scratch & win £10k+ prize stock **page 4**
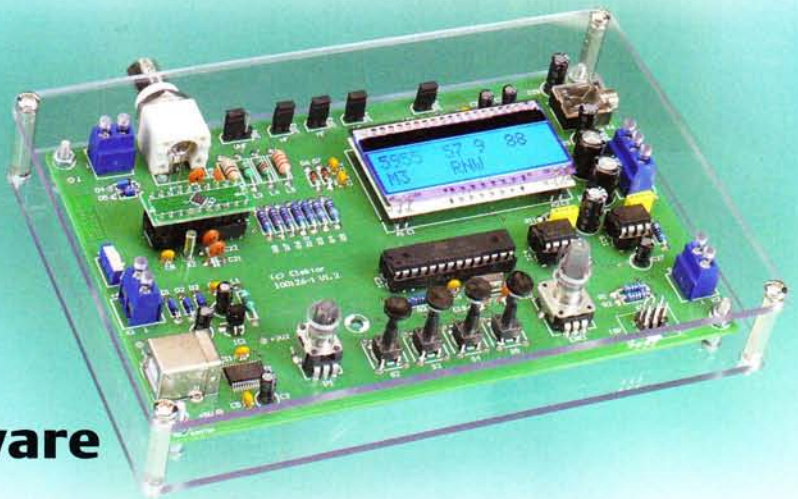
+ focus on circuit simulation

elektor

www.elekt...

Digital Multi-Effects Unit
15 configurable sound effects

Prg:000
Alg:Rev./Chorus

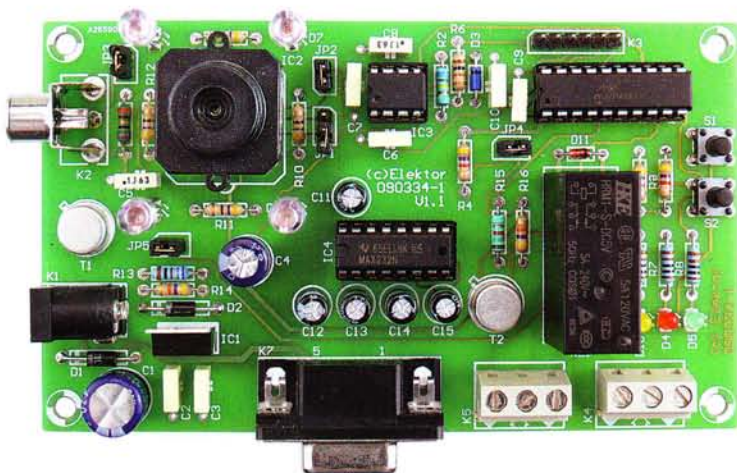## The Elektor DSP Radio (2)
Antennas and PC Software

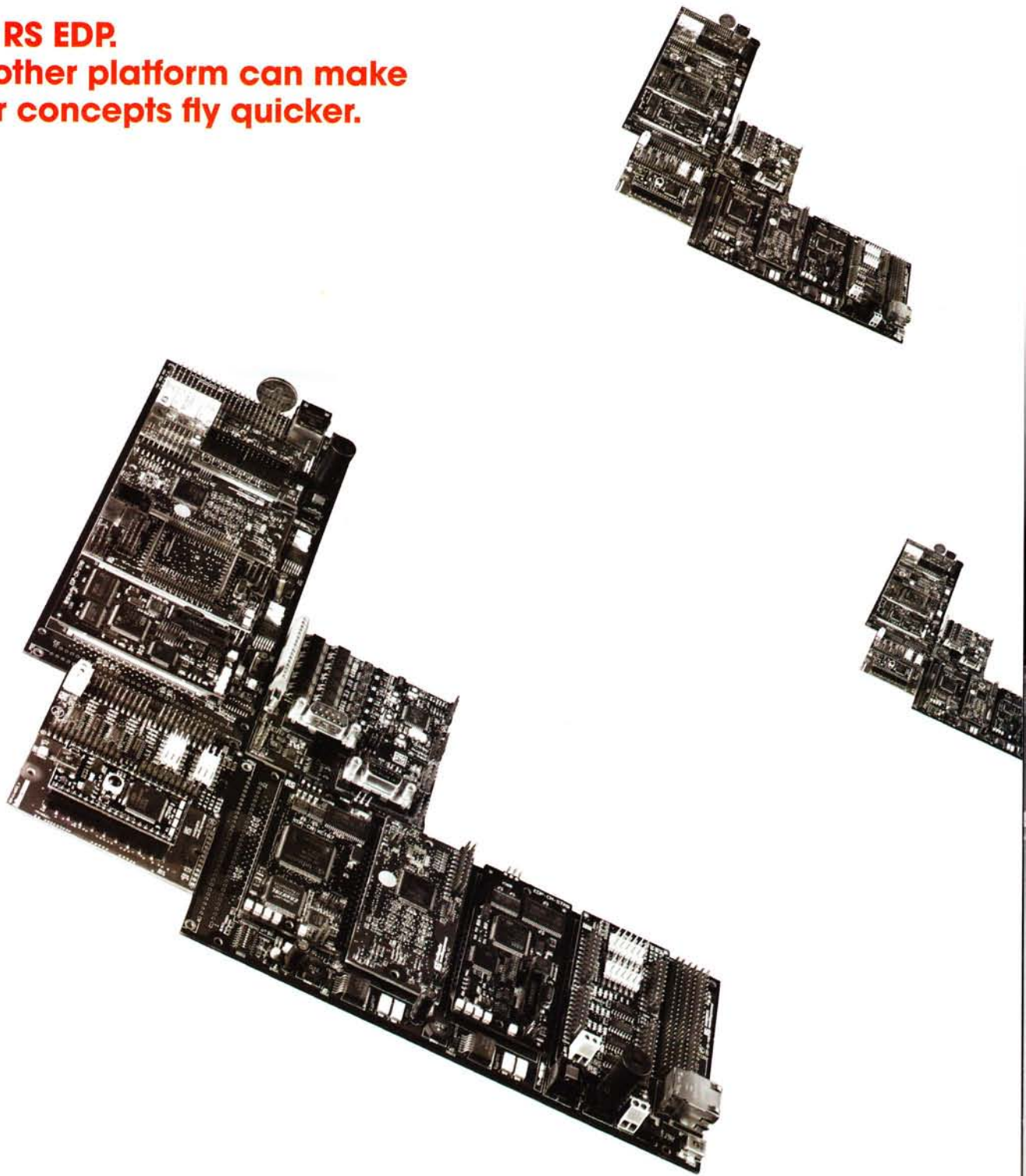Give your projects simple grace with the **Elektor Project Case**

+ Audio DSP Kickoff

+ Unilab V/I Display

Vision System
for small microcontrollers

R38
09

The RS EDP.
No other platform can make
your concepts fly quicker.

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A bit more on simulation!

In this edition, in accordance with our 2010 Publishing Plan, we pay special attention to the issue of electronics simulation. Simulation programs for the PC have been around for decades and in all that time they've become increasingly sophisticated and 'realistic'. But they're not the easiest programs to handle — you have to invest quite some time before you feel confident using them at any level. That is the main reason why many designers still seem to think "All that simulation takes too much time, I'll give it a quick try on the breadboard". But the time and effort in learning about simulation is really well spent. Once you have mastered a simulation program, you'll notice the ease of running such a simulation on a number of different options or settings in a thorough way, without any soldering to do or components to order by mail.

Two articles in this edition approach 'electronics simulation' from a different perspective. There is a hands-on article on the possibilities of the free program LTSpice, which is actually intended to develop switch-mode power supplies, but really has a much wider, largely undiscovered, application range. In 'Virtual Measurements and Predictions' a real-life case is presented of a simulation set up to compare the results of a simulation run against measurements carried out on the actual hardware thing.

If you do not feel like doing any circuit simulation at all, no problem, there's a bunch of classic style projects in this edition to explore, assemble and use in practice. Like an advanced audio effects unit, a high voltage differential probe, a smart video detection system based on an inexpensive PIC microcontroller, and the second part of our beautiful DSP radio.
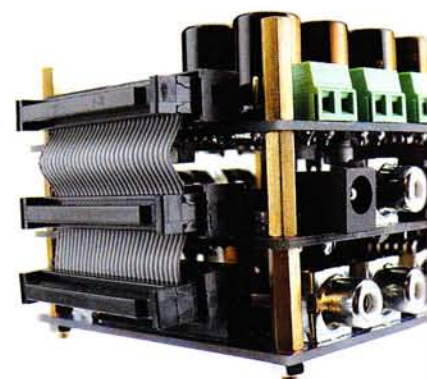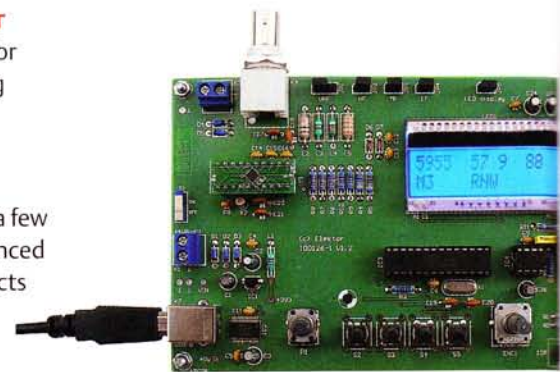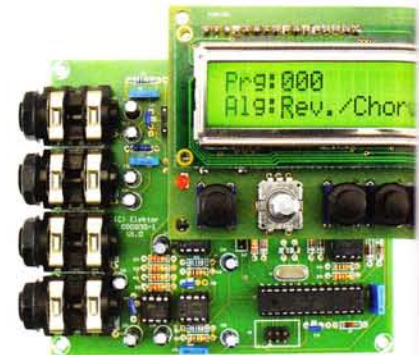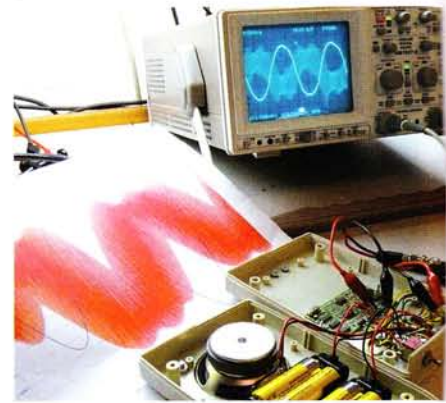
If you haven't already spotted it on pages 4 and 5, we're running a fantastic scratch-and-win sweepstake again this year. There's a mass of prizes in stock, most of them kindly sponsored by our advertisers, including a stunning Hameg digital oscilloscope!

Enjoy the read, the build... and the scratch!

Jan Buiting, Editor

CONTENTS

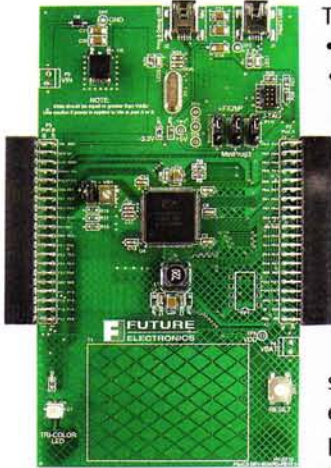# Cypress and Future introduce PSoC 3 Development Board

Cypress Semiconductor Corp. and Future Electronics recently introduced a low-cost development board demonstrating the ease-of-use of Cypress's new PSoC® 3 architecture. The Future Electronics PSoC 3 Development Board shows how developers can leverage the flexible PSoC 3 architecture and the revolutionary PSoC Creator™ Integrated Development Environment (IDE) to streamline designs.

The Future Electronics PSoC 3 Development Board features:
- On-Board Debug/Programmer
- CapSense® Touch Pad Interface
- TriColor — Red/Green/Blue LEDs
- User USB Interface
- Power through USB Ports
- Two Connectors for I/Os and Expansion Boards.
- Optional CAN Oscillator

The two Expansion Board Kit (EBK) connectors on the board enable it to work with a growing ecosystem of solution-based expansion kits that Cypress is developing, including segment LCD, CAN and more. Future will also be developing an expansion board for the new Sharp Pixel in Memory LCD with power consumption 130 times lower than standard LCDs of comparable size. More information on the Future PSoC 3 board is available at www.FutureElectronics.com/cypress. The companies also announced that interested parties can request free onsite introductory demonstrations of the new development board or half-day, hands-on workshops by visiting www.cypress.com/go/FEtraining or by contacting your local Future account manager. PSoC 3 and PSoC 5 devices extend the world's only programmable analog and digital embedded design platform, delivering unmatched time-to-market, integration, and flexibility across 8-, 16-, and 32-bit applications. PSoC 3 and PSoC 5 offer high-precision programmable analog including 12- to 20-bit delta-sigma ADCs, digital logic libraries full of dozens of drop-in peripherals, best-in-class power management and rich connectivity resources. The PSoC Creator IDE introduces a unique schematic-based design methodology along with fully tested, pre-packaged analog and digital peripherals easily customizable through user-intuitive wizards and APIs to meet specific design requirements. More information and free downloads of PSoC Creator are available at www.cypress.com/go/psoccreator.

The free onsite demonstration introduces the PSoC 3 and PSoC 5 architectures and features, shows applications development using PSoC Creator software and the Future Electronics PSoC 3 Development Board, and includes a short lab. The free half-day workshop gives participants hands-on experience using and developing PSoC 3 applications through the PSoC Creator IDE and the CY8CKIT-001 PSoC Development Kit. Attendees will learn how to easily implement high-performance, low-power 8-bit or 32-bit mixed-signal applications with an architecture integrating powerful and programmable analog and digital resources. Interested parties can request introductory demonstrations or workshops by visiting: www.cypress.com/go/FEtraining. The Cypress CY8CKIT-001 PSoC Development Kit is priced at US $249 and is available from Future.

www.FutureElectronics.com/cypress   www.cypress.com/psoc   www.cypress.com/psoctraining

(100555-I)

Robust, easy-to-implement wireless connectivity for portable applications

MK Consultants has introduced a new range of embedded intelligent radio transceivers that operate in the UHF bands of 434, 457, 868 and 915 MHz with a VHF 169 MHz version also available. The iGenesis series operates in narrow band 25 kHz channels producing adjustable RF power up to 15 mW coupled with a receiver sensitivity in excess of −120 dBm and best-in-class blocking immunity against interference.

Housed in a fully shielded SIL package with compact overall dimensions of just 44 mm x 15 mm x 5 mm, the iGenesis series is ideal for the growing number of portable electronics products that require reliable, long range and repeatable wireless interconnectivity. Applications include M2M, Bluetooth and Zigbee replacement for enhanced performance, automatic meter reading, home automation and wireless sensors.

The new modules have a wide operating voltage range of 2.8 VDC to 6 VDC and can achieve a wireless range in excess of one kilometre. A comprehensive software serial data interface including user selectable parameters such as operating frequency, power, data rates, plus transparent or packet protocol selection, eases implementation and allows designers to concentrate on the overall functionality of their products and potentially shorten time-to-market.

iGenesis embedded radio modules have an operating temperature range of −20 °C to +60 °C and meet worldwide radio approval standards. For high volume applications, MK Consultants also offers the modules in a surface mount format suitable for SMD assembly processes.

http://www.mkconsultants.eu

(100555-II)

New PIC24 and dsPIC33 Development System: Easy24-33 v6
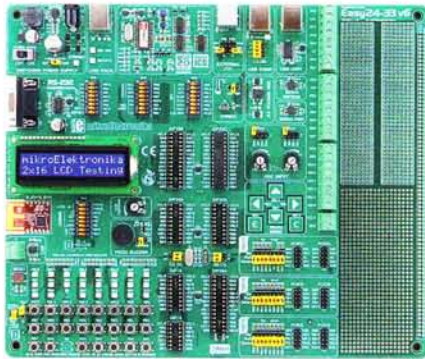
mikroElektronika have introduced their new product called Easy24-33 v6 Development System. The new board is intended for developing and designing devices using 14-, 18-, 20-pin and 28-pin Microchip PIC24 and dsPIC33 microcontrollers. The system includes outstanding features such as USB 2.0 programmer with mikroICD and many peripheral modules such as: Touch Sense, Serial RAM, Serial EEPROM, Piezzo Buzzer and many more. Each feature of the board is supported by practical examples written in mikroC, mikroPascal and mikroBasic PRO for dsPIC30/33 and PIC24 compilers. The new tool comes with comprehensive full colour printed documentation.

www.mikroe.com

(100555-III)

Virtins RTA-168 PC based real time audio analyzer

Virtins Technology have just launched their RTA-168, a PC based real time audio analyzer. It consists of a measurement microphone, an XLR-to-USB converter, and PC virtual instrument software Multi-Instrument. An audio

test CD is also included. All the components of the RTA-168 can be packed into a small black soft pouch case provided and be carried together with your laptop or netbook. It is a handy tool for audio and acoustic measurements in both field and laboratory.

Application examples are frequency response and distortion measurement of a loudspeaker, room/car/auditorium equalization, noise monitoring and analyzing, etc. Combined with the computing power of a modern PC and the powerful virtual instrument software "Multi-Instrument", RTA-168 provides unbeatable performance-to-price ratio compared with any other types of RTA. The main functions of RTA-168 includes: oscilloscope, spectrum analyzer, sound level meter, signal generator, 1/1, 1/3, 1/6, 1/12, 1/24, 1/48, 1/96 octave analysis, equivalent continuous sound level (Leq) measurement, A, B, C, ITU-R 268 frequency weighting, linear and exponential time weighting, parameter measurements (THD, THD+N, SINAD, SMPTE IMD, DIN IMD, CCIF IMD), etc.

The RTA-168 has two standard configurations: RTA-168A (with Multi-Instrument Standard license) and RTA-168B (with Multi-Instrument Standard license). The RTA-168A comes with an ECM999 measurement microphone with extremely flat frequency response within 20-20kHz. The RTA-168B comes with an EMM-6 measurement microphone with extremely flat frequency response within 20 Hz – 20 kHz. Moreover, the frequency response of the microphone is individually calibrated and thus compensated in the software.

Together with the launch of RTA-168, Virtins have released Multi-Instrument 3.2 Build 3. It can be downloaded at: www.virtins.com/MIsetup.exe. If you have not tried this software before, or if you have tried an older version than 3.2, you can download it and have a 21-day fully functional free trial. If you are an existing customer, you are eligible for free upgrading to the same license level you own.

www.virtins.com

(100554-IV)

Tinytag radio data loggers

The Tinytag wireless data logging system distributed by PSE Priggen Special Electronic consists of a receiver which is connected to a PC and a number of radio loggers. Each radio logger is a self-contained, battery powered unit that can receive, log,

store and transmit data to other radio loggers, as well as the central receiver.

Each radio logger (or wireless data logger) has a line of sight range of up to 200 metres (600 ft), but there is no limit to the number of loggers you can have, or to how far away you have them. Data will always find its way back to the PC because it can be relayed from one logger to another logger in range, until it finds its way back to the receiver.

The self-configuring loggers will dynamically adjust to deal with obstructions. If a logger is unable to find a path back to the receiver, its data is stored up to 2 weeks locally until a path becomes available.

This combination of features delivers a very robust radio network which ensures that data is never lost during transmission. The radio software provides mecha-

nisms for third party software integration using Modbus or http accessible CSV files. This software is also fully compatible with 'wired' Tinytag products and existing users will find the interface familiar. Data from the receiver can be accessed over a network. Forwarding alarm messages via E-mail or SMS is also possible.

The radio loggers have got a rugged waterproof (IP67) housing for the operation in harsh industry environment and in the exterior. They are available in versions for the connection of 1, 2 or 4 external temperature probes or with a temperature/humidity probe. Several thermistor probes as well as a compost probe are available as accessories. The operation of the wireless system is licence-free in the European Union at 869.8 MHz ISM band. Versions for Australia and America are also available.

The user replaceable alkaline batteries allow an operation time of up to one year. Low

battery levels are indicated.

Typical applications for this radio data logger system are the monitoring of refrigeration chains, warehouses and buildings, the capturing of environmental data and composting processes.

www.priggen.com

(100554-V)

1590 STOMP boxes – used by electric guitarists everywhere

Hammond Manufacturing's 1590 STOMP die-cast aluminium enclosures are the hous-

ings of choice for leading stomp box manufacturers. Stomp boxes, also known as guitar effect pedals, are, as the name suggests, foot-operated equipment used by electric guitarists to produce preset effects such as distortion, wah-wah, delay, chorus and phaser. Many players will have a bank of up to 20 stomp boxes, each providing a different effect.

The 1590 STOMP boxes are a rugged, easy to machine die-cast aluminium enclosure well able to cope with the demanding on-stage environment in which they will be used. Available in two sizes, 4.39 x 2.34 x 1.06", 112 x 60 x 27mm, and 4.67 x 3.68 x 1.18", 119 x 94 x 34mm, they are finished in a smooth gloss polyester powder paint, which does not chip after machining and

provides a good surface for labels and silk screening.

As would be expected, the standard colours are strong and vibrant. Cobalt Blue, RAL 5013; Green, RAL 6024; Light Gray, RAL 7035; Orange, RAL 2009 and Red, RAL 3011 are available ex-stock; the units can be factory finished in other colours to suit corporate styles. A lap joint seals the units to IP54, protecting against the ingress of dust and water, and the painted finish is only applied to the external surfaces, maintaining RFI integrity.

All units can be supplied factory modified with machining and silk screening to the user's specification; AutoCAD and PDF dimensioned drawings can be downloaded from the Hammond website.

www.hammondmfg.com

(100554-VI)

Low cost, weatherproof antennas for 2.4 GHz applications

Pulse introduces its new low cost, weatherproof antennas for 2.4 GHz devices used in industrial, smart grid metering, security, broadband access, and other machine-to-machine(M2M) applications. These robust, high-efficiency, compact antennas meet Bluetooth, WLAN, WiFi, IEEE 802.11b/g, ZigBee IEEE 802.15.4, and 2.4 GHz ISM band system standards.

Pulse's W5001, W5010, and W5011 antennas function in ruggedized conditions, offering IP65 water ingress protection in accordance with international standard IEC 60529, UV protection, and the ability to withstand 100 mph wind loading. They have a maximum gain of 1.5 dBi in the 2,400 to 2,500 MHz range, an efficiency of 70%, and an operating temperature range

of –40°C to 85°C. The antennas have a small footprint and low profile, measuring only 128mm x 18mm.

Pulse's W5001, W5010, and W5011 antennas are available in three SMA connector configurations: fixed right angle, straight SMA, and RP-SMA. They are packaged in quantities of 20 pieces per bag.

www.pulseeng.com

(100554-VII)

Upgraded 12-GHz sampling oscilloscope

Pico technology's new PicoScope 9201A PC Sampling Oscilloscope is an essential tool for analysing electrical communications standards and characterising connectors, cables, IC packages and printed circuit boards. Unlike a real-time oscilloscope, the PicoScope 9201A builds up a picture of a repetitive signal by combining samples from a large number of cycles of the waveform. This enables it to achieve very high effective sampling rates (up to 5 TS/s) and fast time bases (10 ps/div to 50 ms/div) at a much lower cost than a real-time scope of comparable bandwidth.
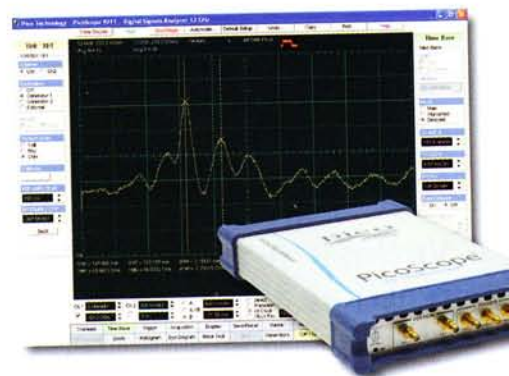
The PicoScope 9201A has two 12-GHz input channels, enabling it to analyse a wide range of communication standards up to 6.6 Gb including Ethernet, SONET/SDH, Fibre Channel, InfiniBand, XAUI, G.703 up to 155 Mb, and ANSI T1 up to STS3. Mask limit tests for all these standards, and many more, are included. The powerful analysis functions include high-resolution cursors, automatic eye-pattern measurements with statistics, and FFT. The PicoScope 9201A also has two trigger inputs: a 1 GHz direct trigger and a 10 GHz prescaled trigger. The PicoScope 9201A has 20% lower noise

and 5 dB better inter-channel isolation than its predecessor, the PicoScope 9201, plus many other improved specifications. The PC software has also been upgraded with over 70 new standard masks such as RapidIO up to 3.125 Gb/s, PCI Express up to 5 Gb/s and Serial ATA up to 3 Gb/s. The scope connects to any Windows XP or Vista computer with a USB 2.0 port.

