneous signal is detected, the servos retain the most recently received correct settings until a new code is correctly received. PCM systems thus offer particularly reliable data transfer. Partly because of the limited bandwidth and the large quantity of data to be transmitted, the repetition rate of the transmitted commands is lower than with conventional PPM systems. One must always bear in mind that the radio signal undergoes analogue processing before the software in the microcontroller has a chance to process it digitally. A distorted signal, which is beyond the PCM system software's

Table 1

Function summary:

- Model finder (a buzzer is activated when the transmitted signal is absent or severely distorted)
- Receiver voltage monitoring with indicator
- Motor voltage monitoring with indicator
- 2-channel switch
- Hot glow (switches on glow plug when motor is idling)
- Servo filter
- Anti-collision lights
- Soft start
- Speed control
- Fail-safe system
- Servo reverse
- Y-cable replacement
- Servo balancing
- Go-slow (reduces servo speed)

capability to analyse, may well be adequate for a PPM system to provide limited control of a model. So a PCM-based system will either work correctly — or not at all. There is no middle ground.

In a PPM system a series of narrow pulses is transmitted, see **Figure 1**. The positions of the

Table 2

Characteristics of the 87LPC762 microcontroller

- Operating voltage 2.7 V to 6.0 V
- Instruction cycle time 300 ns to 600 ns at 20 MHz
- RAM: 128 Bytes
- ROM: 2 or 4 kbyte (87LPC764)
- 2 timers
- 2 analogue comparators with built-in 1.23 V reference
- Full-duplex UART
- I²C communications port
- 2 external and 8 keypad interrupts
- Watchdog with separate oscillator
- On-chip power-on reset
- Low-voltage reset
- Internal RC oscillator without external components or quartz crystal oscillator with capacitors
- Programmable port configuration (quasi-bidirectional)
- Open drain
- Push-pull
- Input only
- Serial programming facility
- Idle and power-down modes

joystick and switches are represented by the time intervals between the pulses. The time interval corresponding to a joystick setting can vary between 0.9 and 2 ms. A complete transmitted frame for all channels occupies 20 ms. This means that the control commands are sent 50 times per second. The extra delay allows the decoder circuit in the receiver to synchronise to the next pulse packet.

As a rule servos are connected to the receiver via a three-conductor cable. The connections are power and ground, and a pulse signal whose length determines the servo arm position. Since there are various connector systems in existence with various pinouts, we have shown the commonest connection arrangements in **Figure 2**.

THE MICROCONTROLLER

The 87LPC762 device chosen belongs to a large family of microcontrollers based around an 80C51 core with a OTP (ROM) program memory between 2K and 4K in size. The 'LPC' in the part code indicates low power, low price and low pin count. The notable features of these controllers are their high processing speed and their integrated RC oscillator and reset circuit. The user thus has 18 I/O pins at his disposal. The main technical characteristics are given in **Table 2**.

A 6 MHz quartz crystal (X1), along with capacitors C1 and C2, is used for the microcontroller's clock. This gives an instruction cycle time of 1 ms. Further information, as well as the complete data sheet, can be found on the Internet at Philips' homepage at <http://www.semiconductors.com/mcu/>.

Servo filter function

The servo filter function of this multi-purpose IC can be combined with many of the other functions from Table 1. The main function is illustrated in **Figure 3**.

Receivers that process PPM signals pass on interference to the servos. Such interfer-

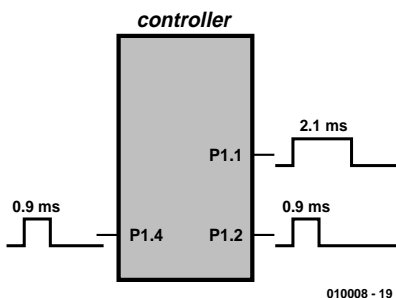


Figure 4. The principle of the servo reverse function.

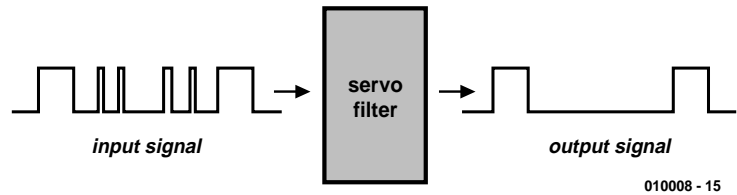


Figure 3. Action of the servo filter.

ence manifests itself in spikes or in errors in the pulse lengths, making the servos either jitter or run out of control.