Unlike traditional, bulky bench-top instruments that contain a PC and a display, the PicoScope 9201A is housed in a compact enclosure that takes up very little space on your workbench. If you're working at a customer's premises, the only extra equipment you need to carry is a laptop and a mains adapter.

The PicoScope 9201A is on sale now, priced at only £5,995. This includes all of the features listed above, software updates downloaded from our website, and technical support.

www.picotech.com

(100554-X)

LED modules, light strips employed in magma crawl lava tunnel exhibit

TT electronics OPTEK Technology's LED module strips and flexible LED light strips have been employed by Adirondack Studios in the design and fabrication of the "Changing Earth" exhibit at the Franklin Institute in Philadelphia, PA. The exhibit includes a Magma Crawl display – a crawl-through fiberglass tunnel that uses a sequence of programmed lights and heating elements to simulate the flow of lava through the earth.

The OVM12F3x7 Series LED module strips are a versatile string of 30 super flux LED modules designed for surface illumination to create decorative or special effects.

Elnec: new series of extremely fast multiprogramming systems

Elnec, Europe's leading Device Programmers manufacturer, have released a new series of extremely fast multiprogramming systems, optimized for programming NAND and NOR Flash memories. These new programmers are able to decrease programming times of NAND and NOR Flash memories up to 70% in comparison to its predecessors. Despite the programming speed, the appeal of these new systems is its universality. They can program more than 53,000 different devices. The new series of multiprogramming systems, designated primarily for programming medium sized batches of device comprises types BeeHive204 and BeeHive208S.

device	Size [bits]	Program+Verify [s]
QB25F640S33 (serial Flash)	800200hx8 bit (64 Mega)	30,7
Am29DL640G (parallel NOR Flash)	400080hx16 bit (64 Mega)	24,0
K8P6415UQB (parallel NOR Flash)	400100hx16 bit (64 Mega)	13,0
K9F1G08U0M (NAND Flash)	8400000hx8 bit (1 Giga)	122,7
AT89C51RD2 (microcontroller)	10000Hx8 (512 kilo)	15,0
PIC18LF452 (microcontroller)	4000Hx16 (256 kilo)	4,0

The BeeHive204 is a concurrent multiprogramming system designated for high volume production programming with minimal operator effort. The programmer has four independent programming modules. Each programming module starts programming at the moment a chip is detected in the socket — independent of the status of other programming modules. For example, three modules can be performing a program, verify or blank check operation while the operator is replacing a programmed chip with a blank one. The BeeHive204 reliably programs a wide range of programmable devices in the ZIF socket with more than 800 models of socket converters available. Programming is also available through the ISP connector. It is important to note that Elnec's software allows two BeeHive204 programmers to be connected and operated simultaneously from one PC, making for even more flexible production solution.

The Stand-alone concurrent multiprogramming system, the BeeHive208S has eight independent programming modules. Except of higher number of programming modules, this programmer differs from the BeeHive204 by its stand-alone mode of operation. That means that the programmer does not have to be connected to an external PC while in operation. The BeeHive208S can be controlled simply through touch screen display and Windows® XP Embed operating system. A 320 GB hard disc drive and the ability to upgrade the software using a Flash drive ensures a reliable production solution without any additional costs.

www.elnec.com   (100554-IX)

Implemented in the Magma Crawl tunnel, the LEDs' response time is fast enough for instantaneous flashing lights, while power consumption is low (0.3 to 0.5W typical depending on color). The LED modules are available in strips ranging from three inches to more than 40 feet.

The OVQ12S30x7 Series flexible LED light strips represent a scalable lighting solution using high brightness LEDs mounted on flexible circuit board. Used to build the convection flow meter and the eruption indicator in the Magma Crawl display, these light strips are highly flexible, providing easy installation and allowing many different configurations.

www.adkstudios.com   www.optekinc.com

(100554-XI)

Give your Projects Simple Grace

With the Elektor ProjectCase
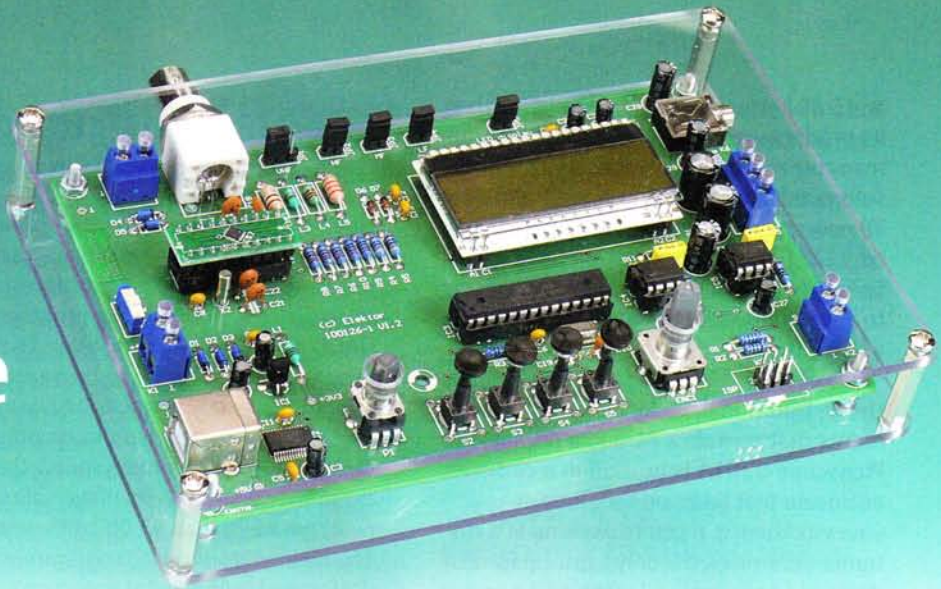
A brief tour of the World Wide Web clearly shows one of the problems facing many electronic enthusiasts: the enclosure. All too often it's a plastic box in a dismal shade of grey, cluttered with an awkward arrangement of connectors, controls and indicators. You spent hours designing a stunning PCB layout, assembled the board neatly, and it ends up in a humdrum plastic box. Isn't there a better way?

The answer is yes, and it's easy. Peter Groen, one of the founders of FabLab in Groningen, The Netherlands, came up with the idea of using simple, transparent polycarbonate (Lexan) panels and a few standoffs. You simply fit the PCB in the middle, with room on all sides for wiring and connectors.

The display can be viewed directly and everything looks a good deal better, with the board nicely visible and well protected. To give you an idea of what it looks like, we used it to package the DSP radio board described elsewhere in this issue of Elektor. The polycarbonate panels are easy to

machine, so making holes for controls, connectors, slots and whatnot shouldn't present any problems. Four sturdy rubber feet on the bottom keep everything rock-solid. Welcome to the ProjectCase!

To make things easier for you, we had a large number of ProjectCase kits made up for us. The panels measure 175 x 115 x 3 mm, which matches the Eurocard format.

The kit also includes all the necessary standoffs for securing the PCB and the polycarbonate panels. With a bit of creative imagination, you can do just about anything

with it. The ProjectCase components are supplied with a protective film. Be sure to leave it on until all the holes, slots and so on have been made.

The price of the Project Case may be found on the Elektor website. We'll be happy to slip a ProjectCase kit into an envelope and send it to you so you can get down to work. Order your ProjectCase kits from the Elektor Shop at www.elektor.com.

(100500-I)
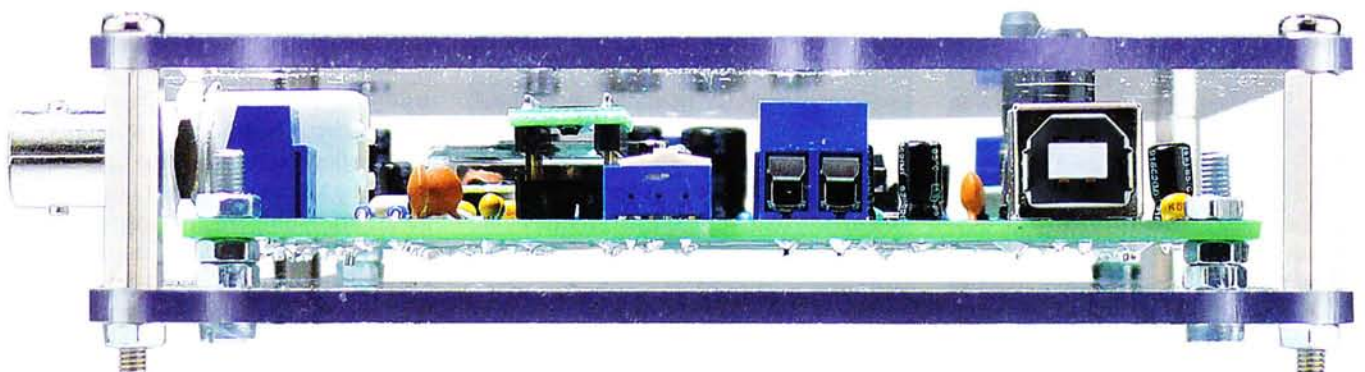
Build a Scrolling LED Message Board in One Day
using
the mbed rapid prototyping platform

By Clemens Valens (Elektor France Editor)

When my friend Grégory showed me a 64×32 pixel two-colour LED panel picked up from an Internet retailer for about 50 dollars, I immediately ordered one too. That would be a great opportunity to put the ARM mbed module I had lying around through its paces. I wondered, would this project be as easy as the mbed crew wants us to believe?

The first time I learned about mbed was when reading an article about it in Circuit Cellar #227. I was immediately seduced by the concept and wanted to try it. Having the right kind of relations I managed to force my way into the mbed beta testing program and got a couple of modules to play with. These beta modules were based on NXP's LPC2368 ARM7 processor. Shortly after mbed went live the ARM Cortex-M3 module using NXP's LPC1768 was introduced and I managed to get one of those too. All I needed to do at that time was find an application for these boards.

For those who did not read Tom Cantrell's article (you have no excuse as it can be downloaded for free [1]) I will first briefly explain the mbed concept.

So what exactly is mbed?
mbed (pronounced "embed") is a rapid prototyping platform for ARM processors, developed and maintained by ARM. Although two different mbed modules exist, both based on NXP chips, I don't think you can still buy the ARM7 one. Other modules using chips from other manufacturers may be expected, but don't hold your breath.

At first blush an mbed module just looks like any other microcontroller evaluation board with some LEDs, a reset button and a USB connector. It's small (25 ×53 mm), it's got 40 pins spaced 0.1" and there isn't much on the board besides the processor, five LEDs, a pushbutton and a USB connector. The top side of the board that is, because when you flip it over you a greeted by a second processor, a DB83848 Ethernet transceiver and an AT45DB161 16 Mbit serial flash memory chip. The second processor, an LPC2148 ARM7 (rebranded as 'mbed interface' on the current LPC1768 module), is almost as powerful as the main processor and this is where all the innovation is lurking.

The 'mbed interface' makes the module appear on your host computer as an MBED (note the capitals) mass storage device and it contains one HTML file. When you open the file you are taken directly

Figure 1. This is the web page displaying when you open the HTML file on the mbed module. Create an account and kick off!

to the mbed login page (if you are connected to the Internet, of course), see **Figure 1**. After creating an account you can start programming right away, because the compiler (free and for unlimited use) is waiting for you online. That's right, there is nothing to install! Plug and play, you are up and running in like five minutes.

But that's not all. The mbed crew did most of the hard work for you by developing a library that supports all the peripherals of the processor and more. It was probably inspired by Arduino as it implements similar concepts like DigitalIn/Out and AnalogIn/Out. This library lets you concentrate on your application instead of getting bogged down in the swamp that goes by the name of Low-Level Register Programming. A real time saver, as ARM processors have about a zillion registers.
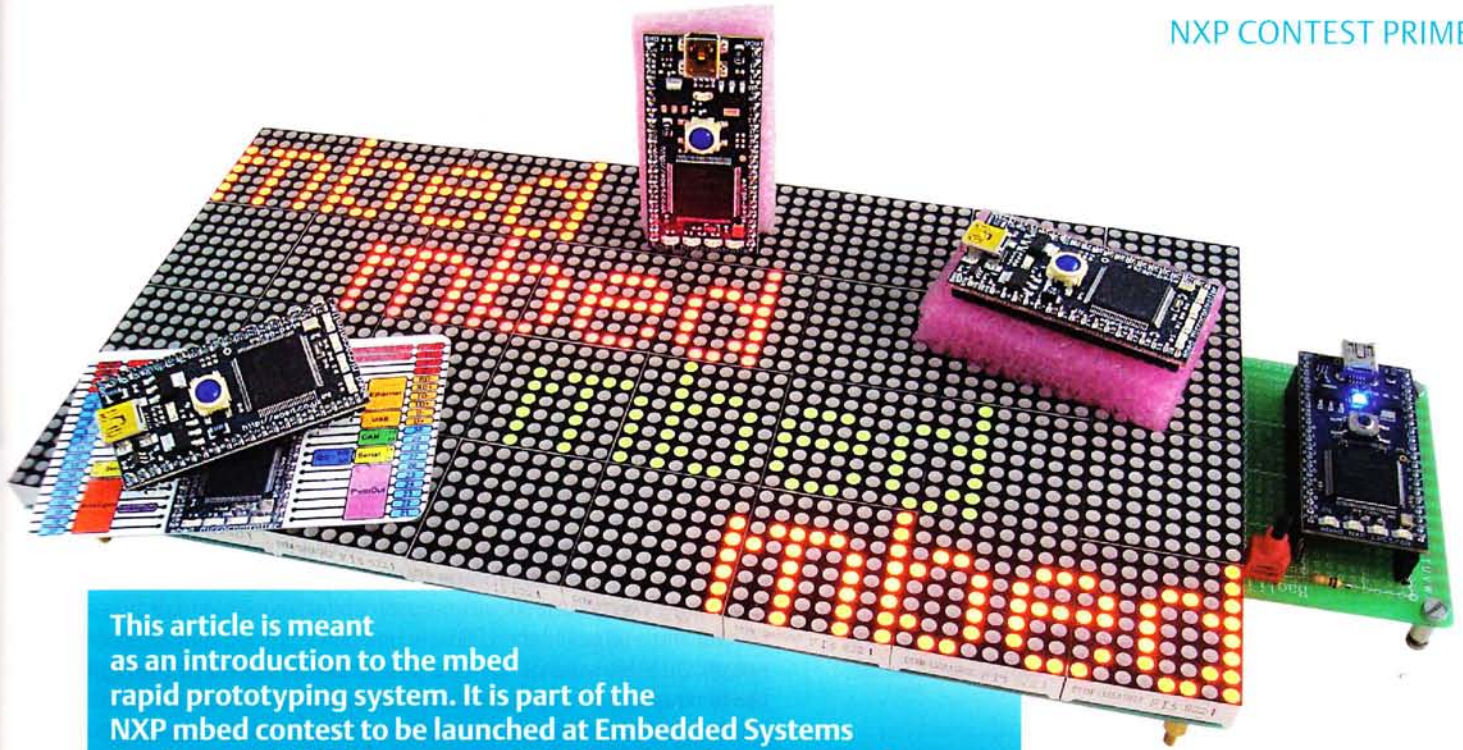
This article is meant
as an introduction to the mbed
rapid prototyping system. It is part of the
NXP mbed contest to be launched at Embedded Systems
Conference (ESC) Boston 2010 (USA), September 20-23, and
organised jointly by Elektor and Circuit Cellar. Stay tuned!

When your online source code compiles without errors the system suggests to download the executable. You save it on the MBED drive and then press the Reset button on the board. The mbed interface chip will now program the mbed processor and your application starts running. Sounds easy, eh? When your project is finished you can download it as a .zip file for archiving purposes or whatever.

If your program is not behaving as it is supposed to be, you're unfortunately more or less on your own. A debugger is not available, although you can add a serial port to the mass storage device so that you can print debug messages to a terminal on a host computer.

The mbed web site [2] plays an important part in the concept. Apart from the compiler you will find on the web site detailed explanations of the mbed library functions in the Handbook, sample projects in the Cookbook, a blog, a forum and more. You can publish your projects for other users to enjoy (hopefully) as well as import projects from other users into your own project. It's all very community-structured and open. The only thing that's definitely not open is the mbed library, which is of course very disappointing.

Why? Because it is now difficult to figure out if the library is efficient or buggy or suited or not for what you are trying to do. The mbed library implements the most common functions deemed useful in many cases, but it doesn't allow you to exploit the processor in all its glory. Understandable, sure, because if it would, it would be a hell of a lot more difficult to use, but annoying for those who need to know how exactly the library tweaks the bits and registers. Of course you can reverse-engineer the executable and inspect the registers, but that sure takes the 'rapid' out of the concept. Also, when the library is updated you may have to do it all over again.

And now that I am bashing, let me also mention the blue power-on LED on the board that shines so bright that it hurts my eyes even in clear daylight. When will designers finally understand that you do not *have* to force ten or so milliamps through a modern LED? A nice glow would have been more than sufficient.

Although the processor has 100 pins, the module only has 40. They
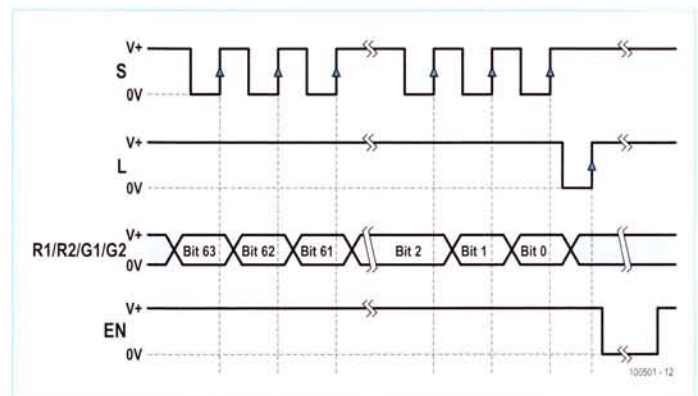
Figure 2. The timing diagram of the 32 line LED panel looks simple enough. However a subtlety is hidden in the four bit data word R1/R2/G1/G2: R1/G1 control the upper 16 lines of the panel, R2/G2 the lower 16 lines. Therefore this pattern has to be repeated only 16 times to draw all 32 lines.
The EN signal controls the power to the LEDs. Activating it (logic Low) for a while after a line has been written gives nicer results than simply activating it all the time.

have been chosen in such a way that all the peripherals integrated in the controller are accessible, but not all at the same time. If, for instance, you need three serial ports, you can't have an I²C bus as well. The power supply (0 V, 4.5 V to 14 V in, USB 5 V out and 3.3 V out), the Ethernet interface and the USB port have dedicated pins, there is a reset input and two IF pins (these are for the USB port of the mbed interface, not the target processor and you are not supposed to use them). The remaining 25 pins can be GPIO or allocated to some other function.
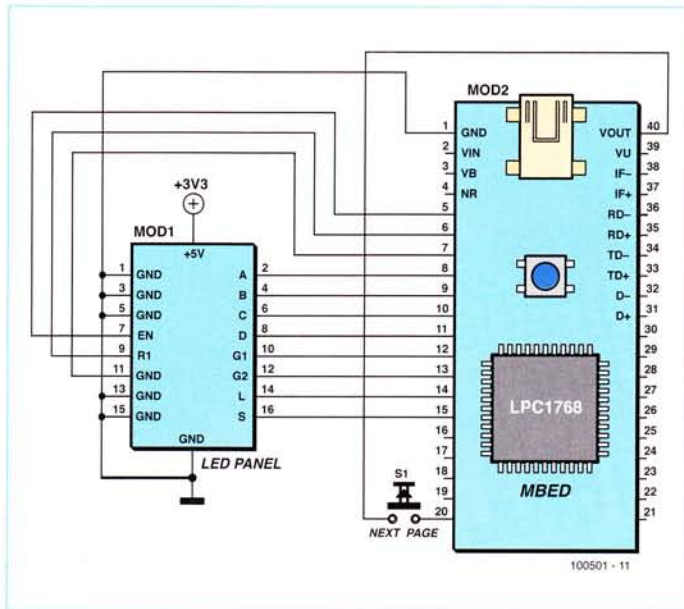
Figure 3. The circuit diagram is really simple since no special parts are needed to control the LED panel. Note that in this basic setup the mbed module is powered from the USB bus, whereas the LED panel needs a separate supply because it needs more current than mbed can provide. Also note that even though the LED panel is specified for 5 V, it will run from 3V3 without any problem. The panel at 5 V driven by a 3V3 mbed will work too.

On to the application

The goal of the project was to build a message board with an easy way to put, well, messages on it! Today's electronics often equates to USB and that's why I ended up using the mbed module. Since the module doubles as a 2 MB USB mass storage device it would be very easy to load message files on it produced on a computer with a proper text editor. That would be an intuitive and comfortable user interface that even non-electronics engineers can understand. The user manual of the LCD panel (DE-DP-DP029~033_Ver1.0_EN.pdf [3]) told me I had to do some bit-banging as its serial interface is not standard SPI or something similar. **Figure 2** shows the timing diagram. The display has 32 rows of 64 two-colour LEDs, i.e. a grand total of 4096 LEDs! The addressing of the 64 rows (32 two-colour rows) is a bit strange because the address bus is only four bits wide and can thus only address 16 rows. This problem was solved by the manufacturer by breaking the display in two 16 row displays and using a 2-bit data bus. The two 16-row displays are addressed in parallel and the two data bits select the red and green LED for the row. The display has only enough memory for two rows, one upper row and one lower row. This means that the display has to be refreshed continuously in order to display a message or image. An advantage of this technique is that the display will not consume 40 amps or so when all the LEDs are on.

After a line is written it should be activated using the EN signal. This signal determines the brightness of the display but as it does so, it also affects the refresh rate. This will in turn govern flickering as well as the scroll speed, since scrolling does not use a separate timer and depends directly on the refresh rate.

A little bit of hardware

How I hooked up the display to the mbed module is shown in **Fig-**

ure 3. Since most mbed pins can be GPIO, I simply started at the first available one (pin 5) and worked my way down. I later added a special feature pushbutton to pin 20 simply because it was nearest to the pushbutton.

For compatibility reasons I ran the display from 3V3 (3.3 V) instead of the specified 5 V, because mbed is 3V3 and the display let me do it. Once I mastered the display I powered it from 5 V and that worked fine too, the advantage being a much higher brightness. The display was powered from a separate bench supply as it requires several amps when all the LEDs are on and the mbed module cannot provide this much. I powered the mbed module itself through the USB port.

I had a bit of a hard time implementing the not terribly complicated serial protocol for the simple reason that the user manual of the display has an error. In the table on page 8 and 9 that I was using the pin numbers of the 'L' and 'S' signals are swapped with respect to the connector pinout shown just above it. The sample code I had extracted from the document worked fine, but my own code didn't. Very frustrating. With this problem out of the way everything went smoothly and I was able to very quickly plot a pixel anywhere on the display in one of the four available colours (red, green, orange and black).

It's interesting to note at this point that the mbed library has pin and bus functions. It seems logical to use the BusOut function for both the address and the data buses, but it turned out that the bus function is much slower than the pin function. I therefore ended up using four DigitalOut calls for the data bus and one BusOut call for the much slower address word. The reason for this performance difference would probably be clear if only the source code of the mbed library had been published.

Dots & fonts

It was now time to start writing characters on the display. Characters need a font and bitmapping a font is an intensely boring and tedious job. In the spirit of letting other people do the hard work for you, I turned to the Internet. Many people have been busy over the years bitmapping fonts and writing utilities to do it automatically, so after some ten minutes I found myself playing with The Dot Factory [4] (**Figure 4**). This Windows utility makes it a snap to convert any font installed on your PC into C source and header files. Select the font, set some options and click the Generate button. The only weird thing is that you can't save the files; you have to copy and paste. That's a minor inconvenience though for a nice free tool, so I created the files without complaining and imported them into my mbed project. Adding the code to display the characters anywhere I wanted them was pretty straightforward too.

Adding an INI file system

The next step was adding a way of putting messages on the system. This was the main reason why I had opted for an mbed module: it has a USB interface and the mbed library provides a convenient file system. And convenient it was. All I had to do was to add 1 (say, one) line of code to my program to get it working:

Figure 4. The Dot Factory:
Now why did the author(s) forget to implement file saving?

```
LocalFileSystem local("local");
```

Now my program was able read the files that I copied to the MBED mass storage device from my PC.

I had decided to use an INI-type file for my message files. I wanted several pages, browsing through the pages would be possible thanks to the pushbutton. Every page would be a section in the INI file and it would be possible to specify, on a section by section basis, the text for each line. With my 8-point Arial font I had room for four lines per page without characters spilling into other lines. I turned again to the Internet to look for an easy INI file C-library. There are many around and I picked one more or less randomly: inih [5] it would be! This turned out to be a good choice as it's a remarkably easy to use library and no more than five minutes were needed to integrate it into my project.

Time to stop

I had now come to a point where 'features' started creeping in. First of course message scrolling: left, right and no scrolling. Then scrolling at any speed. Now that was a tough one. Not so much to implement as it was to look at, because it resulted in pixel bleed. Characters became blurred and the display became uncomfortable to read. This feature was therefore scrapped, but another one quickly found: every line its own colour and its own (x,y) starting position. I tried many more things but finally settled for the features mentioned above plus page skipping. Then I called it a day.

Tool trouble

Building this application took me about one day of work. I did not do it all in one go, but I could have if my agenda had let me. Using the mbed module instead of some other microcontroller board definitely saved enormous amounts of time as did the mbed library. Not a single ARM register was inspected by me and I did not have to refer to the processor's datasheet or programming manual at all. The little mbed experience I had before starting this project probably helped a bit too, because I knew what I could expect from the mbed library.

The only severe hindrance was the mbed server going down or offline. This happened once during my project, wasting a couple of hours and it makes you feel silly. There you are happily programming away and suddenly: Bam! No more tools... And no one you can yell at (what was that mbed helpdesk phone number again?). Another inconvenience is that the compiler or the server seems to get stuck every once in a while. This is rather annoying as it probably means loss of those clever modifications you just added to your code. Exactly what happens I don't know, but you have to close and then reopen the compiler window without being able to save your work. The mbed crew should add some temporary saving for recovery purposes to the environment as this happened to me several times.

Note that you do not *have* to use the online compiler. The executable that you download and program into the target is an ordinary binary, i.e. no proprietary file format. If you can produce such a file with a toolchain you already own you're good to use the mbed board too.
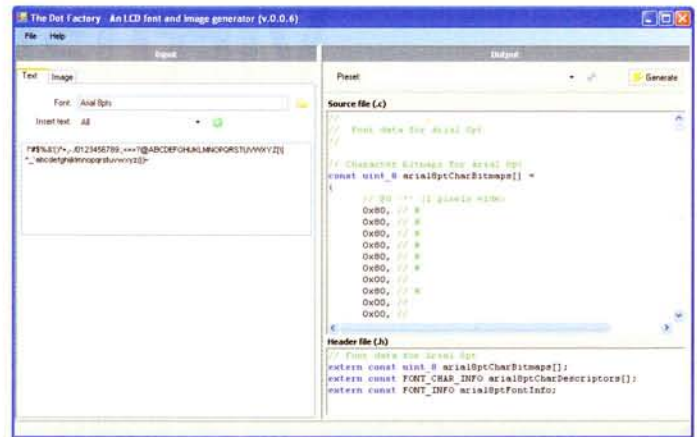
The mbed library can also be used since the header files and a binary archive (.ar) can be downloaded. This is not the easy way however, and, understandably, not really supported by the mbed crew. However, some people have found ways to mbed off line.

Similarly you don't need an mbed board to run the executable. Any LPC1768 board can be used, which means that you don't have to design an mbed module into your product.

Conclusion

While working on the project I was using the beta LPC2368 mbed, but for this article I had to use the LPC1768 mbed. Switching between the two modules turned out to be extremely easy; all I had to do was selecting the right mbed from a dropdown list on the compiler menu. The recompiled code then worked immediately without any modifications.

All in all I think mbed is pretty cool. This project made very low demands on the power of mbed, the Ethernet interface is worth a try too. Although I would not recommend mbed for professional or consumer quality software development since you have little or no control of the tools and the library (what if ARM decides to pull the plug on mbed?) and there is no debugger, it is definitely a great tool for quickly putting a proof of concept together. It is also great for those one-off projects where you need just a little more processing power than you can squeeze from systems like an Arduino.

Hats off to the mbed crew who have come up with a great tool!

(100501)

Internet Links & Literature

[1] *Easy (E)mbed, An Alternative Approach to Embedded Programming*, Tom Cantrell, Circuit Cellar #227, June 2009, pp 68-72. (Free download at www.circuitcellar.com/archives/viewable/Cantrell-227.pdf).

[2] mbed.org

[3] LED panel: www.sureelectronics.net/goods.php?id=718

[4] The Dot Factory: www.pavius.net

[5] INI Not Invented Here: code.google.com/p/inih/

[6] Software: www.elektor.com/100501

Virtual Measurements and Predictions

By Thijs Beckers (Elektor Netherlands Editorial)

As an electronics enthusiast or engineer, you'll know that the design of electronic circuits can be complex. These days, however, you can get help from computer programs, so you'll know in advance if your design will function as intended. Or perhaps you won't? We compared the measurements made on a real hardware prototype with the simulated data produced by a leading program and drew our conclusions.
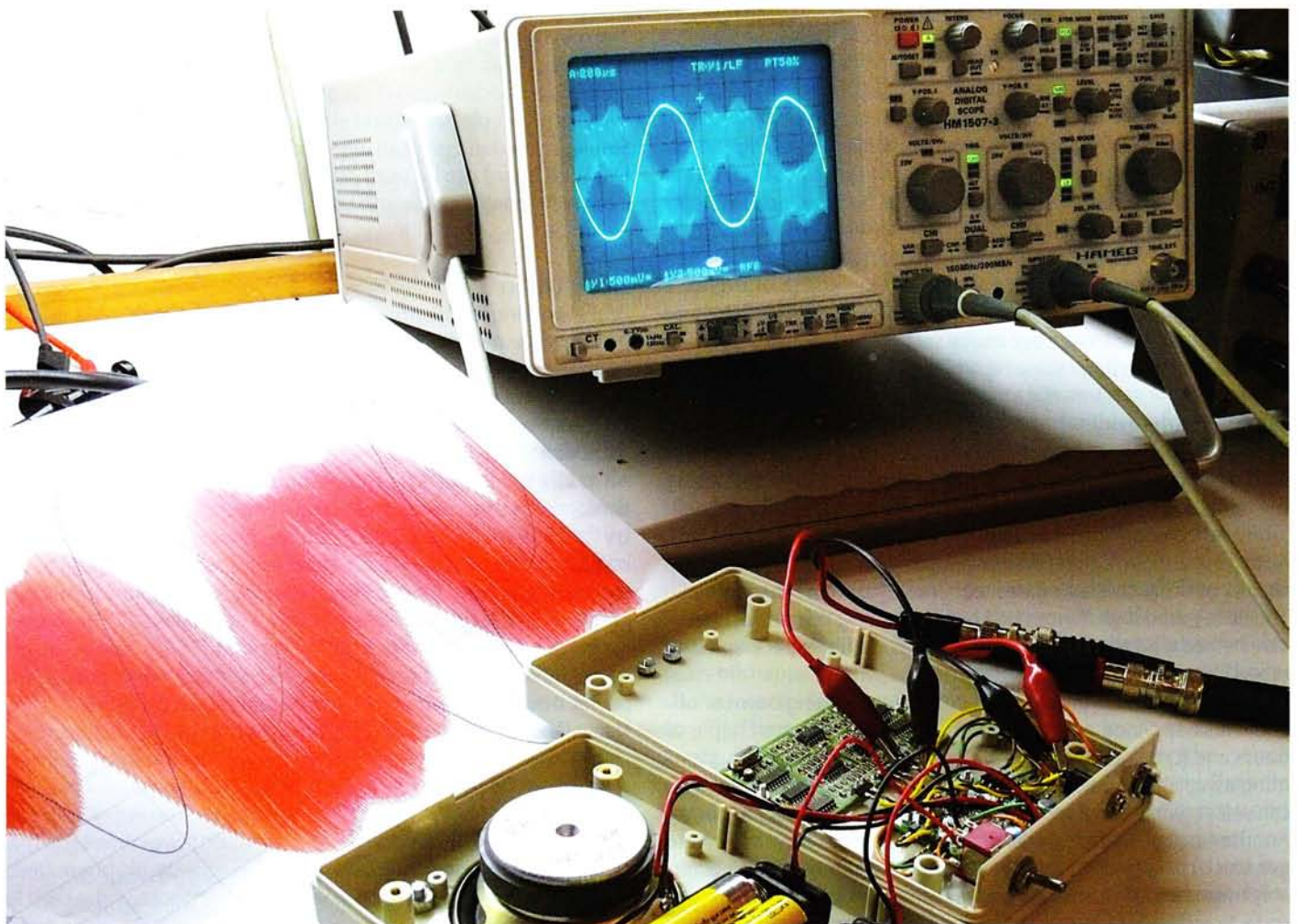
These days the design of electronic circuits happens on the PC. In a completely virtual environment the circuit diagram ('schematic') is drawn, its operation is simulated and the circuit board layout is produced. The designer puts a lot of trust in the simulation results. To be completely sure, however, a (hardware) prototype is usually built that can then be used to confirm that the circuit works as it should. But how good are those simulations really? We decided to try out Proteus, from Labcenter Electronics [1], a leading CAD package for circuit drawings, PCB layouts and simulations.

Circuit simulations: how reliable are they?

For the simulation we picked the circuit for the frequency shifter from the portable sound system with feedback suppression, which was published earlier this year in the February 2010 edition of Elektor. This circuit contains analogue as well as digital building blocks, and it isn't the easiest design to simulate, partly due to the high switching frequencies and phase shifts.

We picked several interesting points in the circuit where we would like to see the waveform of the signal; these are shown as A to F in the circuit diagram on the following page. The most interesting one is of course the output signal (F), but designers obviously also like to know what happens at intermediate stages of the circuit, so that they can be sure that the circuit behaves as expected.

CAD

To simulate this circuit we'll use the Proteus 7 Professional package from Labcenter Electronics. This package in fact consists of two programs: ISIS and ARES. ISIS is used to draw the circuit diagram and simulate it, whereas ARES is used to design the PCB layout. In this article we're not interested in the PCB layout so we'll ignore the latter program.

The Proteus Design Suite is specifically made to simulate embedded systems and has the facility to simulate microcontroller code in a mixed-mode SPICE environment. Labcenter calls this VSM, Virtual System Modelling. Of course it can also cope with 'normal' SPICE models. We aren't really asking a lot of the simulation software with our frequency shifter, but the extensive functions of the software will be welcomed by many.

Before you can start the simulation you have to input the circuit into ISIS. Unfortunately, this means that the schematic has to be completely redrawn. The various CAD packages on the market all have their own format for storing the drawings and generally aren't able to export or import other formats.

Once the circuit has been drawn, you can add probes and measuring instruments to your hearts content. Theoretically, you could add a complete arsenal of measuring instruments to each point in the circuit, but this is of course not necessary. Having selected several interesting points in the circuit and put a virtual probe there, you are in a position to visualise the signal at those points.

Simulation

When the probes have been put in place, you can choose several virtual measuring instruments, in this case three oscilloscopes and an FFT analyser. For each of the virtual instruments you have to specify which probe they have to use. Next, you can start the SPICE simulation by hovering the mouse pointer above a measurement window and pressing the space bar. Some time later (depending on the processing power of the PC) the simulated voltage appears on the screen.

Tips and tricks

The opamp models chosen for the simulation were the 'ideal' opamp models, which reduces the number of required SPICE nodes from a hundred down to only a dozen per opamp. This speeds up the simulation significantly, but still offers sufficient accuracy for this application.

The simulation for the clock generator around the crystals has also been turned off at the component level. The SPICE model for the 4060 has a parameter ('clock') allowing the operating frequency of this component to be set, irrespective of the value of the clock signal at its clock inputs. You could even have chosen to replace the 4060 and 74HC74 flip-flops with a pulse generator that output the required clock pulses, but this would have deviated too much from the original circuit.

These 'tricks' are very useful to know and give speedy and good results. It's therefore important that you know what simulation options there are and what settings give the best results. For other tips and tricks you can refer to the manual, the technical support department and last but not least, several forums where you can ask any questions you may have.

Results

The details of the simulation results are shown in the spread on the following page. We were pleasantly surprised by the fantastic results. The measured and simulated values corresponded almost perfectly. The real-world measurements only deviated slightly from the ideal (simulated) values. The slight mismatches were probably caused by component tolerances. Not only could differences be measured between a number of frequency shifters that had been built, but there was also an audible difference. Some clearly sounded better than others did.

Hats off for Proteus, which in our opinion passed the 'test' with flying colours. Not only is the end-result (signal F) amazingly correct, but all the intermediate points that have some rather 'strange' waveforms also correspond to the measured signals. No complaints here then.

Do It Yourself

On the web page created for this article [2] you can find the ISIS circuit including the virtual measuring instruments for you to download. A demo version of Proteus 7 Professional is available for download as well, so you can verify the results above and carry out further simulations if you want. The demo version of Proteus does not allow you to save your designs, nor create print layouts, nor create circuits containing microcontrollers. However, you are able to modify the program code in existing (example) designs with microcontrollers.
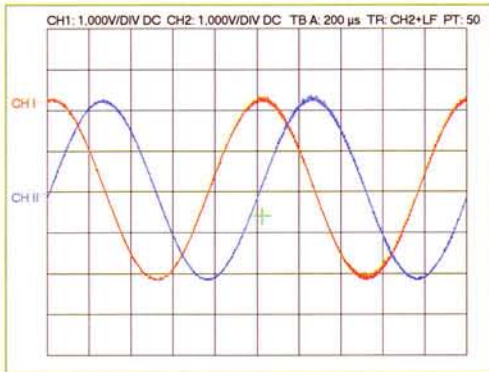
As an example, try to run a simulation of about 20 ms or more, where the input signal and the inverted output signal are shown in one graph. You should see that the signals have a slight difference in frequency and that one signal appears to 'overtake' the other (double-clicking on the title of the scope image opens up the graph in a new window).

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Hardware versus software

On the left-hand page you can see the measurement results for the measurements carried out on the real-life circuit, while the right-hand page shows the results of the simulations. On the circuit diagram a number of points (A to F) are indicated that we thought would be interesting to measure and compare with the simulation. The signals at points A and B should have a phase difference of 90 degrees

[2] Signal at C and D

At point C is the high frequency square-wave modulated signal. At point D is the filtered version of the signal at C. Due to the small difference in the fundamental frequency and the first and subsequent harmonics there doesn't appear to be much difference between C and D.
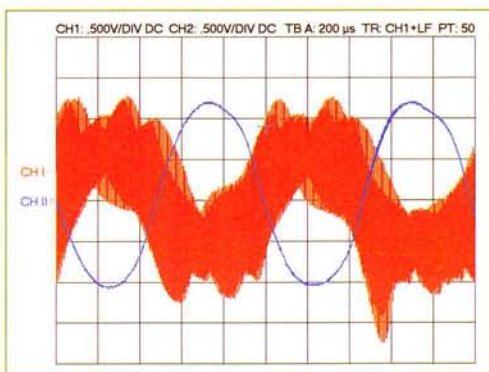

[1] Signal at A and B

The signals at these points should be the same as the input signal, but with a phase difference of 90 degrees between them.

That means that when the voltage of one of the signals is at zero, the voltage of the other signal should at its maximum or minimum (depending on the phase). This is more or less the case in the measurement; it isn't exactly 90 degrees, but it comes very close.
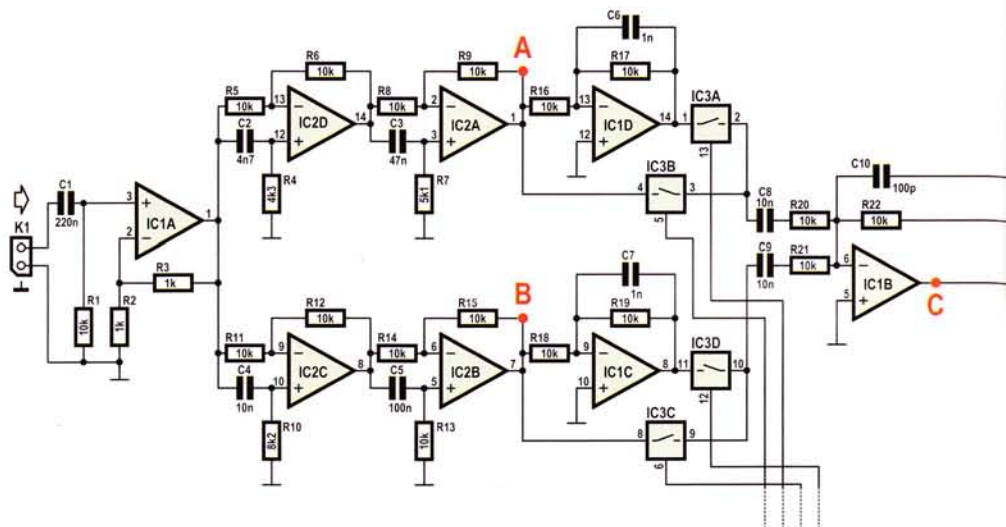

[3] Signal at E and F

At point E you can almost recognise the output signal (modulated by various mixer products). At point F all redundant RF signals have been removed from the signal and the result of the frequency shifter becomes apparent. The output sine-wave is slightly distorted, which is partly caused by the tolerances of the components used.

[4] FFT output signal

When we perform an FFT analysis on the input and output signals it's easy to see the frequency shifting that has taken place. The original signal had a frequency of 1 kHz, whereas the output signal was about 1.01 kHz, i.e. it has become 10 Hz higher, which was the intention of this 'feedback-killer'.
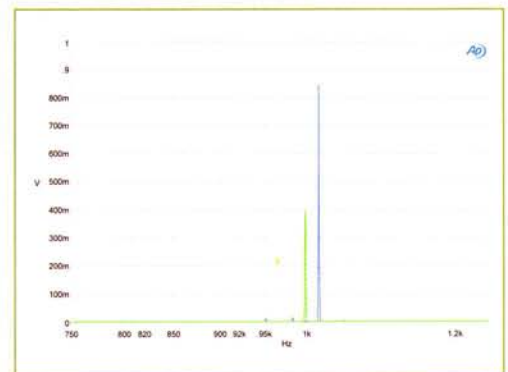
between them. C and D should be the modulated and the filtered, modulated signal respectively. At E is the output signal with a nasty looking RF signal superimposed on it and at F is the clean input signal, but with an increased frequency of about 10 Hz.

[5] Signal at A and B
In the simulation you can also see that the phase shift isn't exactly 90 degrees. It's clear that the design for the phase shifter is a compromise; the bandwidth in which the circuit works 'perfectly' is limited.
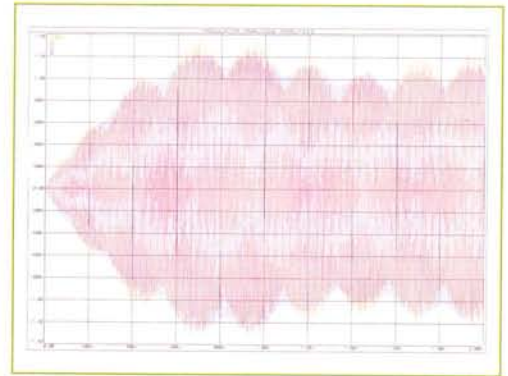
[6] Signal at C and D
Despite the complex RF waveform of the signal, the SPICE simulator from Proteus keeps its cool and shows a very close resemblance to the measured signal from the actual hardware.

Internet Links

[1] www.labcenter.com

[2] www.elektor.com/100359

[8] FFT output signal
Even the simulation of the FFT of the output signal shows an exact 10 Hz frequency shift... What more could you ask for?

[7] Signal at E and F
There is not much to say about the signal at E and the filtered output signals. The only difference is that the hardware version has some distortion due to the component tolerances, which is something that the software hasn't taken into account in this case.

Simulation Beats Trial and Error

By Dr Günter Spanner (Germany)

Simulation tools allow circuits to be tested and characterised even before the first components have been ordered. LTSpice is a powerful program and available free of charge. Here we use two examples to show how this program can be used for circuit analysis, among its other features.

Simulation programs have become indispensable for the commercial development of electronic products. The traditional development method for new products, consisting of building a lab prototype, designing an initial PCB layout and producing a pilot series before final production release testing, became outdated a good while ago.

Nowadays new circuit designs are thoroughly tested before the first real components are ordered. In addition to basic circuit functionality, this covers the full allowable operating voltage range and
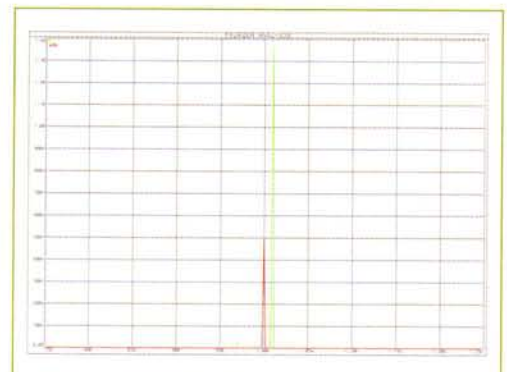
beyond – which does not incur physical risks with a simulation. The worst case situation in which the circuit must continue to operate reliably can be determined with a computer simulation by configuring a wide range of temperatures and highly improbable component tolerances.

Until a few years ago, usable simulator programs were as dear as hen's teeth, with prices of full-function versions easily reaching the three-figure range. Today tools with outstanding features are available on the Web for free download. This makes circuit simulation extremely attractive even for

non-professional users.

In this article we describe the LTSpice simulation program (previously known as SwitcherCAD), which is available free of charge from Linear Technology [1]. The core of the program, which computes the actual simulation, is based on the well known Spice simulation program, but Linear Technology has refined this program, in particular to obtain better results for switching converters (which are after all their main products). A convenient user interface has also been wrapped around the simulation software to provide features such as visual schematic design and editing. For more
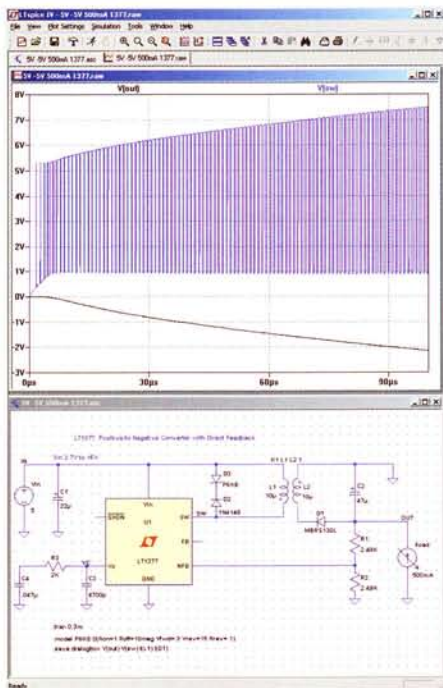

Figure 1. Convenient presentation of the schematic and the simulation results (example from the LTSpice installation package).
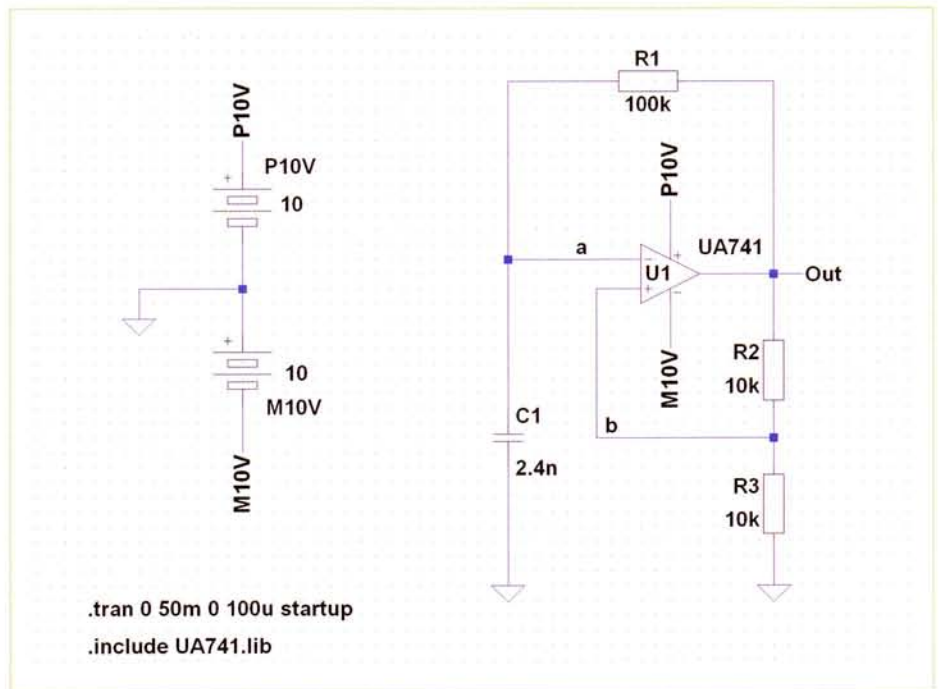


```
.tran 0 50m 0 100u startup
.include UA741.lib
```

Figure 2. A square wave generator. The opamp is simulated using a circuit not shown here; the circuit model is located in the UA741.lib file.