The output signal of the receiver is first analysed by the microcontroller, before it is sent to the appropriate output or converted into some other action.

Incoming pulses must have a duration of between 769 ms and 2304 ms. After that, the controller waits for approximately 16 ms before starting to check the receiver signal again. The microcontroller now expects a valid signal within approximately the next 9 ms. Errors that occur are counted and when a preset value is reached, the 'fail safe program' is activated.

Because of this interference rejection function the output signal of the receiver must have a repetition rate of between 36 Hz and 57 Hz. Drop-outs in the pulse of up to 40 ms in duration are ignored by the software. Also, the amplitude of the signal at the output of the receiver must be at least 0.6 V_{DD} to drive the Schmitt trigger input of the microcontroller. Spikes on long servo wires that do not reach this level will not be recognised. It is therefore recommended that if the servo connections are long (as on aircraft wings for example) the servo filter (i.e., the circuit board with the multi-purpose IC) be placed as near as possible to the servo. The result of all these precautions is a servo control signal practically free of interference and whose quality comes very close to that achieved in PCM systems.

SERVO REVERSE

Practically every modern radio control transmitter offers the facility to reverse the direction of the servo's motion. Why should this function not be incorporated into the multi-

purpose IC? There are at least two good reasons in favour: two servos can now be connected, and we can easily control a pair of ailerons using separate servos driven from the same channel. Secondly, sometimes a mirror-image servo is required to operate elevators individually. The reason for this is that the relationship between movement of the servo and that of the elevator can depend on where the servo is fitted.

Figure 4 shows the principle of the servo reverse function using the multi-purpose IC. The practical realisation appears in **Figure 5**. The printed circuit board shown in **Figure 6** has been developed for this application, which offers both the servo reverse function and the servo filter function. This printed circuit board (order code 010008-2 from Readers Services) also offers the third function of a replacement for a Y-cable, as long as the reverse sense of the motion of one of the servos is taken into account.

RECEIVER VOLTAGE MONITORING

An adequate supply voltage is vital in a remote controlled model. Continuous checking of the battery voltage is essential to guarantee safety for the spectators (and for the model!). For this reason this function is built in to three of the four application circuit boards for the device; only in the case of the servo reverse circuit board described above is it dispensed with, in the interests of compactness. The special feature of the voltage monitoring system employed here is that brief breaks in the supply are detected and permanently stored.

One of the two comparators in the microcontroller is dedicated to the voltage monitoring function (**Figure 8**). Using the internal 1.23 V ref-

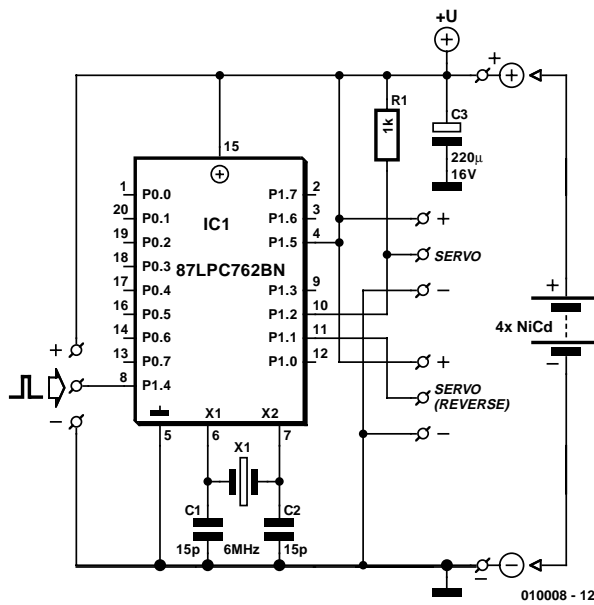


Figure 5. Circuit for the servo reverse function using the multi-purpose IC.

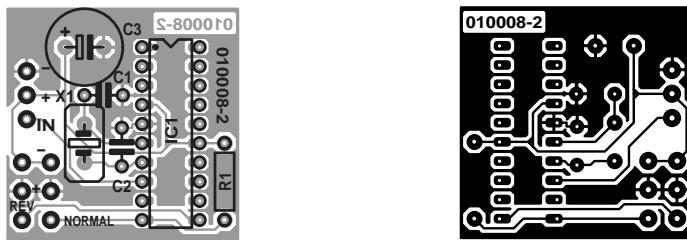


Figure 6. The printed circuit board for building the circuit of Figure 5 also offers the servo filter and Y-cable replacement functions.