Practical experience with the free simulation tool LTSpice

information on Spice and a good introduction to how simulation tools work, see the October 2006 issue of Elektor [2].

Schematic editor

Installation of LTSpice is easy and does not require any explanation here. The implementation of the schematic editor is quite nice and more than justifies the use of this program. In addition to basic component symbols, the program comes with an extensive component library. Most circuit symbols are US style; a European resistor symbol is available in the library under the 'Misc.' category.

The individual components can be positioned freely and interconnected easily (a few notes for users are provided in an extra file that can be downloaded from the Elektor website [3]). All components can be rotated and flipped as necessary to produce a readily understandable schematic diagram. There is also a component editor that allows users to generate their own component symbols.

Semiconductor devices in particular are composed of a large number of internal components, so that their behaviour can be simulated as realistically as possible. These subcircuits are represented by Spice models, and the necessary data can often be obtained free of charge from the manufacturer or elsewhere on the Web. As a Spice derivative, LTSpice naturally supports integration of these Spice models.

Figure 1 shows a typical working environment for circuit analysis. The schematic is shown in the bottom window, while the simulation results are displayed in the top window. The inverting voltage converter shown here is an example circuit included in the library of the installation package.
To find out what this simulation program can do, we tested it with two relatively simple example circuits.

Example 1: Square wave generator

Our first example simulation is a simple square wave generator. The circuit shown in Figure 2 is built around a standard opamp
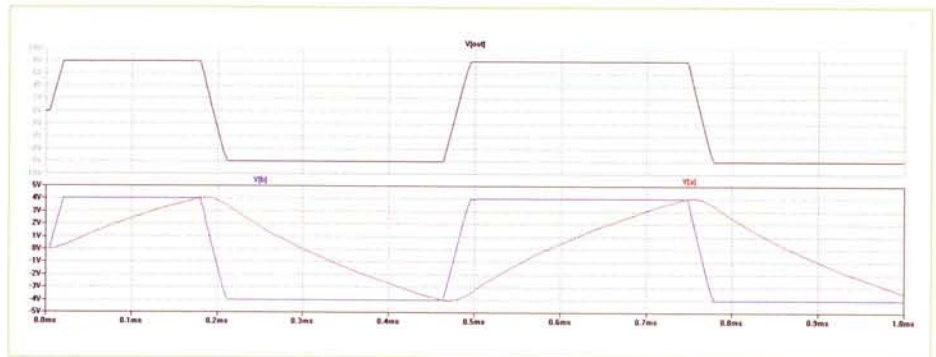

Figure 3. The waveforms at test points 'a', 'b' and 'out' (see Figure 2).

and should be familiar to every electronics enthusiast.

After the schematic is entered, the model for the operational amplifier (a 741), which can be found at various places on the Web including [4], must be saved in the ROOT:\ Program Files\LTC\LTspiceIV\lib\sub folder. There it is available to all simulations. Alternatively, you can select an opamp type available in the standard library.

After you start the simulation, you can use a virtual probe to measure voltages, currents, power levels and other quantities at various points in the circuit. The corresponding waveforms are shown graphically.

Figure 3 shows a virtual oscilloscope display of the signals at three test points:

a: the inverting input of the opamp
b: the non-inverting input of the opamp
out: the output of the opamp

From the measured output waveform, you can see right away that the 741 (as is well known) is not a rail-to-rail type; the output saturation voltage is around 20% lower than the supply voltage. In addition, the slew rate of 0.2 V/µs is depicted realistically.

Test point 'a' shows the voltage on the 2.4-nF capacitor; it is a slightly bowed triangle waveform with alternating positive and negative slopes. A square wave signal can be seen at test point 'b' (non-inverting input). Due to voltage divider R2/R3, its amplitude is exactly half that of the signal at the 'out' test point.

With the aid of this simulation, circuit

operation can be explained to relatively inexperienced users. When the voltage on the capacitor rises above or falls below the reference voltage at the node of voltage divider R2/R3, the opamp output changes state (switches from the positive saturation level to the negative saturation level or vice versa). As a result, the capacitor is periodically charged and discharged.

Example 2: Percussion generator

The behaviour of the second circuit (Figure 4) is less obvious, even for experienced designers. The reverse feedback is provided by a network in this circuit. If you view the
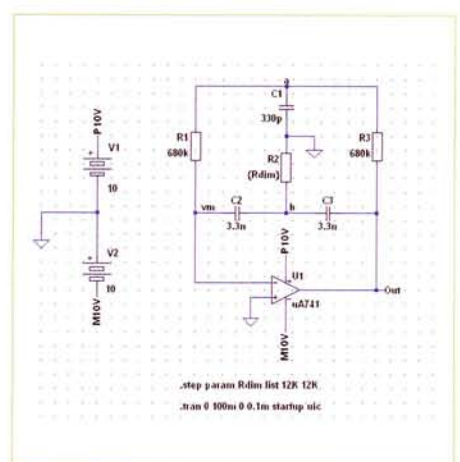

Figure 4. The behaviour of this circuit is not as obvious as that of the first example.
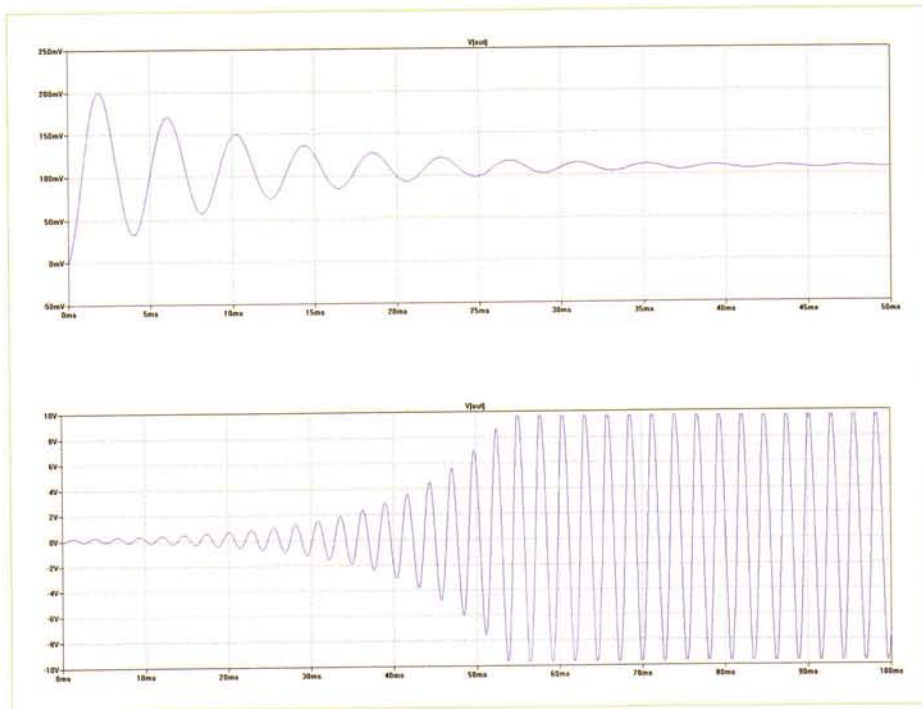
Figure 5. The attenuation of the oscillation waveform depends on the value of resistor R2 (see Figure 4). Values below the critical value result in a waveform with increasing amplitude.
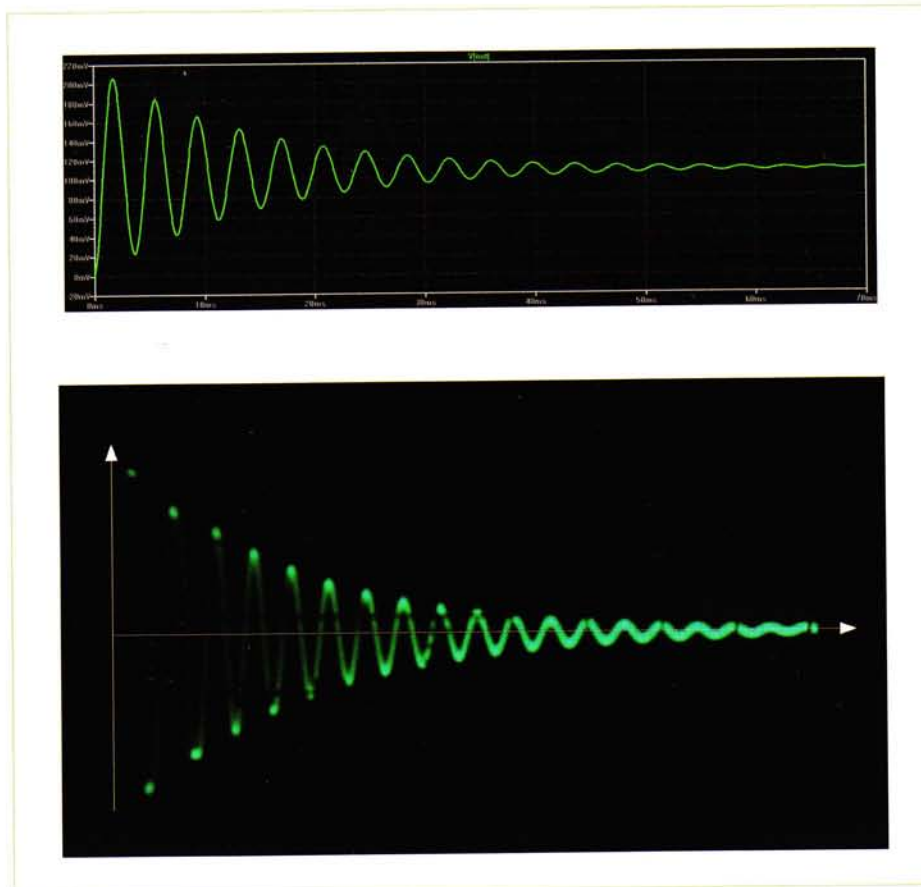

Figure 6. A perfect fit:
simulation result versus waveform measured with real hardware.

simulation results, you can clearly see that this network has a rather unusual property. Two simulation results are shown in Figure 5. For the upper curve, the value of resistor R2 was set above the critical value (around 17 kΩ with the indicated component values). This circuit generates an exponentially decaying sine wave signal after the supply voltage is switched on. This also explains the name of the circuit: if the output signal is converted into an acoustic signal, it sounds like a light tap on a drum head. However, if the value of R2 is reduced below the critical value, the circuit produces an exponentially increasing sine wave signal as shown in the lower curve, which is ultimately limited by the maximum output voltage of the opamp.

If the value of R2 could be set to exactly the critical value Rcrit, the circuit would generate a sine wave signal with constant amplitude. However, this cannot be achieved in practice.

Figure 6 shows a comparison between the simulation result and an oscilloscope signal recorded with an actual implementation of this circuit. These images demonstrate the outstanding quality of the simulation software.

To further illustrate the extensive features of the simulation tool, a parametric plot is shown in Figure 7. It represents the output voltage versus the voltage at test point 'b'. You can immediately recognise the phase shift between the two signals and the time decay of the signal amplitudes.

Other useful functions

In addition to transient analysis (time domain analysis), you can generate an AC analysis with LTSpice. This is useful for examining the frequency response of a network.
For this analysis, a tuneable signal generator must be connected to the input of the network. Here again, test points can be defined with the virtual probe. Figure 8 shows the frequency response of the feedback network of the percussion generator circuit. The middle curve illustrates the situation

when R2 has the critical value, which yields the narrow impedance notch clearly visible here. The curves to the left and right illustrate the situations for oscillation with decaying or increasing amplitude.

Even more complex circuits can be studied in detail with the aid of a simulation program such as LTSpice. The virtual measurement results are usually astonishingly close to the results measured with the real hardware.

However, you should always be aware of the limitations of computer simulations. Although programs such as LTSpice have reached a high level of maturity, aberrations can always occur in practice. In particular, it takes considerable effort to fully model ambient conditions such as electromagnetic interference, radiated high frequency interference and noise components in simulations.

Finally, you should always bear in mind that Murphy's Law also applies to simulations: a circuit that works perfectly as a simulation can always fail in practice.

On the other hand, a circuit that doesn't work properly as a simulation is unlikely to prove satisfactory in real hardware.

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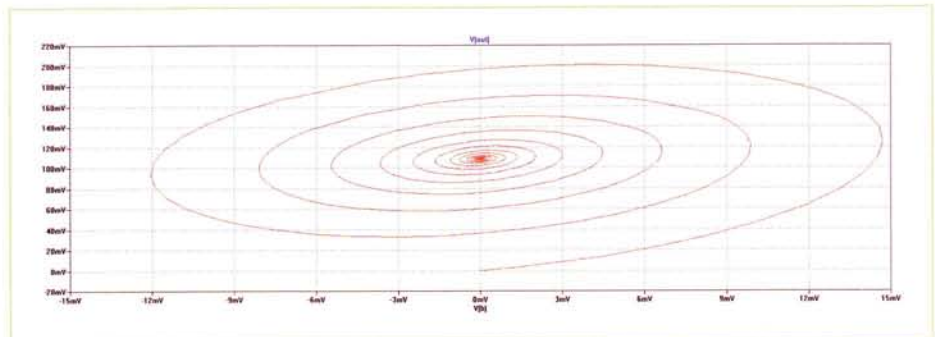
Figure 7. A parametric plot of the output voltage versus the voltage at test point 'b'. The phase shift is clearly visible here.
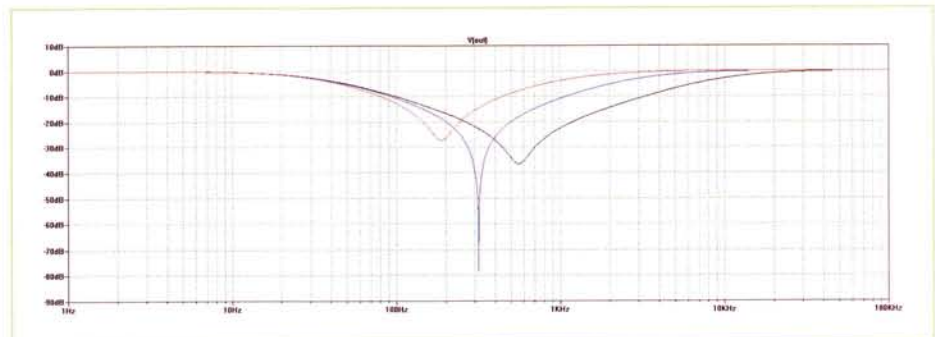
Figure 8. LTSpice can also show the frequency response of a network (in this case the feedback network of the circuit in Figure 4).

Internet Links

[1] www.linear.com/designtools/software/ltspice.jsp

[2] www.elektor.com/060207

[3] www.elektor.com/060206

[4] http://focus.ti.com/docs/prod/folders/print/ua741.html#technicaldocuments

Digital Multi-Effects Unit

With 15 configurable sound effects and a user-friendly HMI

By Hubert Bollig (Germany)
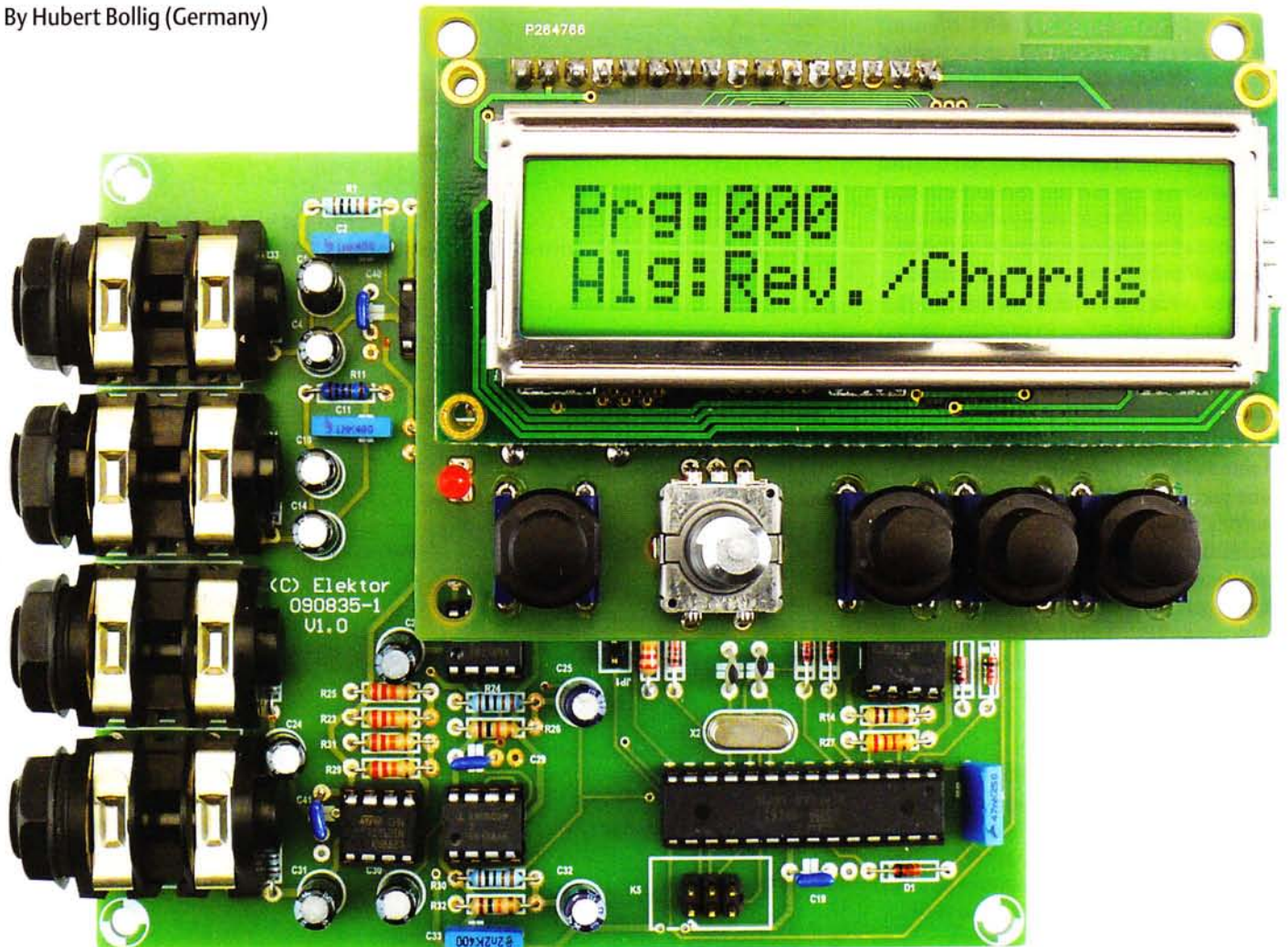
It's a simple fact: every recording sounds better with the right sound effects. Here we prove that it's possible to generate a variety of effects digitally, including reverb, chorus and flanger effects, without having to work yourself to the bone with DSP programming. The circuit is built around a highly integrated effects chip and features an intelligent user interface with an LCD. The result is a treat for the eye and the ear.

Sound effects are what gives music that special touch. An effects generator is an indispensable element of the kit of every modern musician, both professional and amateur. In the mid-1980s, the only components available to designers of effects units were bucket-brigade ICs and opamps, but in the course of the digital revolution these components have increasingly been replaced by digital signal processor (DSP) ICs. The growing market presence of these

Features

- Multi-effect generator with 15 effect algorithms
- Based on the FV-1 effect processor IC
- Frequency response 20 Hz to 15 kHz
- Maximum delay time 700 ms
- 64 memory locations for preset configuration profiles
- Separate user interface
- MIDI In port
- Effect selection and parameter control via MIDI port
- Integrated ramp generator for producing attack/decay effects
- Easy assembly using standard components (except the FV-1 IC)
- Free assembler program for generating user-programmed effect algorithms

highly integrated ICs spelled an end to the DIY era for electronic enthusiasts with an interest in music, as the components and the necessary software simply became too complex. In addition, DSP ICs are not readily available and are almost impossible to solder by hand.

Everything in one IC

Fortunately, this unsatisfactory situation has changed in the last few years. This probably results from the fact that sound effect capability is being built into more and more mixing panels, stomp boxes and other musical equipment. Application-specific ICs such as the FV-1 IC [1] from Spin Semiconductors [2] represent a relatively economical option for equipment manufacturers.

All the components necessary for building an effects unit are already integrated in the IC: high-quality audio A/D and D/A converters, delay RAM (for delaying audio signals), four LFOs (two sine wave and two ramp), and three auxiliary audio inputs. As you can see from the block diagram (Figure 1), these components are grouped around a DSP core with a 24-bit ALU. The DSP does not require user programming, since seven algorithms (for corresponding effects – hall, chorus, flanger, etc.) are stored in ROM by the manufacturer. Another eight effect algorithms can be held in an external EEPROM and input via the I2C port. Users can generate these algorithms themselves with the aid of a special assembler (SpinAsm) [3]. A development environment and various examples can be downloaded free of charge.

Now you may be thinking that such a special IC is virtually impossible to obtain for personal use, but that's not the case. There are three sources that also sell to private parties: the online shops of Profusion PLC in the UK [4], Small Bear Electronics in the USA [5], and Das Musikding in Germany [6].

Instead of spending several months wrestling with DRAM timing, special setup registers and so on, you can obtain the full functionality of an effects unit in the form of a single IC. This IC is intended to be used with a user interface consisting of a rotary switch for selecting the effect algorithm

and three potentiometers for adjusting the effect parameters. However, our aim here is to achieve more than the basic circuit has to offer.

User-friendly operation

We want our effects unit to have the following convenience features:

- LCD display
- an EEPROM for holding eight supplementary effect algorithms (as previously mentioned)
- a ramp generator for attack and decay effects
- 64 preset profiles for user-defined effect settings
- and of course, a MIDI interface for musicians who shudder at the thought of

buttons (which includes the author)

For this reason, in our design a microcontroller looks after the tasks that would otherwise be handled by a rotary switch and a set of three potentiometers. Digital selection of the effect algorithm and generating the analogue voltages for configuring the effects are easy tasks for a microcontroller. For this job we chose an Atmel ATmega8. It has an integrated EEPROM, which is ideally suited to storing the preset configuration settings.

The user interface with its LCD module and buttons is housed on a separate PCB, which provides several advantages. If you want to install the effects unit in an existing synthesizer or amplifier, it's easy to fit the user
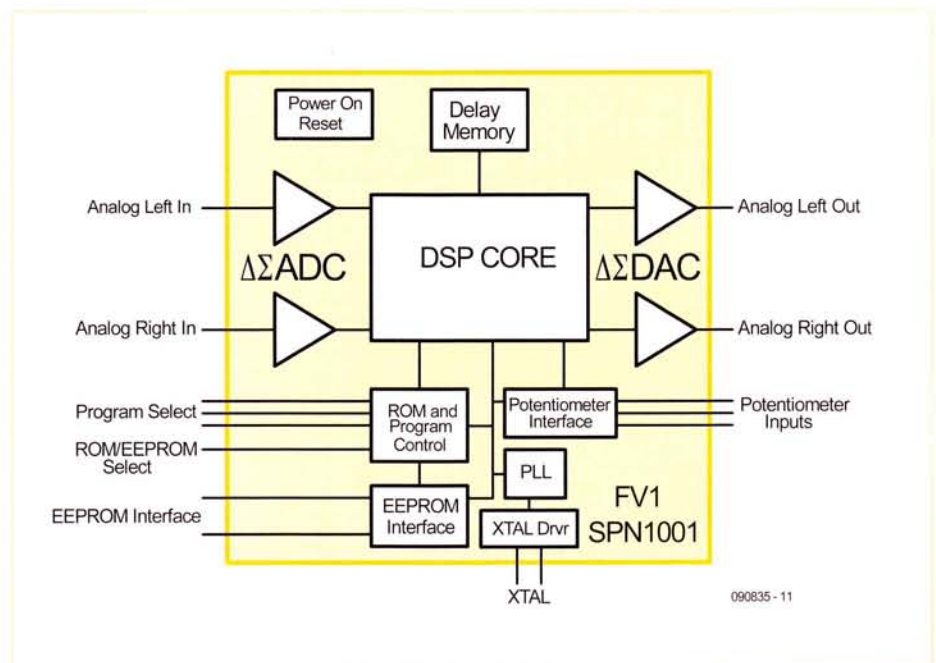

Figure 1. The FV-1 IC is sound effects processor featuring a DSP core, ADC, DAC and additional components. Seven predefined effect algorithms are stored in ROM.
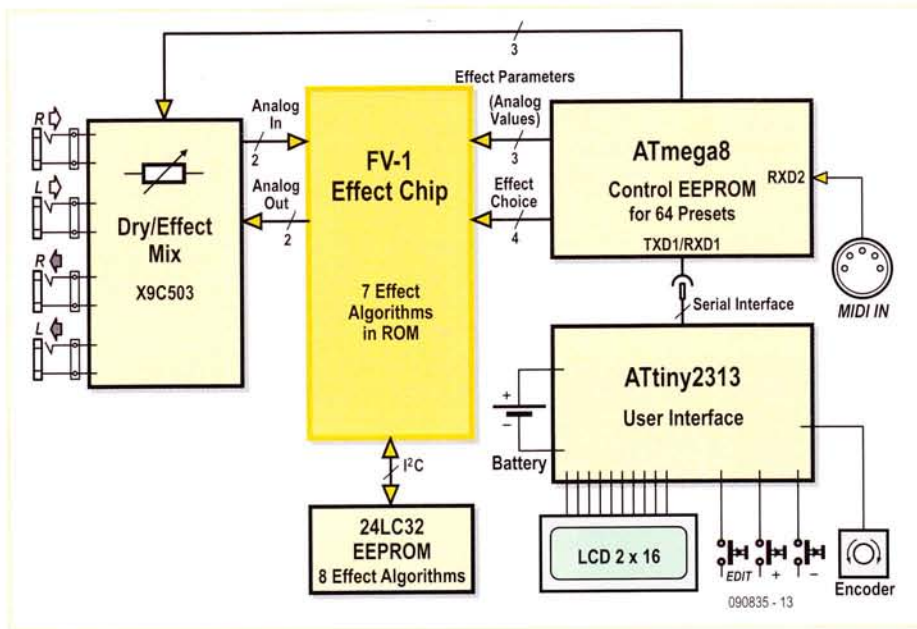
Figure 2. The ATmega microcontroller controls the effects processor IC via four digital lines and three analogue lines. The user interface has its own microcontroller.

any I/O pins left for the controls and indicators, so some form of I/O expansion was anyhow necessary. The end result is that the overall circuit offers maximum flexibility for changes and extensions, and if you wish you can also use the user interface for other purposes.

The EEPROM for the additional effects can also be seen in the block diagram of the circuit (Figure 2). The block at the left in this diagram allows the original (dry) signal to be mixed with the signal emerging from the effects processor. In line with the design philosophy of the unit, dry/effect mixing is also controlled by the ATmega with the aid of two digital potentiometers.

interface board in a separate enclosure. The connection in this case is provided by a three-way cable, and there's no need to disfigure the front panel of your sound gear. This approach also simplifies the layout of the main PCB, but the key consideration is that we gave the user interface its own intelligence in the form of an ATtiny microcontroller. Here the author made a virtue out of a necessity: the ATmega did not have

The circuit

Armed with the previous description, it's fairly easy to understand the operation of the circuitry on the main board (Figure 3). The audio signals from the input connectors are fed directly to the inputs of the FV-1 IC via coupling capacitors, and to the inputs of the buffer amplifiers for dry/effect mixing (IC1a and IC1b). The outputs of IC1a and IC1b are fed to the inputs of the electronic potentiometers (X9C503). The output signals of the effect processor are fed to the other inputs of the digital potentiometers. The signals from the potentiometers pass through buffer amplifiers on their way to the output connectors.

The external circuitry of the FV-1 IC corresponds to the recommend circuit of the manufacturer's data sheet. Capacitors C3 (100 nF) and C17 (10 µF) must be located as close as possible to corresponding pins of the IC. LED D2 is a clipping indicator, which lights up when the internal ADC or DAC is close to saturation or in saturation [1]. It should also be mentioned that the DSP operates with a supply voltage of 3.3 V. The effect algorithm is selected by a 4-bit signal applied to pins T0 and S0–S2. The level on T0 determines whether an internal effect algorithm (hard-coded in ROM) or an external algorithm (stored in the 24LC32) is to be used. Diodes D1 and D5–D7 together with resistors R7 and R19, R20, R21 are simple level converters from 5 V to 3.3 V. Diodes D9 and D10 are reserved for future
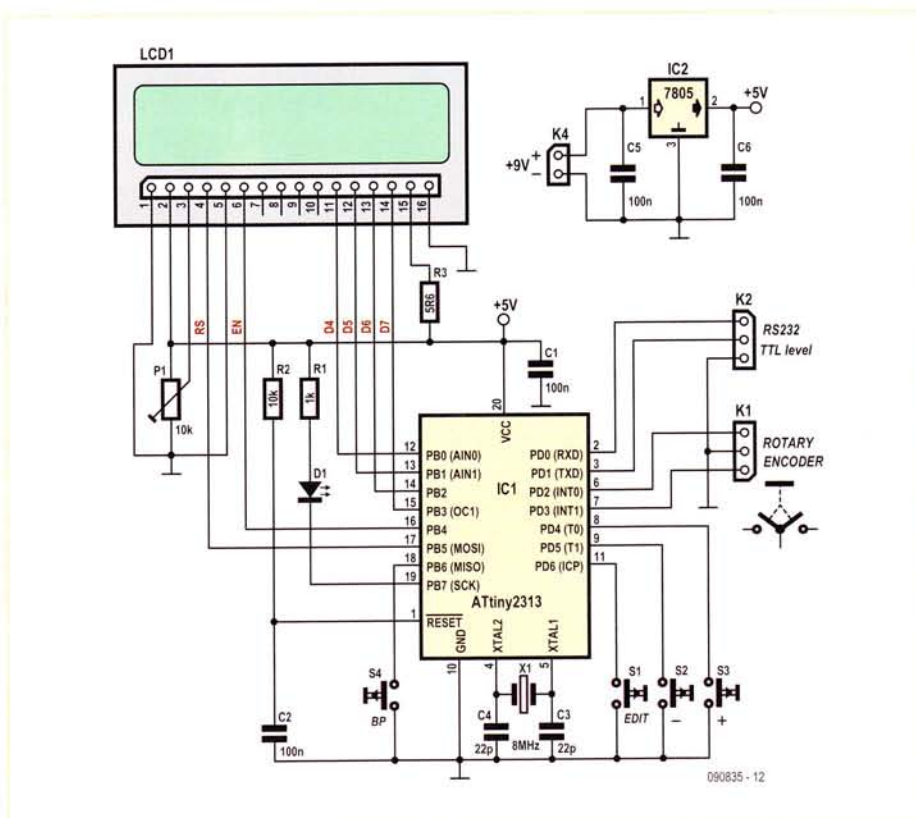

Figure 4. Schematic diagram of the user interface. The LCD module is operated in 4-bit mode, and connector K2 is connected to K3 on the main board.

extensions.

The three analogue inputs POT0–POT2 can be used to adjust the specific effect parameters of each effect algorithm. The analogue voltages for POT0–POT2 are generated by the ATmega in the form of PWM signals. For this purpose, the 5-V PWM signals are first reduced to 3.3 V by voltage dividers (R9/R10, etc.) and then converted to DC voltages by low-pass filters.

The effect type and effect parameter settings, along with the dry/effect mix setting, are combined to form a preset profile and stored in the internal EEPROM of the ATmega8. The 512 bytes of EEPROM memory provide exactly enough space for 64 preset profiles. JP2 is provided for write protection (write protection is enabled when pin 7 is tied to VCC and disabled when pin 7 is not tied to VCC).

Effect selection

There are two options for sending commands to the ATmega to select a preset profile:

• The serial port (K3) and associated user interface
• The MIDI port (K7)
•

The implementation of the MIDI port with an optocoupler is standard. The only thing that should be mentioned here is that the optocoupler used for this (a 5N139) is simpler and less expensive than the PC900 stipulated by the MIDI specification. The serial MIDI data is clocked into the INT0 input at 31,250 baud.

The user interface is connected to the other serial port (RxD/TxD). Diodes D3 and D4 allow the main PCB and the user interface PCB to be powered from separate supply voltages. For example, the user interface may operate under battery power. Of course, it is also possible to power the user interface from the main PCB.

The circuit of the user interface is straightforward. The LCD module is operated in 4-bit mode. The input components consist of four buttons and a rotary encoder, which is connected to K1. Pay careful attention to
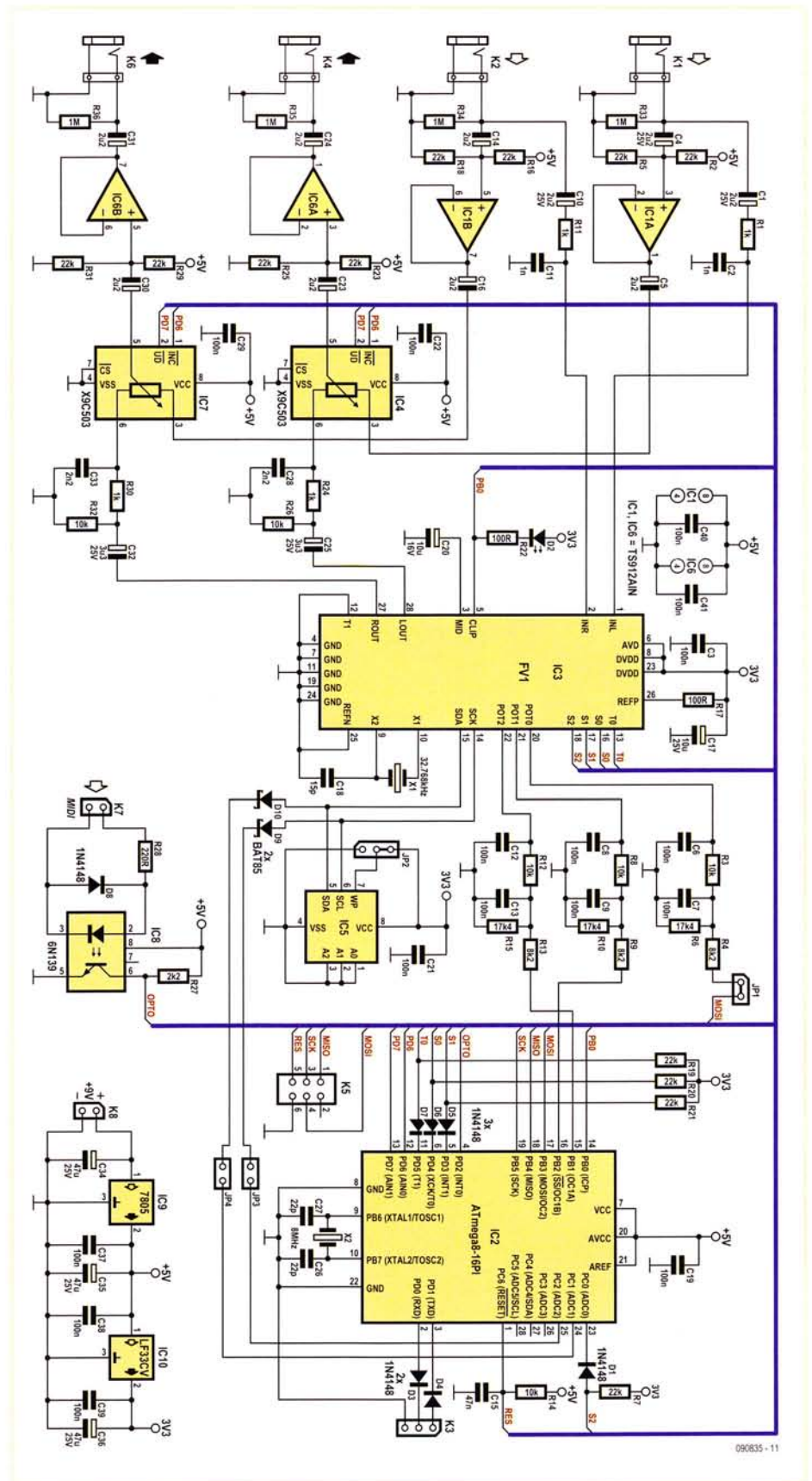

Figure 3. Schematic diagram of the main board, with the buffer amplifiers for the signal inputs and outputs and the digital potentiometers for dry/effect mixing shown on the left. The three resistor/capacitor networks in the middle of the diagram convert 5-V PWM signals into DC control voltages with a range of 0 to 3.3 V.

Figure 5. Component layout of the main board. The FV-1 IC is the only SMD component.

Figure 6. User interface PCB.

COMPONENT LISTS

Main Board

Resistors
R1,R11,R24,R30 = 1kΩ
R2,R5,R7,R16,R18,R19,R20,R21,R
  23,R25,R29,R31 = 22kΩ
R3,R8,R12,R14,R26,R32 = 10kΩ
R4,R9,R13 = 8.2kΩ
R6,R10,R15 = 17.4kΩ (1 %,
  250 mW)
R17,R22 = 100Ω
R27 = 2.2kΩ
R28 = 220Ω
R33,R34,R35,R36 = 1MΩ

Capacitors
C1,C4,C5,C10,C14,C16,C23,C24,C
  30,C31 = 2.2µF 16V
C2,C11,C28,C33 = 1nF 400V
C3,C6,C7,C8,C9,C12,C13,C19,C21
  ,C22,C29,C37,C38,C39,C40,C41
  = 100nF, ceramic
C17,C20 = 10µF 16V
C15 = 47nF
C18 = 1nF, ceramic)
C25,C32 = 3.3µF 16V
C26,C27 = 22pF
C34,C35,C36 = 47µF 16V

Semiconductors
D1,D3,D4,D5,D6,D7,D8 = 1N4148
D2 = LED, 3mm, low current
D9,D10 = BAT85
IC1,IC6 = TS912, dual opamp,
  rail to rail (ST Microelectronics
  TS912IN)
IC2 = ATmega8-16PU, pro-
  grammed, Elektor order #
  090835-41 [8]
IC3 = SPN1001-FV1, Spin
  Semiconductor
IC4,IC7 = X9C503 electronic po-
  tentiometer (Xicor X9C503P,
  Farnell: 179485)
IC5 = 24LC32 (Microchip 24LC32A-
  I/P), programmed, order #
  090835-31 [8]
IC8 = 6N139 Optocoupler (Vishay
  Semiconductor)
IC9 = 7805
IC10 = LF33CV (ST
  Microelectronics)

Miscellaneous
X1 = 32.68kHz quartz crystal
X2 = 8MHz quartz crystal
K1,K2,K4,K6 = 2-way screw termi-
  nal block, PCB mount
K3 = 3-pin pinheader, lead pitch
  0.1 in.
K5 = 6-pin (2x3) pinheader, lead
  pitch 0.1 in.
K7,K8 = 2-pin pinheader, lead
  pitch 0.1 in.
JP1,JP3,JP4 = 2-pin pinheader, lead
  pitch 0.1 in., with jumper
JP2 = 3-pin pinheader, lead pitch
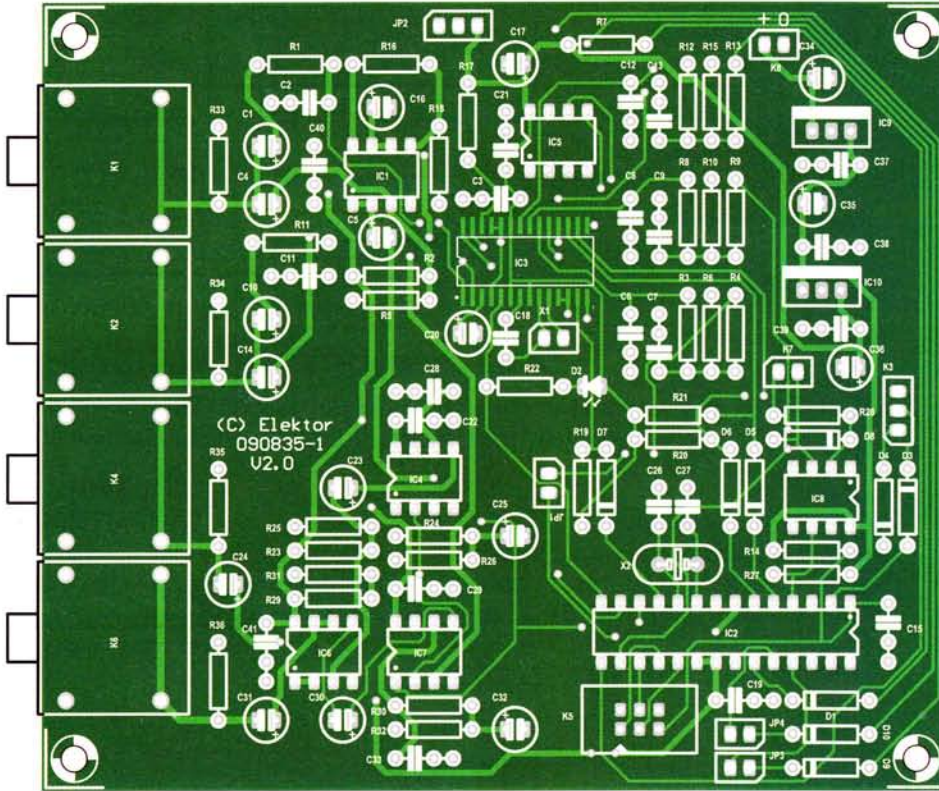 0.1 in., with jumper

PCB # 090835-1, see [8]
or
Kit of parts # 090835-71, contains alll compo-
nents including boards and programmed mi-
crocontrollers, EEPROM, see [8]

User Interface
Resistors
R1 = 1kΩ
R2 = 10kΩ
R3 = 5.6Ω
P1 = 10kΩ trimpot

Capacitors
C1,C2,C5,C6 = 100nF
C3,C4 = 22pF

Semiconductors
D1 = LED, 3mm, low current
IC1 = ATtiny2313-20PU, programmed, Elektor
order # 090835-42 [8]
IC2 = 7805

Miscellaneous
X1 = 8MHz quartz crystal

K1 = Rotary encoder, e.g. Alps type
EC11E15204aE
K2 = 3-pin pinheader, lead pitch 0.1 in.
K4 = 2-pin pinheader, lead pitch 0.1 in.
S1,S2,S3,S4 = pushbutton, Multimec type
3FTL6
LCD1 = LCD 2x16 characters, Displaytech
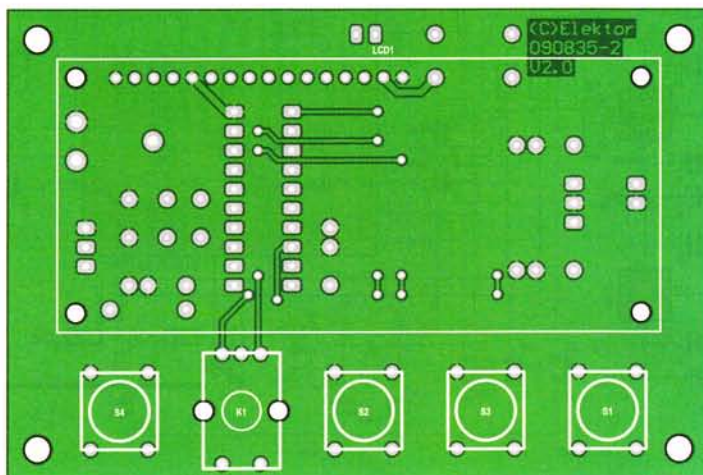type 162C
PCB, order # 090835-2, see [8])
or
Kit of parts # 090835-71, contains alll com-
ponents including boards and programmed
microcontrollers, EEPROM, see [8]

the required pin assignments when select-
ing a component for this (check the data
sheet).

Assembly and programming
To simplify assembly, a kit with all the nec-
essary components (including the PCBs,
pre-programmed microcontrollers and a
programmed EEPROM) is available from the
Elektor Shop. The PCBs (Figures 5 and 6),
the pre-programmed microcontrollers and
the programmed EEPROM are also available

separately from the Elektor Shop.
All components except the FV-1 IC are
leaded types. The only SMD is the effects
processor, but it is relatively easy to solder
by hand since the pin spacing is 1.27 mm.
If you cannot find 17.4 kΩ resistors for R6,
R10 and R15, you can use 15 kΩ and 2.2 kΩ
resistors connected in series.

The source and hex files for programming
the microcontrollers and the hex files for
programming the EEPROMs (external

and internal to the ATmega) are available
on the web page for this article [8]. The
EEPROM file for the ATmega contains ini-
tialisation values for 63 preset profiles. As
downloading program code to the flash
memory of the ATmega also erases the
internal EEPROM, the EEPROM must be pro-
grammed last.
The flash memories of the AVR microcon-
trollers can be programmed with an STK500
development board (among other options),
which can also be used for writing data to

Operation

A total of 15 effect algorithms (effect types) are available. The name of the currently selected effect algorithm is shown on the first display page. Each effect algorithm has three specific effect parameters, which are combined to form a preset profile. The unit can store up to 64 preset profiles in memory.

The dry/effect mix ratio and ramp generator setting are also stored in each preset profile.

The manual included in the download file [8] contains a table of all effect algorithms and their configurable parameters.

After the unit is switched on, profile '000' is loaded and shown in the first line of the display. The currently selected effect algorithm is shown in the second line of the display.

The next or previous preset profile can be selected by pressing the '+' and '–' buttons.

The Bypass (BP) button takes the effect unit out of the audio signal path.

The rotary encoder can be used to adjust the parameter values. The currently selected parameter is indicated on the display by a blinking cursor. Clockwise rotation increases the parameter value, while anticlockwise rotation reduces the value.

Press the Edit button to jump to the next parameter page or select the next parameter. If you have already reached the final parameter, pressing the Edit button takes you back to the first display page.

To save a preset profile, first press and hold the Edit button and them press the '+' button.

If you do not wish to save a setting, simply press the '+' or '–' button. Any changes you have made to the profile up to this point will be lost.

The downloadable manual also includes information on controlling the unit via the MIDI interface.

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the internal EEPROM. An ISP port (K5) is also provided for the ATmega to enable in-circuit programming and downloading of firmware updates. Jumper JP1 must be removed in order to program the microcontroller via the ISP port.

You need an EEPROM programmer for the external EEPROM, but you can also manage with a bit of supplementary hardware and the PonyProg program [9]. The Pony-Prog website has a design for a simple parallel port to I2C adapter circuit, which can easily be assembled on a piece of prototyping board. Incidentally, this handy program is also suitable for programming the AVR microcontrollers.

The zip file, which (as usual with Elektor projects) can be downloaded free of charge, also includes a Read Me file with information to help you sort out the hex files, as well as several screen shots showing the correct fuse settings.

Initial operation

As usual, you should power up the circuit for the first time with the IC sockets empty. Check the supply voltages (5 V and 3.3 V) at the IC sockets. Next, fit the ICs in their sockets and connect the user interface to the main board. The welcome message should appear on the LCD module after power is switched on. Press the Edit button to navigate to the settings screen for the first parameter. Measure the voltage on the POT0 terminal. It should be possible to adjust it over the range of 0 to 3.3 V with the rotary encoder. It should never exceed 3.5 V under any conditions.

The Clip LED (D2) should light up briefly when power is switched on. The circuit is now ready for use, and you can connect a line level signal to its input.

The operating instructions are summarised in the inset. A detailed user guide (in German) is included in the download file [8].