COMPONENTS LIST
(PCB shown in Figure 6)
010008-2

Resistors:

R1 = 1kΩ

Capacitors:

C1, C2 = 15pF

C3 = 220µF 16V radial

Semiconductor:

IC1 = 87LPC762BN (programmed,
order code **010008-41**)

Miscellaneous:

X1 = 6MHz quartz crystal

2 off servo cable with
female connector

1 off servo cable with plug
PCB, order code **010008-2**

reference voltage and external resistors R4 and R5 allows the microcontroller to react when the voltage falls below 4.8 V: it turns on permanently the LED connected to port pin P1.6 (Figure 8), and alters the flash rate of the anti-collision lights featured on circuit board 010008-3, to be described in the second instalment in this series. Once lit, the LED stays permanently on and can only be turned off when the microcontroller is reset and supplied with a voltage of at

least 4.7 V. The LED will thus reliably indicate if the supply voltage has at any moment fallen below a safe level. This provides an early indication that the battery should be changed. The switching level for this function can be set as required by selecting suitable values of R4 and R5 in the potential divider. If resistors of the right value cannot be obtained, the desired voltage level can of course be set using a small potentiometer in the place of R4.

TWO-CHANNEL SWITCH

A few extra features can add that certain something to a remote controlled model. A firefloat, for example, might feature a working water pump; a rescue vehicle might feature lights or a horn; or an aeroplane might have landing lights. In many radio transmitters, one joystick channel is used for each switching function, using up a proportional channel. It is not exactly an economical use of the channel capacity to use a full proportional channel to transmit a simple on/off state. Using the multi-purpose IC in the circuit of Figure 9 (with corresponding printed circuit board in Figure 10), moving the joystick in the positive direction operates one switch while moving it in the opposite direction

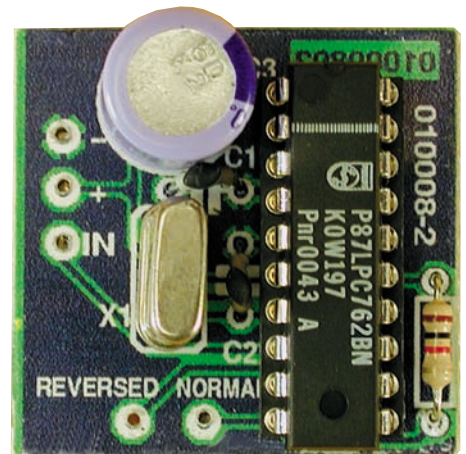


Figure 7. The populated servo reverse circuit board.

operates another. The output state of the switch is latched: that is, the output state toggles on each activation and is held until the next one.

The power MOSFET for channel 2 is driven from microcontroller pin P1.0 and switches when the input pulse width is less than 1.2 ms. The FET for channel 1 is driven from pin P1.1 and switches when the input pulse width is greater than 1.8 ms. The MOSFETs used can switch currents of up to around 10 A without additional cooling and can operate at voltages of up to 55 V.

MODEL FINDER

Anyone who has ever made a forced landing in a cornfield will know how tedious it can be to find one's model again. Moving the joystick wildly to and fro in the hope of hearing the noise of the servos is only occasionally successful. Usually it is only possible to find the model by searching through the field system-

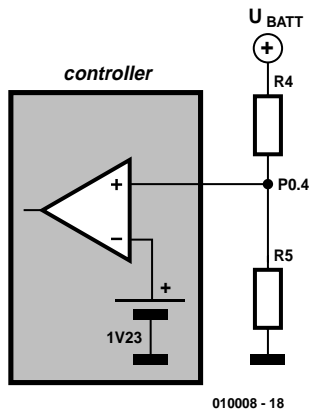


Figure 8. Internal comparator in the microcontroller.

atically — which doesn't always endear you to the farmer! The microcontroller provides on pin P1.7 an active low signal that can drive a buzzer (with built-in electronics) directly. The alarm is automatically activated after 2 seconds if the receiver does not provide a valid signal, either because the transmitter has been turned off or because the signal is severely distorted. This function is available on all the printed circuit boards except the servo reverse board. If this function is not required, the buzzer can be dispensed with.