(090835-I)

Internet Links

[1] www.spinsemi.com/Products/datasheets/spn1001/FV-1.pdf

[2] www.spinsemi.com

[3] www.spinsemi.com/Products/datasheets/spn1001-dev/SPINAsmUserManual.pdf

[4] www.profusionplc.com

[5] www.smallbearelec.com/home.html

[6] www.musikding.de/

[7] http://ww1.microchip.com/downloads/en/DeviceDoc/21713g.pdf

[8] http://www.elektor.com/090835

[9] www.lancos.com/prog.html

The Elektor DSP Radio (2)
Antennas and PC software

By Burkhard Kainka (Germany)

An important advantage of the Elektor DSP radio over other world receivers is the flexibility the user has to change its characteristics, both by modifying the antenna and front-end circuit configuration and by modifying the software. The unit can be used in its various configurations as a PC-controlled radio or as a portable world receiver.
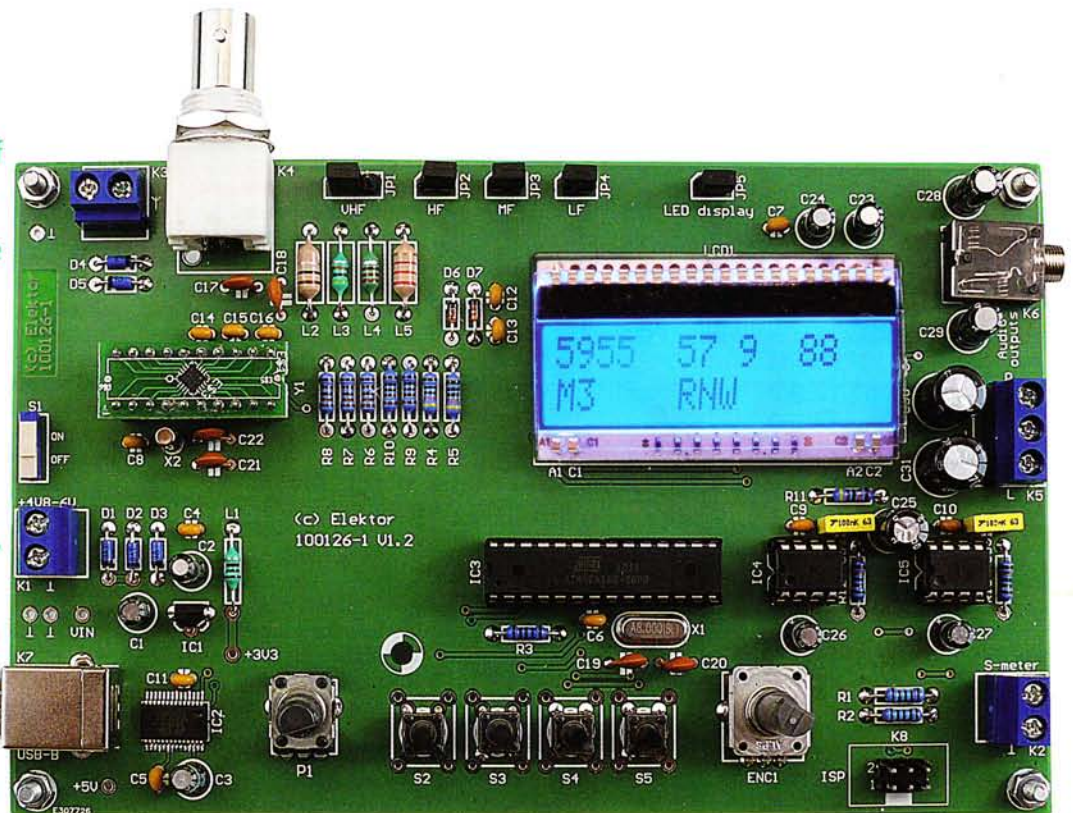
The circuit diagrams shown here illustrate the input circuit of the receiver we published in the first article in this series. For each possible antenna configuration the relevant jumper settings appear in red and the connected antennas and their accompanying coils appear in green.

Figure 1 shows the standard configuration of the front-end circuit, with all jumpers in their default positions. Switching diodes D6 and D7 are represented symbolically as shortwave and mediumwave switches. You can see from the diagram how in each case a part of the inductance in the circuit is shorted. A common antenna is used for all frequency ranges: one example of a suitable design is a 50 cm whip. Another possibility is to use an outdoor antenna: the author has successfully used a 10 m longwire antenna connected via 30 m of coaxial cable. Although this antenna is not ideal

for FM reception, it nevertheless gave good results over all frequency bands from longwave to FM.

If separate antennas are to be used for FM and AM reception, JP1 can be fitted to take the bottom of FM coil L2 to ground. The open pin at JP1 (connected to L3 and C15) then forms the new AM antenna input for long-, medium- and shortwave (Figure 2). For indoor operation on long- and mediumwave ferrite antennas are often better than wire antennas as they have greater immunity to the types of electric field interference often found inside houses. Figure 3 and Figure 4 show two alternative possibilities. In the first example the coils on the ferrite rod form an additional part of the receiver circuit, while in the second example they replace the fixed inductors on the receiver board, which are taken out of circuit by removing jumpers JP2, JP3 and JP4.

The suggested turn counts given are suitable for use with a 10 mm diameter ferrite rod with a length of between 9 cm and 15 cm. The automatic tuning function of the receiver means that the exact inductance value is not critical. Experiments show that the ferrite rod provides good signal amplitude and interference rejection at frequencies of up to 10 MHz, and this is why both circuit variations also include a shortwave coil on the ferrite rod.

A further interesting option is the tuned loop antenna shown in Figure 5. A wire loop with a total length of 4 m takes the place of JP2. It is possible to make the loop smaller, and the wire length and shape can also be varied. Again, because of the automatic tuning, high signal levels can be achieved, approaching the performance possible with longwire antennas. The radio's display will show the capacitance in the tuned circuit:
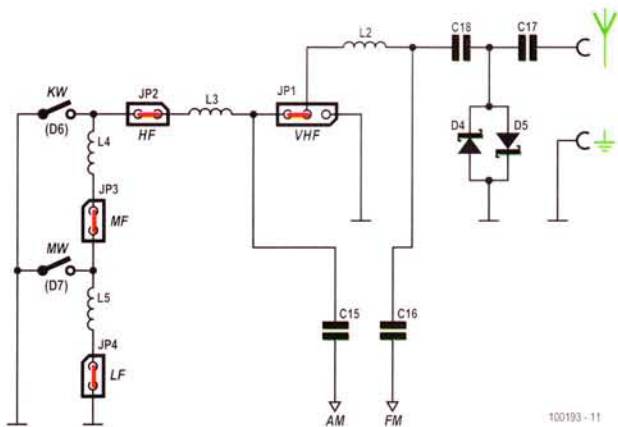
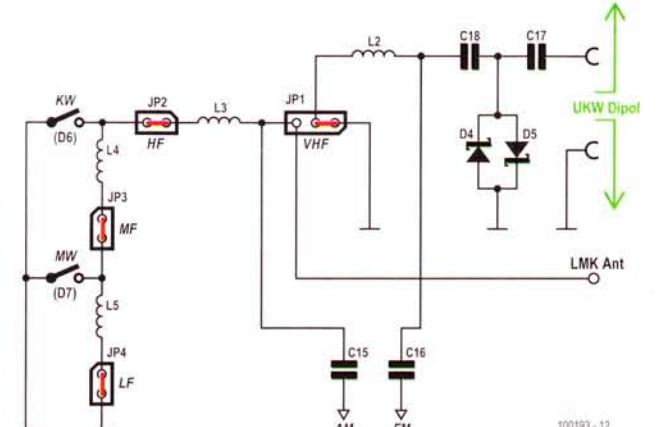Figure 1. Standard configuration of the front-end circuit.

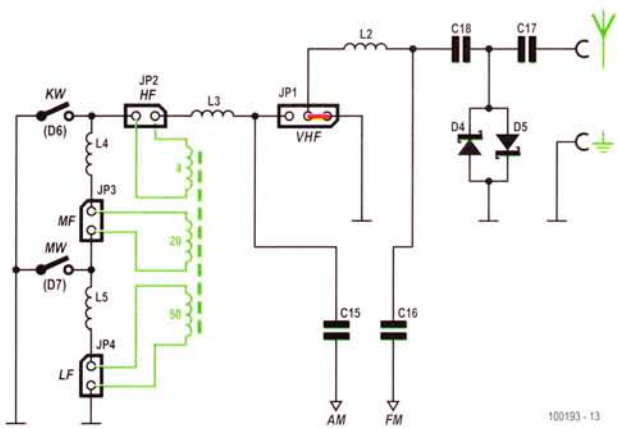Figure 2. Separate connection of FM and AM antennas.

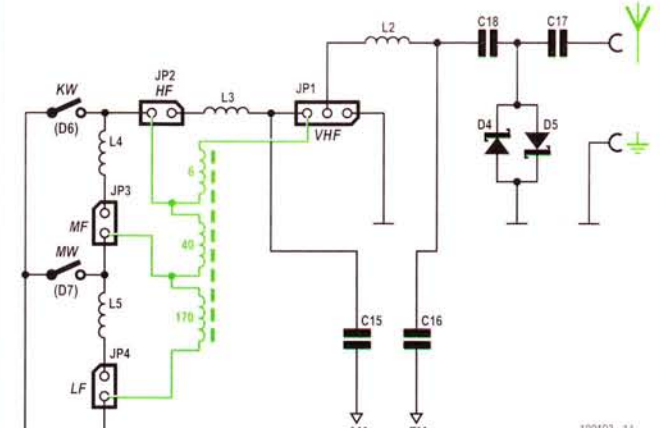Figure 3. Connecting a ferrite rod antenna for use up to around 10 MHz.

Figure 4. In this variation the coils on the ferrite rod antenna replace the inductors on the receiver board.

as long as this remains between 10 pF and 500 pF the antenna will operate optimally. It is worth noting that the same antenna can be pressed into service for medium- and longwave reception, in each case forming a part of the total inductance in the tuned circuit.

Finally, Figure 6 shows a possibility for shortwave DXing. Jumpers JP2 to JP4 are not fitted and instead we fit a small transformer at the AM input. The turn counts given are intended for use with a 5 mm former with a tuning slug suitable for operation up to 15 MHz. The same coil was used in the 'Preselector for Elektor SDR' article in the December 2009 issue of Elektor and is available from Conrad Electronics (order code 516651) or from ModulBus (RF coil kit T1.4) [1].

The input coil replaces the inductor in the tuned circuit on the receiver board, and

opens up the possibility of changing the turn count on the antenna side to optimise matching to the antenna. For example, a small coupling winding as shown, consisting of just one turn, will give good matching to a 50 Ω antenna.

PC control

PC control opens up a whole new dimension of possibilities to the Elektor DSP radio beyond its use as a portable receiver. The radio can be controlled over a USB interface using a specially-developed program called ElektorDSP1, written in Visual Basic. The program, including source code, is available for free download from the Elektor web pages accompanying this article [2].

The first step on starting the program is to establish which COM port is to be used: the default is COM1. If, for example, COM4 is to be used, edit the 'COM' text box in the user

interface (Figure 7) appropriately and click on 'Open'. When the program is closed the COM port setting is saved in the file COM. ini, and the value will be restored when the program is next started. All frequencies and station names are stored in the file DSPfreq. ini, and these are also loaded when the program starts.

The program allows the user to choose a frequency from 20 presets for the FM band and a further 20 presets for the AM bands. The frequencies can be edited individually. For each preset a station name can be given for reference: in the case of an FM preset this string will be shown on the PC's screen, but the display on the receiver will show the station name as transmitted by RDS. In the case of AM presets the station names are sent to the receiver along with the frequency data, and the receiver will display this string when the preset is selected
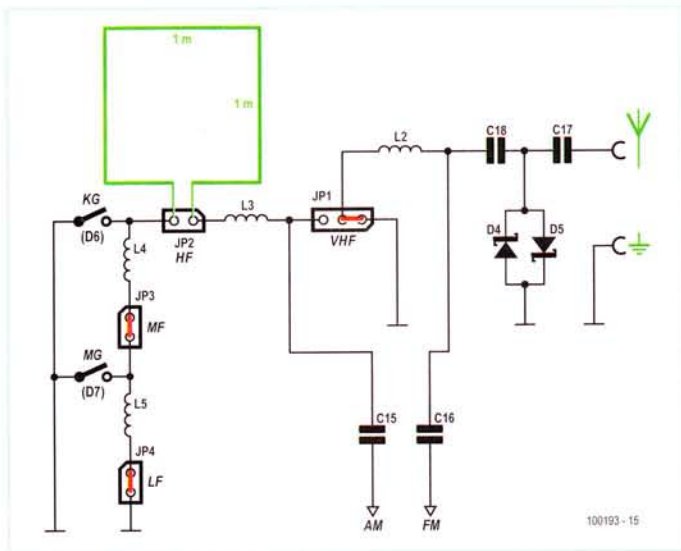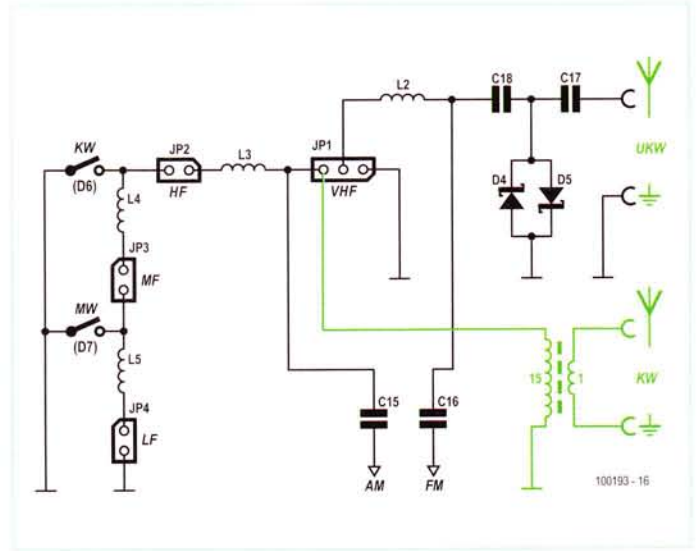
Figure 5. Connecting a loop antenna for AM.

Figure 6. Antenna coupling using an input coil for DX reception.

(by pressing S5 for longer than 0.5 s). This makes tuning to AM presets a little more user-friendly.

The settings can be stored in the receiver as power-up defaults by pressing S5 for more than two seconds. All settings, including any changes to the various parameters described below, will then be copied into EEPROM. To erase all these stored settings, hold down S5 when turning the receiver on: everything will then be reset to standard values.

The Si4735 device in the DSP radio can be configured in a wide variety of ways that determine its reception characteristics. At power-up all these parameters are set to sensible default values, but it is possible to alter their values to suit particular reception requirements or conditions. This possibility was kept in mind when developing the embedded software for the radio, and it is possible to control eight important parameters for AM and FM reception from the PC. If desired, it is possible to set up the receiver with an entirely fresh batch of settings and then save them for when the radio is used as a portable receiver.

FM parameters

In the interests of reducing noise, FM transmitters add 'emphasis' (boost of high frequencies) to the signal. This must be compensated for in the receiver ('de-emphasis'). The Si4735 features a switchable de-emphasis filter.

When signal strength is low many FM receivers produce a poor stereo output with considerable interference; some radios automatically switch to mono mode in these conditions. The DSP radio offers a better solution, smoothly and almost imperceptibly transitioning between stereo and mono modes. The upper and lower signal strength limits for the transition are adjustable.

In very weak signal conditions the DSP radio chip uses a 'soft mute' function rather than the sudden muting provided in some receivers, gently attenuating the output and thus also the noise. The parameters controlling this are the 'mute rate' (speed of volume change), the maximum attenuation ('mute max') and the SNR (signal-to-noise

Table 1. FM properties

Property	Valid range	Effect
De-emphasis	0 to 2	Treble cut
Stereo	0 to 80 dBµV	Stereo mode above this signal strength
Mono	0 to 80 dBµV	Mono mode below this signal strength
Mute rate	0 to 100 dB/s	Soft mute rate of volume change
Mute max	0 to 32 dB	Soft mute maximum attenuation
Mute SNR	0 to 25 dB	Soft mute SNR threshold
Seek SNR	0 to 20 dB	Automatic search SNR threshold
Seek RSSI	0 to 60 dBµV	Automatic search signal strength threshold

Table 2. AM properties

Property	Valid range	Effect
De-emphasis	0 to 1	Treble cut
Filter	1 to 6 kHz	Filter bandwidth
Mute rate	0 to 100 dB/s	Soft mute rate of volume change
Mute slope	0 to 100 dB/s	Soft mute attenuation function slope
Mute max	0 to 32 dB	Soft mute maximum attenuation
Mute SNR	0 to 25 dB	Soft mute SNR threshold
Seek SNR	0 to 20 dB	Automatic search SNR threshold
Seek RSSI	0 to 60 dBµV	Automatic search signal strength threshold

ratio) threshold below which muting is triggered. It is worth experimenting with these parameters, especially when tuning in to weak stations.

The 'seek SNR' and 'seek RSSI' parameters affect the behaviour of the automatic station search function. Stations will only be found if they exceed the specified signal strength (RSSI) and SNR thresholds.

Table 1 gives an overview of the FM parameters.

AM parameters

The parameters governing AM reception (Table 2) are similar to those for FM. An extra is that the bandwidth of the receiver is adjustable to one of a number of predefined values. The parameter can take on the value 0 (6 kHz), 1 (4 kHz), 2 (3 kHz), 3 (2 kHz) or 4 (1 kHz). Here '2 kHz' (the default value) means that the will accept signals out to 2 kHz away from the tuned frequency on either side, and thus corresponds to an actual IF filter bandwidth of 4 kHz. For strong stations, using a wider

bandwidth can improve audio quality, while for DX reception a narrower bandwidth will help reduce interference. The de-emphasis control has a similar effect, and can be used as a crude treble control.

In AM mode there are four parameters that control the soft mute function. The highly effective automatic level control in the receiver IC increases gain when the signal level falls, which as a consequence also increases noise. If the input level falls below a preset threshold, however, the volume will be reduced in proportion. The signal level threshold, the slope of the muting function, the speed with which the function acts and the maximum degree of muting are all adjustable. The default values are designed to work well when

Figure 7. The user interface presented by the ElektorDSP1 program.

Listing 1

Low-level subroutines

```
Declare Sub Get_int_status()
Declare Sub Rx_volume()
Declare Sub Power_down()
Declare Sub Init_am()
Declare Sub Init_fm()
Declare Sub Am_tune_freq()
Declare Sub Fm_tune_freq()
Declare Sub Fm_seek_freq_up()
Declare Sub Fm_seek_freq_down()
Declare Sub Am_seek_freq_up()
Declare Sub Am_seek_freq_down()
Declare Sub Fm_tune_status()
Declare Sub Fm_rsq_status()
Declare Sub Am_tune_status()
Declare Sub Am_tune_status_stop()
Declare Sub Am_rsq_status()
Declare Sub Fm_start()
Declare Sub Am_start()
Declare Sub Am_seek_step_9khz()
Declare Sub Am_seek_step_5khz()
Declare Sub Am_seek_step_1khz()
Declare Sub Init_rds()
Declare Sub Fm_rds_status()
Declare Sub Rds()
```

Listing 2

AM tuning and band switching

```
Sub Am_tune_freq()
  If Fam > 500 Then
    If Fam > 2000 Then
      Portb.0 = 1        'SW
      Portc.3 = 0
    Else
      Portb.0 = 0        'MW
      Portc.3 = 1
    End If
  Else
      Portb.0 = 0        'LW
      Portc.3 = 0
  End If
  I2cstart
  I2cwbyte 34
  I2cwbyte &H40
  I2cwbyte &H00
  H = High(fam)
  L = Low(fam)
  I2cwbyte H
  I2cwbyte L
  I2cwbyte &H00
  I2cstop
End Sub
```

Listing 3

Decoding serial commands

```
$baud = 38400
'*********** RS232 control *******************
    D = Inkey(#1)
    If D = 102 Then F_control           'f, Freq
    If D = 109 Then Mam_control         'm, Memory AM
    If D = 110 Then Mfm_control         'n, Memory FM
    If D = 112 Then Properties          'p, Property
    If D = 105 Then Pc_control_i2c      'i, I2C command
    If D = 106 Then Rdsout = 1          'j: RDS output
    If D = 107 Then Rdsout = 0          'k: RDS output off
    If D = 114 Then Print Rssi          'r: RSSI
    If D = 115 Then Print Snr           's: SNR
```

receiving strong stations. The volume is attenuated by a maximum of 16 dB when the input signal level falls below 10 dBµV. DXers will likely not find this behaviour ideal as weak stations will fluctuate unnecessarily in volume. So, decide whether the primary purpose of the receiver is to listen to stronger stations or DX reception, and set the parameters accordingly. Many DXers will want to disable the soft mute function entirely: the simplest way to do this is to set 'mute max' to zero.

Listing 5

Adjustable properties

```
Sub Properties
  Print "Property"
  Input D
  If D = 1 Then Prop = &H1100        'FM_DEEMPHASIS
  If D = 2 Then Prop = &H1105        'FM_BLEND_STEREO_THRESHOLD
  If D = 3 Then Prop = &H1106        'FM_BLEND_MONO_THRESHOLD
  If D = 4 Then Prop = &H1300        'FM_SOFT_MUTE_RATE
  If D = 5 Then Prop = &H1302
'FM_SOFT_MUTE_MAX_ATTENUATION
  If D = 6 Then Prop = &H1303
'FM_SOFT_MUTE_SNR_THRESHOLD
  If D = 7 Then Prop = &H1403
'FM_SEEK_TUNE_SNR_THRESHOLD
  If D = 8 Then Prop = &H1404
'FM_SEEK_TUNE_RSSI_THRESHOLD
  If D = 9 Then Prop = &H3100        'AM_DEEMPHASIS
  If D = 10 Then Prop = &H3102       'AM_CHANNEL_FILTER
  If D = 11 Then Prop = &H3300       'AM_SOFT_MUTE_RATE
  If D = 12 Then Prop = &H3301       'AM_SOFT_MUTE_SLOPE
  If D = 13 Then Prop = &H3302
'AM_SOFT_MUTE_MAX_ATTENUATION
  If D = 14 Then Prop = &H3303
'AM_SOFT_MUTE_SNR_THRESHOLD
  If D = 15 Then Prop = &H3403       'AM_SEEK_SNR_THRESHOLD
  If N = 16 Then Prop = &H3404       'AM_SEEK_RSSI_THRESHOLD
  If N = 0 Then Prop = &H4000        'Volume
  Input Dat
  Property
End Sub
```

Program-it-yourself

The firmware in the DSP radio was developed using Bascom, and the source code is available, along with a hex file, for free download at [3]. The DSP radio printed circuit board has an in-system programming connector for the ATmega168, which allows you to modify the firmware in the radio to your heart's content. If low-level programming is not your cup of tea, the ElektorDSP1 PC control software described above allows you to customise the receiver without writing a line of code. Furthermore, it is possible to use the USB interface to communicate directly with the firmware and alter the receiver's behaviour. In most cases a simple terminal program is all you need, although it is of course possible to develop programs on the PC to control the radio in specific ways.

The Bascom software runs to over a thousand lines of source code, too much to describe in detail here. However, the program includes a large number of ready-made subroutines with self-explanatory names that control some of the basic functions listed in the Si4735 datasheet (see Listing 1).

The subroutine Am_tune_freq (Listing 2) is worth a closer look. It shows how a command is typically constructed for transmission to the Si4735 over the I2C bus: the device responds to address 34. In this instance the command code is &H40. After that come four bytes of parameters, including the frequency in kilohertz as a more-significant and a less-significant byte. The AM tuning command also selects the required band via the switching diodes connected to ports B.0 and C.3.

Another aspect of the software worth a look from the point of view of developing special software for manipulating the receiver's parameters is the way the serial port is handled. The interface runs at 38.4 kbaud and appears to the PC as a virtual COM port (see Listing 3). Each command is headed by a single character. For example, a lowercase 'f' results in the subroutine F_control being called, which sets a new frequency. Commands 'm' and 'n' allow the preset

memory to be programmed. The 'p' command gives access to the various 'properties' or 'parameters' of the DSP IC and the 'i' command gives direct low-level access to the Si4735 via the subroutine Pc_control_i2c. Read commands are also available. For example, it is possible to read the current signal strength or signal-to-noise ratio, and to gain access to the RDS data.

Listing 4 shows how a frequency is set. A frequency in the AM band is set by a command such as 'f5955<Enter>': you can try this out using a simple terminal program. In this example, the receiver is tuned to 5955 kHz. FM band frequencies are also given in kilohertz, such as 'f102800<Enter>' for 102.8 MHz. The firmware carries out

appropriate initialisations when switching between FM and AM modes. After a frequency command the unit returns to normal manual operation mode, and so PC commands and manual commands can be executed alternately. The source code of the ElektorDSP1 program gives an example of how to control the receiver in this way using Visual Basic.