(010008-1)

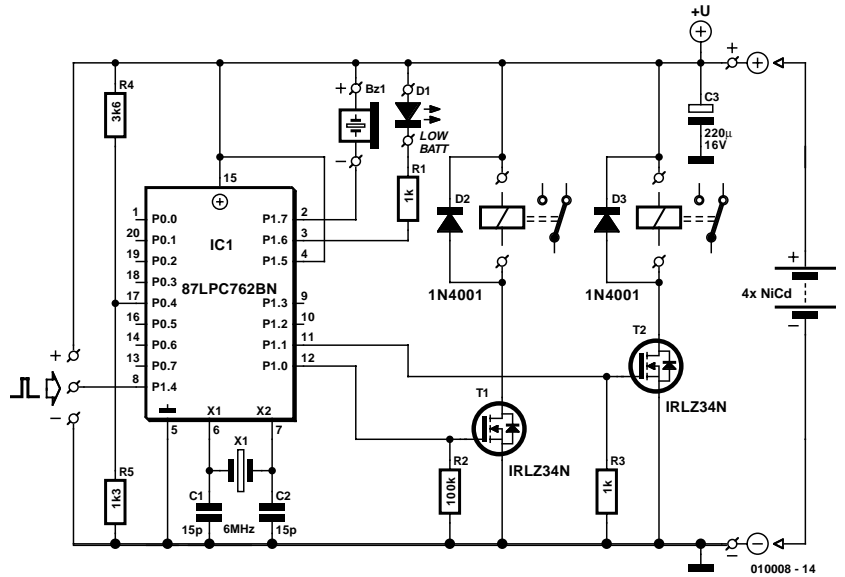


Figure 9. Circuit of the 2 channel switch board, also offering battery voltage monitoring and model finder functions.

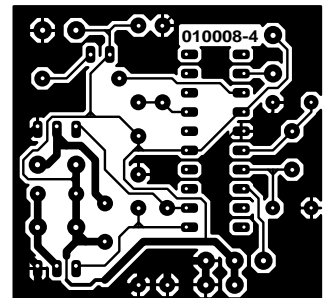
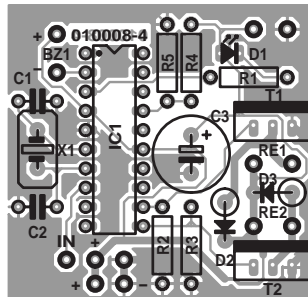


Figure 10. Printed circuit board for building the 2 channel switch.

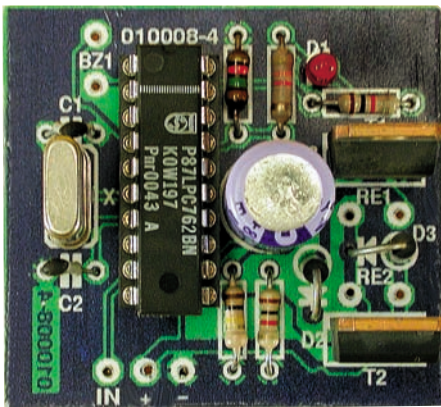


Figure 11. The populated 2 channel switch circuit board.

In the next instalment we will present the two remaining circuit boards, offering speed control, soft start, motor voltage monitoring, BEC

(battery elimination circuit), anti-collision lights, fail-safe system, 'hot glow', differential rudder control, and 'go-slow' functions.

Web addresses

Microcontroller:

www.semiconductors.com/mcu/

N-channel MOSFET:

www.irf.com/product-info/datasheets/data/irlz34n.pdf

COMPONENTS LIST (PCB shown in Figure 10) 010008-4

Resistors:

R1, R3 = 1kΩ

R2 = 100kΩ

R4 = 3kΩ

R5 = 1kΩ

Capacitors:

C1, C2 = 15pF

C3 = 220µF 16V radial

Semiconductors:

D1 = LED, red, low-current

D2, D3 = 1N4001

IC1 = 87LPC762BN, programmed, order code **010008-41**

T1, T2 = BUZ71/IRLZ34N

Miscellaneous:

Bz1 = DC buzzer, 5V

X1 = 6MHz quartz crystal

1 off servo cable with plug

PCB, order code **010008-4**

Disk (source code file), order code **010008-11**