Listing 5 shows the range of functions available via the 'p' command. A total of sixteen different receiver properties can be adjusted. For example, to increase the receiver bandwidth, use a terminal to send the sequence 'p10<Enter>2<Enter>'. To disable the soft mute function, adjust property number 13 (AM_SOFT_MUTE_MAX_

ATTENUATION) by sending the sequence 'p13<Enter>0<Enter>'. Even the overall volume can be adjusted in this way, using property number 0. For maximum volume send 'p0<Enter>63<Enter>'.

Listing 6 shows how to access the low-level features of the Si4735. Each command is introduced by the lower-case letter 'i'. For I2C data transfer the next character (which specifies the subcommand) should be 'C'. The following bytes are transferred as raw binary values. The first two bytes are counts of the number of bytes to be sent and received respectively, and the following bytes are then sent verbatim in the natural order. The microcontroller takes care of prefixing the bytes on the I2C bus with the

Listing 4

Tuning over the serial port

```
Sub F_control()
  Print "Tune"
  Input Fin
  If Am = 1 Then
    If Fin >= 60000 Then
      Power_down
      Waitms 10
      Am = 0
      Fm_start
      Waitms 100
    End If
  End If
  If Am = 0 Then
    If Fin < 60000 Then
      Power_down
      Waitms 10
      Am = 1
      Am_start
      Waitms 100
    End If
  End If
  If Am = 1 Then
    Fam = Fin
    Am_tune_freq
    Waitms 250
    Am_tune_status
  End If
  If Am = 0 Then
    Fin = Fin / 10
    Ffm = Fin
    Fm_tune_freq
    Waitms 250
    Fm_tune_status
  End If
End Sub
```

Listing 6

Direct I2C bus access

```
'*** PC I2C commands ****
Sub Pc_control_i2c
  Print #1 , "I2C"
  Do
    Get #1 , Command     'I2C write and read to Si4735
    If Command = 67 Then          '"C"
      Get #1 , Bytesout
      Get #1 , Bytesin
      For N = 1 To Bytesout
        Get #1 , Di2c(n)
      Next N
      I2cstart
      I2cwbyte 34
      For N = 1 To Bytesout
        I2cwbyte Di2c(n)
      Next N
      I2cstop
      If Bytesin > 0 Then
        I2cstart
        I2cwbyte 35
        While Bytesin > 1
          Bytesin = Bytesin - 1
          I2crbyte D , Ack
          Put #1 , D
        Wend
        I2crbyte D , Nack
        Put #1 , D
        I2cstop
      End If
    End If
    If Command = 65 Then    'LCD Line 1
      Input #1 , Text1
      Locate 1 , 1
      Lcd Text1
    End If
    If Command = 66 Then    'LCD Line 2
      Input #1 , Text1
      Locate 2 , 1
      Lcd Text1
    End If
  Loop
End Sub
```

start command and the device address. The receiver IC is then addressed again in read mode. The requested number of bytes is then read and passed back to the host PC. The 'A' and 'B' subcommands of the 'i' command are used to send text to be shown on the receiver's LCD panel: the two commands allow the two lines of the display to be written independently.

The receiver does not automatically return to manual mode after the 'i' command is completed; instead, the device remains under PC control.

In summary, the Elektor DSP radio offers a wide range of options, well beyond those offered by a conventional world receiver. In particular, it offers an unprecedented array of customisation possibilities. We hope that many readers will build the receiver and exchange their experiences on the forum on the Elektor website, and we feel sure that you will come up with ideas far in advance of anything we have thought of!

(100193)

### Internet Links

[1] www.ak-modul-bus.de
    (site only available in German: search for
    'HF Spulen Bausatz T1.4')

[2] www.elektor.com/100193
    (web pages accompanying this article)

[3] www.elektor.com/100126
    (first article in this series)

Examined: Hameg HMO2524

By Harry Baggen and Ton Giesberts (Elektor Netherlands Editorial / Elektor Labs)

To an electronics designer an oscilloscope is something like an extension of his brain. As soon as any trifling bit of hardware has been slapped it needs to be checked. An oscilloscope is then an indispensable tool (in addition to the equally indispensable multimeter, of course). You use it to 'just' check whether and oscillator is working properly or whether there is a signal at a certain place in the circuit. Alternatively you can use it to make more accurate measurements of more complex waveforms. Nowadays, all kinds of bus signals from microcontroller circuits
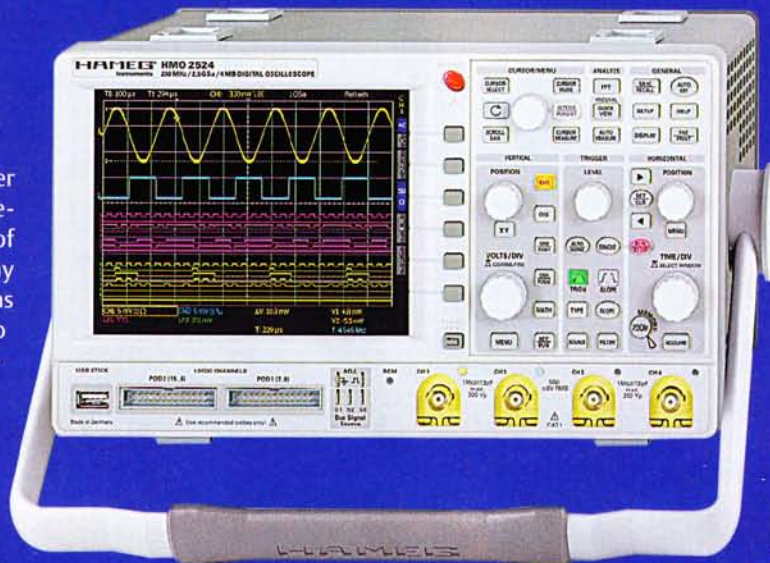
was left in the hands of the design team.

The name Hameg conjures up fond memories for many electronics engineers. This German company was one of the first companies to make affordable and usable oscilloscopes available, long before China and other Asian countries started to compete with their products. The prices were such that even for hobby use it became interesting to purchase such a scope. In the meantime the company has specialised in products with a very good price/quality ratio and a high reliability. That the products made by Hameg have a good reputation is also shown by the fact that the renowned test equipment manufacturer Rohde & Schwarz

a scope which can cope with both analogue and digital signals

are added to that too. You could use a special logic analyser for this, or a scope which can cope with both analogue and digital signals.

Some time ago, Hameg made an offer to Elektor, for us to try out one of their new oscilloscopes for a while. Now, the Elektor lab is always keen to do that. Especially if it concerns one of the more expensive products produced by Hameg. After making an appointment with Mr. Grimm, he arrived a few weeks later at Elektor House with a brand new scope and the necessary accessories. He gave an elaborate demonstration of the capabilities of this scope, after which it

bought the company in 2005 to expand their product range at the bottom end. And we probably won't have to tell Elektor readers about the quality (and also the price) of R&S gear, these are at the top of the range!

In recent years Hameg has expanded their product range at the top end with more expensive and more powerful models. One of the reasons is a better fit with the products that its big brother R&S has to offer. And it is one of these models that our designers were allowed to test drive for a while: the HMO 2524. This is a powerful, digital, 4-channel scope with an analogue bandwidth of 250 MHz and a real-time sampling-

rate of 2.5 GSamples/s (when operating in 2-channel mode). The price of the base version of this scope is around 3300 Euros (excl. VAT). The scope can also be used as a 16-channel logic analyser. Additional software options are available to test buses with protocols such as I2C, SPI and RS-232. Of course, for our practical test we had several 'pods' (adaptors to make the digital connections) available to us and the required software options were also activated in the scope. Various types of probes were obviously present as well and we also had an active probe with an input capacitance of only 0.9 pF (suitable for measurements up to 1 GHz).

The first thing you notice when switching the scope on is the bright and clear display, which has a resolution of 640 x 480 pixels and an LED backlight, it is a pleasure to look at! In the lab we have several digital 'all-round' scopes in the price range from 1000 to 3000 Euros and none of those comes even close to competing with this.

The number of options is overwhelming, you will have to invest quite a bit of time before becoming familiar with all the niceties. However, the user-friendliness does suffer a little as a result, because of all these options and with only occasional use, even the most simple settings can be a chore to find in the menus. During our first measuring attempts, Ton, who has more than 20 years experience in the lab and has worked with many different scopes, did need quite some time to dig through a number of menus to finally find the appropriate settings to give a clear test signal on the screen. The quality of the displayed signal, after some effort, is outstanding however. Moreover, the scope reacts quickly to changes in settings, something that is regrettably not the case with many digital scopes. But a scope with this many options forces you to have a good read of the user manual (something that our designers — just like other users of complex equipment — do only rarely and then reluctantly). So the number of features is extremely large. There are numerous trigger modes, there is a sizeable buffer of 4 MSamples

where you can search for signals in various ways, there is a very usable FFT-function and there are many cursor measurements as well as mathematical functions that you can program yourself.

What's more, we haven't even mentioned the digital signal measurements yet. This gives you the ability of measuring 16 digital signals simultaneously, at a rate of up to 1.25 GSamples/s, also with very extensive trigger options. This does, however, require special pods to connect the digital signals under test to the connections on the scope. Depending in the type of bus you want to look at you may have to install additional software (available for I2C, SPI, UART/RS232).

With this many signals displayed on the screen at the same time, it can become very cluttered. To keep everything well-organised, Hameg have made a very handy scroll function that allows you to move through a virtual screen which is a lot bigger than the real screen. We like it a lot!

This new series of oscilloscopes from Hameg, which are, considering the price, above their existing range of analogue/digital equipment, can easily compete on capabilities and features with comparable products from other scope manufacturers such as Agilent, LeCroy and Tektronix. The HMO2524 is an excellent and reliable test instrument for both analogue and digital measurements. But if you are considering purchasing such an instrument you should have the applications to put its many abilities to good use and you will have to invest the time to become proficient with all its features (but this is equally true for any scope in this price range).

Unfortunately the scope had to go back to Hameg. We will especially miss that crisp and bright screen. Is it really not possible to swap it for a few of those other oscilloscopes that we have sitting around here?

(100420-I)

Internet Link: www.hameg.com

Know what you're measuring

By Thijs Beckers (Elektor Netherlands Editorial)

At our editorial offices we receive quite a few submissions describing ideas and designs for electronic circuits from readers and authors. All submissions are looked at and receive a cursory assessment as to their usefulness and correct operation, as part of an initial sorting procedure, before putting some serious effort into it. This also applied to a submission for sorting the best (audio) capacitor from a collection of capacitors. At first glance the described method showed much promise: without special (expensive) test equipment it would be possible to compare the quality of capacitors with each other and select the better one(s). This looked like a very interesting design to us. The measuring method is described below.
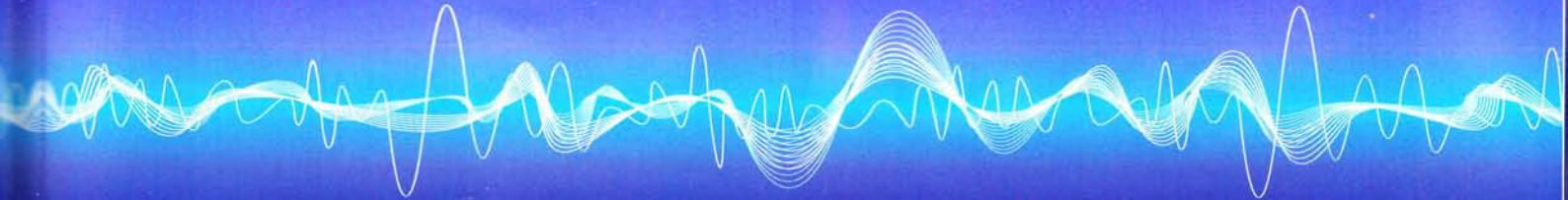
The figures show the test arrangement. The top circuit is used for testing bipolar capacitors. The bottom circuit provides polarised electrolytic capacitors with a bias voltage for testing those types. In each case the bottom capacitor (C2 and C4 respectively) is the 'reference capacitor' and the top capacitor (C1 and C3 respectively) is the 'CUT' (Capacitor Under Test). The idea behind measuring with a bias voltage is that a polarised electrolytic capacitor will distort (even) more when the voltage across the capacitor is allowed to become negative, which is potentially the case with an audio signal. By giving the electrolytic capacitor a sufficiently high bias voltage, a negative voltage across its terminals can be prevented.

The oscillator generates a square wave, which is then divided across both capacitors. The voltage across both capacitors is then visualised using an oscilloscope. By operating the oscilloscope in X-Y mode, specific figures are created which depend on the differences between the capacitors. A 'perfect' diagonal line means that both capacitors are exactly identical, meaning the CUT is of the same quality as the 'reference capacitor'. Any deviation from the diagonal, such as hysteresis, asymmetry or non-linearity means that the CUT is different from the reference capacitor. These differences could conceivably be a larger series resistance, dielectric absorption and other non-linearities. Within this context, it's obvious that the reference capacitor
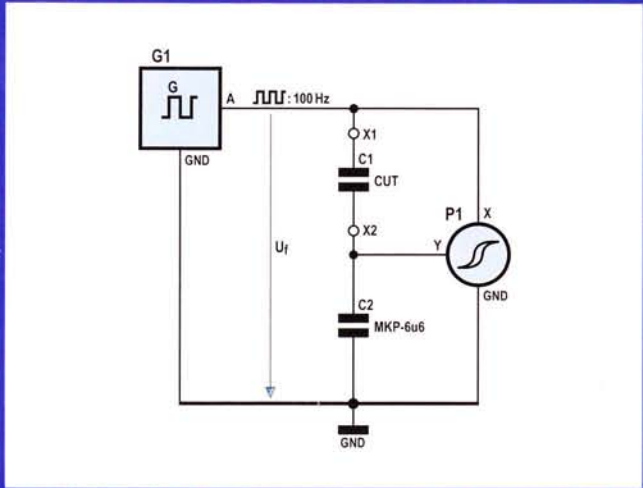
And because we were also unable to find a way, without using expensive equipment, to establish whether the difference was caused be a difference in capacitance, we regrettably had to reject this article proposal. The problem with this circuit is that you do not really know what you are measuring exactly. That makes the proposed method unsuitable for detecting quality differences between various types of capacitor.

it is not possible to establish the quality of a capacitor with 'cheap' equipment

needs to be a specimen of high quality.
During our measurements in our test set-up the idea looked promising. It was indeed possible to clearly see the differences between different types of capacitors. There was an obvious non-linear distortion of the diagonal when a tantalum capacitor was compared to an MKT type, and the difference between an MKT and MKP device was clearly smaller than tantalum MKT, but still clearly visible.

What we also experienced during these measurements, is that a difference in capacitance also resulted in an alleged (quality) deviation. When comparing a 4.7 uF Wima MKP-capacitor (the reference) with a 4.7 uF MKT capacitor the difference became smaller (the 'eye' of the diagonal became narrower) when a 100 nF capacitor was connected in parallel with the MKT device... even when an electrolytic capacitor of 1 uF was connected in parallel with the MKT, the difference from the reference capacitor became smaller. And when making the measurement using a square wave at high frequency (several hundred kHz) very strange things started happening on the scope screen; at both ends of the diagonal there were strange peaks which could not be explained by any differences between the capacitors and which are potentially caused by parasitic inductances in the test set-up.
A (small) difference in capacitance results in such a large deviation that we were unable to establish whether we were dealing with a bad capacitor or just a difference in capacitance.

Unfortunately we have to conclude that for the time being it is not possible to establish the quality of a capacitor with 'cheap' equipment. But we fortunately still have our ears, which we can use to test the capacitors in the signal path for what is still the most important characteristic in audio circuits: the sound quality.

(100482)
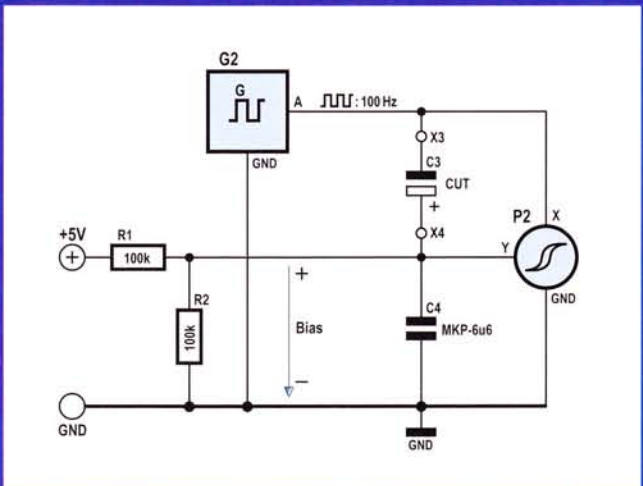
Tiny samples, big footprint

We are slowly becoming more aware of our impact on the environmental and carbon footprint but something happened in the lab the other week that made us smile: for an upcoming project one of our team, Clemens Valens ordered four IC samples from the chip manufacturer Analog Devices.

One week later the editor of our French edition of Elektor took delivery of a surprisingly large (at least 30 litre) parcel. The complete parcel weighed-in at 1.3 kg and on opening he found, surrounded by packing material, two more boxes in which were nestled some 50 cm long plastic tubes containing the miniscule samples. The total volume of the chips themselves can hardly be more than 1 cubic centimetre!

From the labels on the box the package began its journey to France in the Philippines. Needless to say, Clemens was relieved he wasn't paying for the samples or the post and packing on that one!

(100240)

And dropping by at Elektor...

Our position in the centre of Holland near the German border and close to a motorway means that we are often pleased to welcome guests from Germany. Recently Harry Sievers, Franz Ruffer and Patrick Schmidt dropped in. They represent the distributor CS&S electronic (www.csselectronic.com). Their company carries high-tech Korean products including boards from TeleChips using ARM-compatible controllers with multimedia-capable chipsets. They are intended for mass market applications such as portable devices and communications equipment. These powerful boards look interesting but the Codecs are protected by strict licences so we are unlikely to see them making an appearance in an Elektor multimedia player in the near future but stay tuned – we're working on it...

(090928)

Useful spares or toxic waste?

The Feedback form on the Elektor website is the first place to go if you have a question about any of our projects or products (as well as giving us feedback on any article that caught your eye). General technical questions can best be posted on the reader's forum on our website. There are exceptions though. Unable to find an answer in the forums a reader wrote to us recently with a question that's been bothering him: "I am sure I am not alone in having a large box of old components lying around under my bench. Is it possible to mix newer RoHS compliant components with older non-RoHS components? Any tips from the Elektor lab?"

We put the question to Antoine Authier, head of the laboratory here at Elektor:
"It depends entirely on whether the components are fitted to equipment which will then go on to be sold commercially. Here in the Elektor labs our production facilities conform to the current regulations for commercial production. With very few exceptions we use RoHS compliant components and lead-free solder. For purely personal use however there is no reason why you cannot mix RoHS with non-RoHS in the same design although the aim of course is to reduce hazardous substances in the environment".

Antoine also recommends using lead-tin solder; it's easier to make a clean joint. Lead-free solder has a higher melting point so we can assume that RoHS compliant components are designed to tolerate the higher temperature encountered during the soldering phase but sadly this is not always the case and we still occasionally get problems! On the workbench at home the average hobbyist isn't able to control the temperature and recommended soldering time to the same degree. In general the older non-RoHS component types that soldered without trouble are likewise in their RoHS variant also likely to be fairly trouble-free.

(100220)

Juggling Audio Bits
Audio DSP for DIY applications

By Harry Baggen (Elektor Netherlands Editorial)

Audio hobbyists usually confine their hobby to the analogue domain, since the opportunities for doing your own experimenting in the digital domain are very limited. There is very little affordable equipment available that allows a multitude of digital-audio processing operations and getting started with DSPs yourself requires a considerable depth of knowledge of this subject matter. With the modules from miniDSP you can easily realise all kinds of audio processing functions without the need to become intimately familiar with digital signal processing.

In the audio world, DIY is most certainly not dead yet. There are still plenty of people who design/build amplifiers and speaker boxes or modify existing equipment. Despite the fact that the source material (the music) and the playback equipment are usually digital, there are still sufficient opportunities for tinkering to your heart's content with the analogue stages that follow. Nevertheless, you will have noticed that more and more audio processing is taking place in the digital domain, ranging from signal processors to digital power amplifiers. This is quite a complicated world for the average audio hobbyist, where there are now few or no opportunities for doing anything yourself. Still, it is becoming increasingly interesting to familiarise yourself with the possibilities that these digital technologies have to offer. A digital crossover filter for an active loudspeaker system comes to mind, which would allow easy adjustment of the crossover frequency, slope, filter type and delay between the drivers.

It is true that there are a few systems on the market which offer you the possibility of experimenting yourself (for example the amplifier module AS2.100 from Hypex, which contains two digital power stages and a DSP for filtering and equalisation, or the semi-professional DCX2496), but overall the choice is very limited.

The young company miniDSP, established in Hong Kong, comprises several engineers who came up with the idea of marketing flexible and affordable digital audio products for both hobbyists and professional audio equipment manufacturers. The objective here was to keep the programmability of these products as simple as possible. In practice this means that you buy a board with a DSP on it and then you choose the required functionality from the software applications that they offer. This software is configured using a comfortable user-interface on your PC. So you only have to type in a crossover frequency and a slope for the filter and the DSP will do exactly what you have asked it to do. There is no need for extensive knowledge of DSPs and programming, and (almost) none for digital audio technology.

The basis of the entire system is the miniDSP board, a printed circuit board measuring 7.5 by 7.5 cm, which contains a DSP and a number of connectors. To this you can add an I/O-board, which has a number of digital inputs and outputs (miniDIGI) and a board with four

digital power stages (miniAMP). All boards have identical dimensions and can easily be stacked together and interconnected. Using the so-called audio plug-ins you can determine the functionality of the DSP board, that is, the software you put into the DSP. At the moment, plug-ins are available for 2- and 4-way crossover filters with equaliser (in simple and advanced implementations) and a mixer with 31-band graphic equaliser. Work is in progress for other plug-ins.

The plug-ins work in combination with Adobe Air, which you have to install on your computer beforehand. Double-clicking the downloaded plug-in is then sufficient to install it.
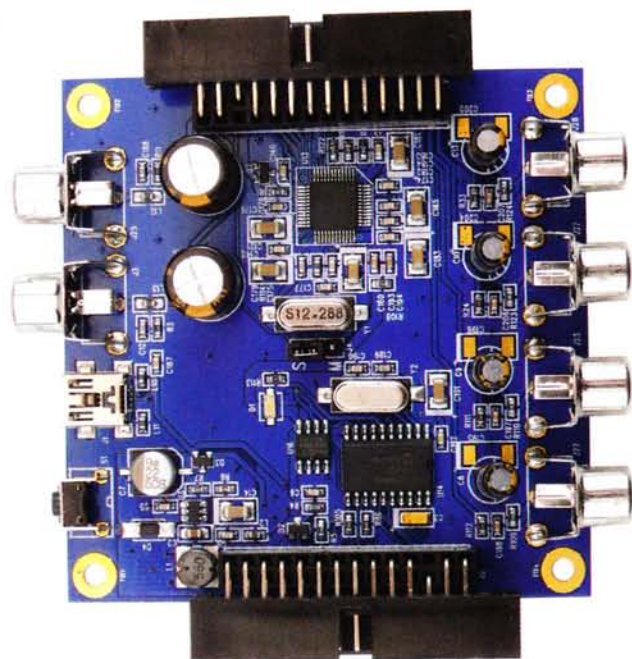
Figure 1. The miniDSP board has two analogue inputs and four outputs.
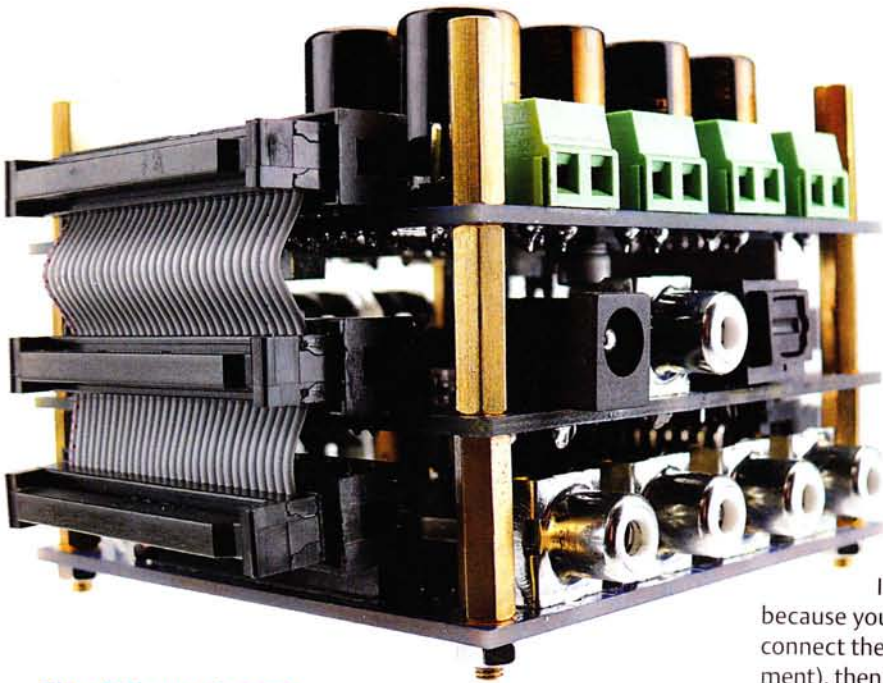
The different boards

First, let's take a look at the hardware. At the core of the miniDSP board (Figure 1) is a ADAU1701 from Analog Devices. This processor was developer specifically for audio applications. Most of the calculations are carried out in 56-bit double precision mode for an accurate result. The IC, in addition to the DSP, also contains 24-bit A/D- and D/A-converters which operate according to the sigma-delta principle, which ensures a large dynamic range. The program for the DSP is automatically read from the serial EEPROM on the board when the power supply it turned on. In addition to the DSP and EEPROM the miniDSP board also contains a PIC18F14K50, which

takes care of the communications between the computer and DSP, using a USB-interface. The board can be easily connected to other audio components using the RCA connectors (2 inputs, 4 outputs). The board uses audiophile quality electrolytic capacitors from Nichicon for a minimal influence on the analogue signals. On each side of the board there is a header for interconnecting all the relevant signals to other miniDSP boards. There is also the option of connecting a potentiometer, which can be used to control the volume in the DSP.

If you require digital inputs or outputs (for example, because you want to go directly from a CD player, or you want to connect the processed signal in digital form to some other equipment), then your needs are met with the miniDIGI board (Figure 2). Here you will find two coaxial and two digital inputs and a coaxial and optical output. This board is available with and without transformers for galvanic isolation of the coax connections. The heart of this board consists of a sample-rate converter (SRC43821 from TI) which will convert all sample rates up to 216 kHz to 48 kHz, which is the operating sample rate of the DSP. Using a jumper you can select which of the 4 inputs is used and another jumper is used to select which signal is routed to the digital outputs (for example the input signal or the DSP processed signal). The board can also be used to de-jitter an S/PDIF-input signal. The last of the currently available boards is the miniAMP (Figure 3). This board contains
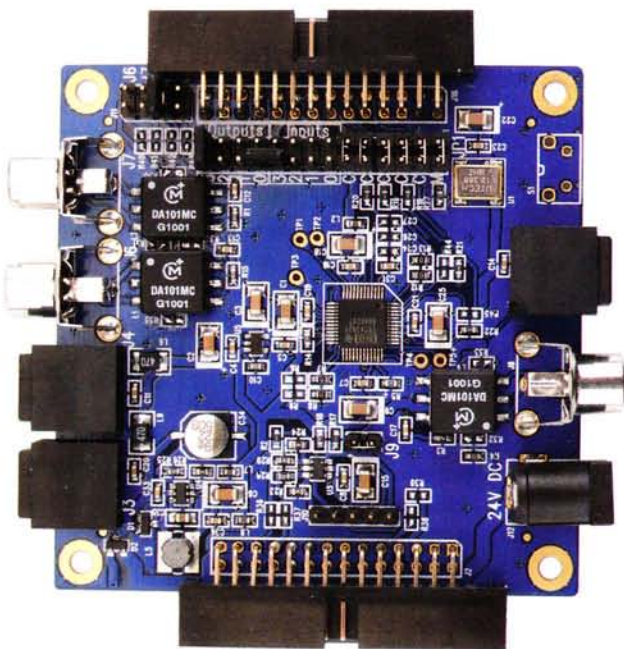
Figure 2. The miniDIGI board contains a sample rate converter and offers a substantial number of inputs and outputs.
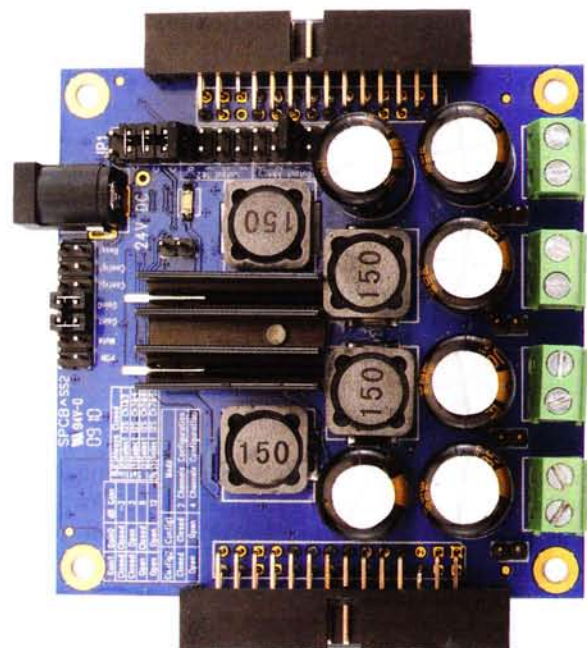
Figure 3. The third board, the miniAMP, contains a class-D amplifier-IC with four output channels, which can also be bridge connected.
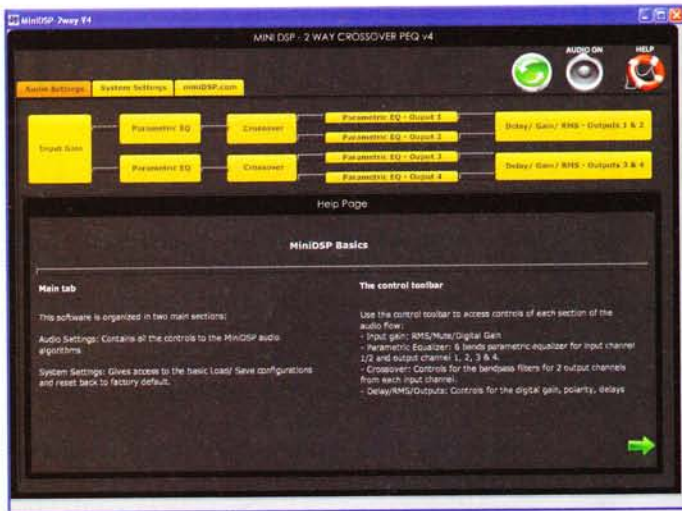
Figure 4. The design of the software. From this overview you can select each function block.
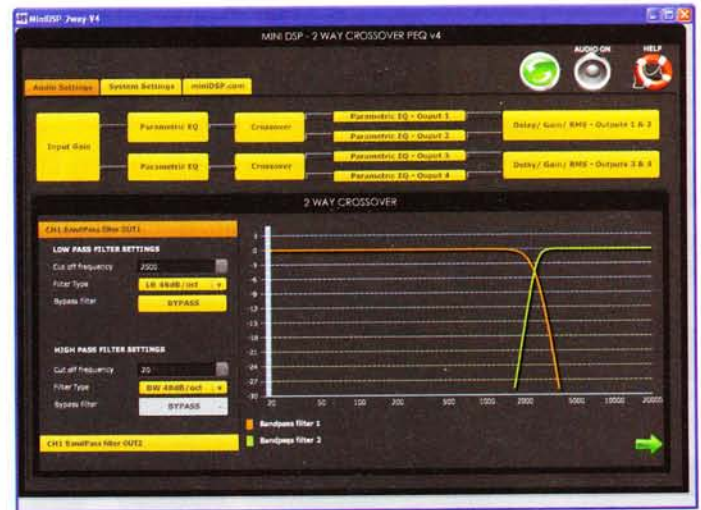
Figure 5. Here we can see the part for the crossover filter. You can select the crossover frequency, type of filter and slope. At the same time you can choose a band-pass filter.

four digital amplifiers, which can be bridge-connected for a stereo version with a higher output power. The power amplifier IC on this board is a TAS5704 (also from TI). This can generate an audio power of 4 x 10 W into 4 Ω or 2 x 20 W into 8 Ω in bridge configuration. Thanks to the class-D design of this amplifier it has a high efficiency (90%) and a small heatsink mounted directly on the PCB suffices for the cooling. The electrolytic capacitors are made by Nichicon, but these capacitors are not necessary in bridge configuration and can be linked out using jumpers. The TAS5704 is presented with the audio signal in digital form, so the D/A converters of the DSP are therefore not used. Although the output power of this board is not all that high, it is ideal for experimenting because you have 4 power amplifiers at your disposal. Just connect the drivers via a few cables to the screw terminals and you can immediately evaluate the effect on the sound of any of the DSP settings that you make.

The printed circuit boards all have identical dimensions and can be assembled one on top of the other with stand-offs. Short ribbon-cable interconnects are supplied to make the connections between the boards. The power supply for the miniDSP board an be provided by the USB connector mounted on the board. The miniDIGI board and miniAMP board both contain a standard DC power supply jack for connecting a mains power supply. The power supply voltage may range from 4.5 to 24 V for the miniDSP and miniDIGI, the miniAMP board requires 12 to 24 V.

Connecting and testing

For this test we received a complete set, comprising a miniDSP, miniDIGI and miniAMP. We selected a miniDSP version with an input range of 2 V, so that you can directly connect the output from a CD player to it (there is also a version with a 0.9 V sensitivity available, to which you could easily add a voltage divider, if need be). The boards are connected together using very short interconnecting cables (see opening photo). There are plenty of possibilities with this complete set of boards, but in practice you would only buy those boards that you need for your particular application. If, for example, you want to add a digital crossover filter to a loudspeaker box with analogue inputs and use existing power amplifiers then

you will only need to buy a miniDSP board. If you would like digital inputs with that then you will also have to buy the miniDIGI board. And should you want to experiment regularly with several different loudspeaker boxes, then the miniAMP board could be a very handy accessory.

The boards are supplied completely assembled. You effectively only have to connect a power supply and you are ready to get started. Well, that is after you have set all the jumpers on all the boards appropriately. In particular the miniDIGI board contains a large number of jumper settings, among others, for the correct interconnections of the I2S signals between the boards. This can be quite confusing initially. Fortunately the miniDSP website contains documentation with various example configurations, which show exactly which jumpers have to be placed where for a certain combination of boards or application. In addition, the developers have advised us that they are continually busy with further improvements and expansion of the documentation. Even during the time we were doing our evaluation there appeared several updates to the software and documentation.

For the software-example for this test we choose a two-way filter with built-in parametric equalisers. This is an excellent test project for such a system.

In the first instance we connected the entire module to our Audio Precision System II to check the operation and perform a few measurements. The module was powered from a large regulated mains power supply (power supply voltage was 24 V).

To get a feeling for the affect that the DSP has on the analogue audio signal, all the filters in the software were set to 'straight through' and we supplied a digital signal via the miniDIGI board. We then measured the distortion of all the analogue outputs of the DSP. These were very low, about 0.005% at 1 kHz at nearly 100% signal level. This value changed very little when we switched to the analogue inputs of the DSP, that is, the A/D converters are now also
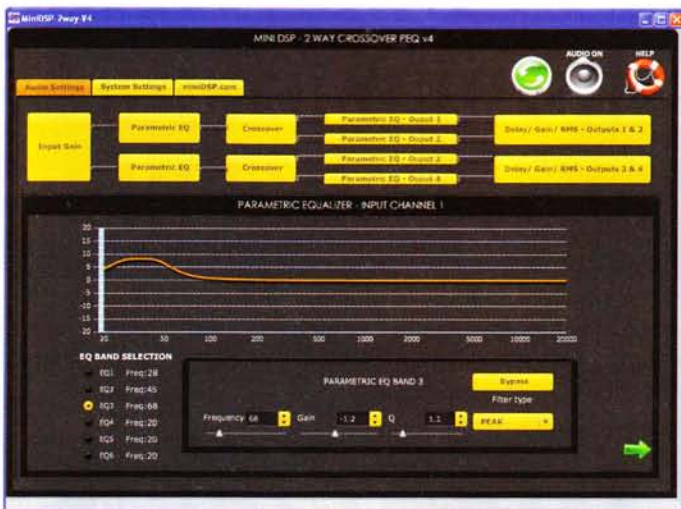
Figure 6. Every parametric equaliser of the two-way plug-in allows a maximum of six corrections (frequency, value, Q factor). Here a simple bass-boost has been realised.

in the signal path. The signal-to-noise ratio is at more than 90 dB (linear), which corresponds to the value that AD specifies for the ADAU1701.

The class-D amplifier on the miniAMP board operates at a frequency of about 400 kHz (changes slightly based on the applied sample frequency). The residual of this frequency at the outputs has an acceptable magnitude of about 100 mVp. The harmonic distortion in single-ended configuration amounts to about 0.07% at 100 Hz and 1 kHz/1 W/8 Ω. The output power at 1% THD and 24 V power supply voltage turns out to be 14.5 W into 4 Ω and 8 W into 8 Ω. This is very close to the specifications from TI. We didn't measure the bridge configuration, but this will certainly deliver 20 W into 8 Ω, as reported in the data sheet for the miniAMP.

In practice

After all this it was time to play with the software and have a look at how we could realise a practical circuit. After installing Adobe Air runtime (free download from the Adobe website) and download-ing the two-way crossover plug-in from the miniDSP website, you only need to double-click the downloaded .air-file after which the installation is automatically carried out and shortcuts are also cre-ated. Now we only need to connect the DSP-board, using a cable with a mini-USB connector, to the PC and we are ready to start experimenting.

When the program is first started the tabbed page 'Audio Settings' appears, which shows a block-diagrammatic overview of the design of the software. The tabbed page 'System Settings' contains a num-ber of general settings, such as the type of input signal (analogue or digital), activating an external volume control, and saving and loading of the configurations (in xml-format). In this way you can save all the settings that you have made and easily retrieve them at some later time. There is also a button here to return everything to the default settings. Finally there is a third tab that brings you to the miniDSP website. All plug-ins follow the same general design. When clicking on one of the function blocks on the 'Audio Set-tings' page, the corresponding settings appear. With the two-way
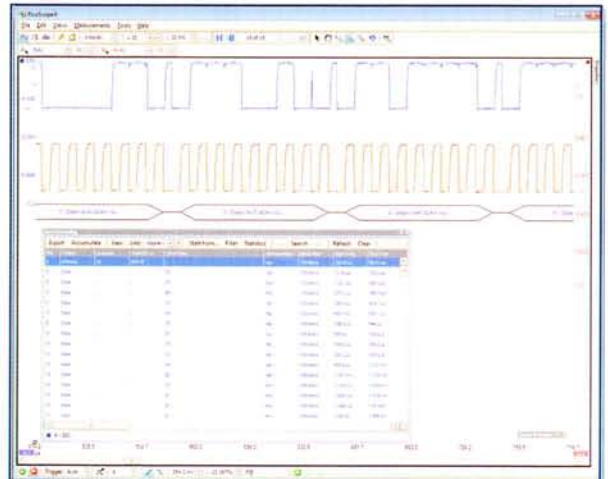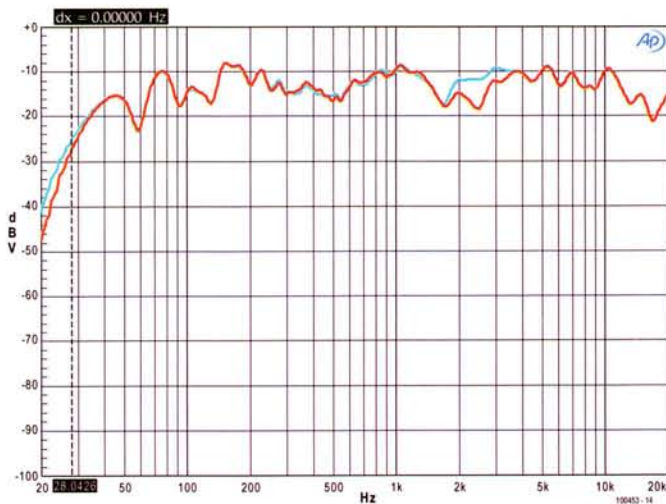
Figure 7. MLS-measurement where the time delay between woofer and tweeter was changed in small steps until the response around the crossover frequency and time delay was measured, Red = no time delay, blue = 0.2 ms time delay, about 7 cm.

crossover plug-in you can select in the first block the attenuation of the input signal. This is followed by a parametric equaliser for each channel with up to 6 configuration settings. Subsequently we arrive at the crossover part where for each driver you can select the crossover frequency, type of filter and the filter slope. The available filter types are Bessel, Linkwitz-Riley (12, 24 and 48 dB/octave) and Butterworth (6 to 48 dB/octave). You can also expand each filter to a band-pass type, for example to protect a small woofer from frequencies that are too low or to combine a two-way system with an already existing active sub-woofer. This section is then followed by the 6-band parametric equaliser, which allows individual corrections to be made for each driver. In the final block you can set the attenuation and time delay for each driver. The latter is particularly important with the Linkwitz-Riley filter type because the woofer and tweeter need to have their acoustic centres perfectly above each other. The correction has quite a wide range, up to 7.5 ms, that is equivalent to more than 2.5 m.

Once all the necessary settings have been made you can click the green 'Synchronize' button and the software will make the connection with the (via the USB-cable connected) miniDSP board. The settings you made are then sent to and stored on the miniDSP board. The firmware for the DSP is also automatically updated, if that is necessary. The green button disappears after synchronising, which appears a little strange at first. Is no further synchronisation possible? There is. After synchronising, all the changes you now make are immediately transmitted to the miniDSP board, in other words, you have a 'live' connection. So you can experiment to your heart's content and change all kinds of things. You can, for example, connect a music signal and hear immediately what the effect of each individual change is on the sound image. So while experimenting with our two-way speaker box we could very easily change the level of the tweeter while listening to it and even play with the time delay between woofer and tweeter. A fantastic opportunity for experimenting with DIY loudspeaker systems! When the program on the PC is closed the last settings are automatically stored in the memory on the miniDSP board.

When you're done experimenting, you can mount all the boards in the loudspeaker enclosure, together with a suitable power supply and power amplifier. As already mentioned, the miniAMP is very handy for experiments, but for a final loudspeaker system it would be better to choose a few bigger (and better quality) power amplifiers.

Very handy

With the miniDSP-system you can in a very simple manner and a very short time assemble an audio circuit which would be much more difficult to realise using analogue components. Consider an active crossover filter with delay time correction and a few frequency corrections. With this system it is done in the blink of an eye. You do not require any knowledge of DSPs, but you will need, of course, knowledge and experience in the area of filter and loudspeaker technology. But this you will need anyway, if you are going to develop your own loudspeaker systems. The method of directly changing the settings from your computer works really well and even encourages you try out many more things than you were perhaps originally planning to do. After all the experimenting you can simply build a final version by building the miniDSP board with power supply and suitable power amplifiers in a loudspeaker box. And, if after a while you decide that these settings are not quite ideal, then it is very quick to connect a PC and make the necessary corrections.

Despite the extensive capabilities of the miniDSP system it is very affordable. For most applications you will only need the miniDSP board and you're done for $99 plus $10 for the plug-in. One board on its own is capable of driving two two-way speaker boxes. However in most cases it will be more convenient to build a miniDSP board into each speaker box, in addition, the board has sufficient computing power to implement a four-way filter.

The audio quality of the miniDSP board is quite good, but we can imagine that some audio enthusiasts are not satisfied with this and would prefer to use other A/D- and D/A-converters than those that are built into the DSP. This is also a possibility with this board, by using the I2S-in- and outputs on the miniDSP board. And there are many more adjustments and options... The most important thing is that with these boards you can very easily make a start with digital audio processing. And all that for a very attractive price!

(100453-I)

Internet Link

www.minidsp.com

miniDSP, rev A (0,9 V) of rev B (2,0 V): $99.00

minDIGI rev A (less S/PDIF-trabsformers): $55.00

minDIGI rev A (met S/PDIF-trafo's): $60.00

miniAMP: $60.00

Audio plug-ins: $10.00 each

Kits of various combinations of boards are available as well.

Dual Voltage/Current Display
For a symmetrical Unilab power supply

By Ton Giesberts (Elektor Labs)

The Unilab power supply module described in the April 2010 edition is a compact, adjustable switch-mode power supply unit that is very well suited to building a single or dual lab power supply. For this application, we have developed a special meter circuit in the Elektor Labs with a four-line display that shows the output voltages and currents of a symmetrical configuration with two Unilab circuit boards.

If you wish to build your own lab power supply, it's only natural to want to equip it with separate meters for voltage and cur-

rent. With a symmetrical power supply, this amounts to two sets of two meters. Although a variety of small, ready-made

display modules with LCD or LED readout are commercially available, they have two drawbacks. The first is that they aren't espe-
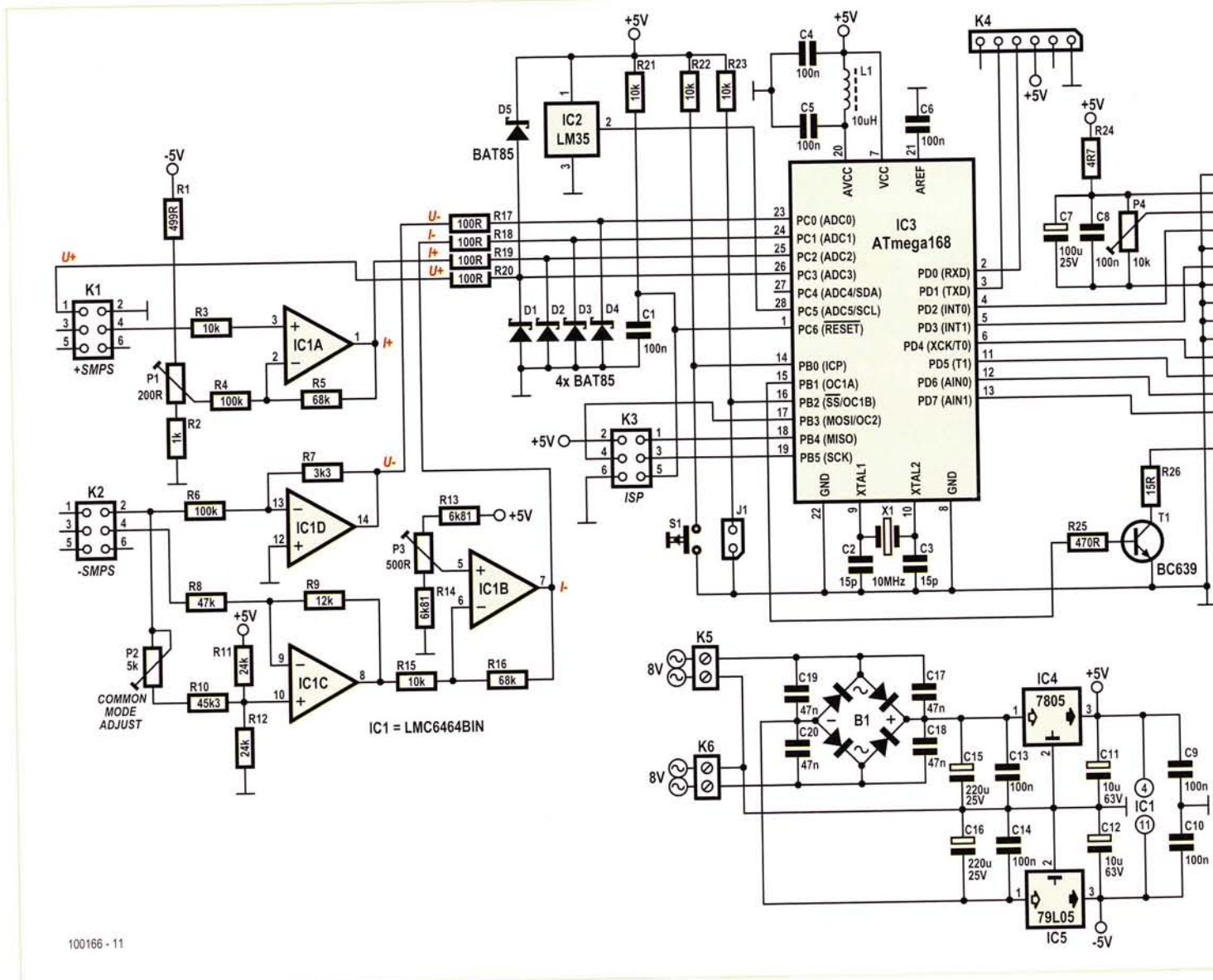
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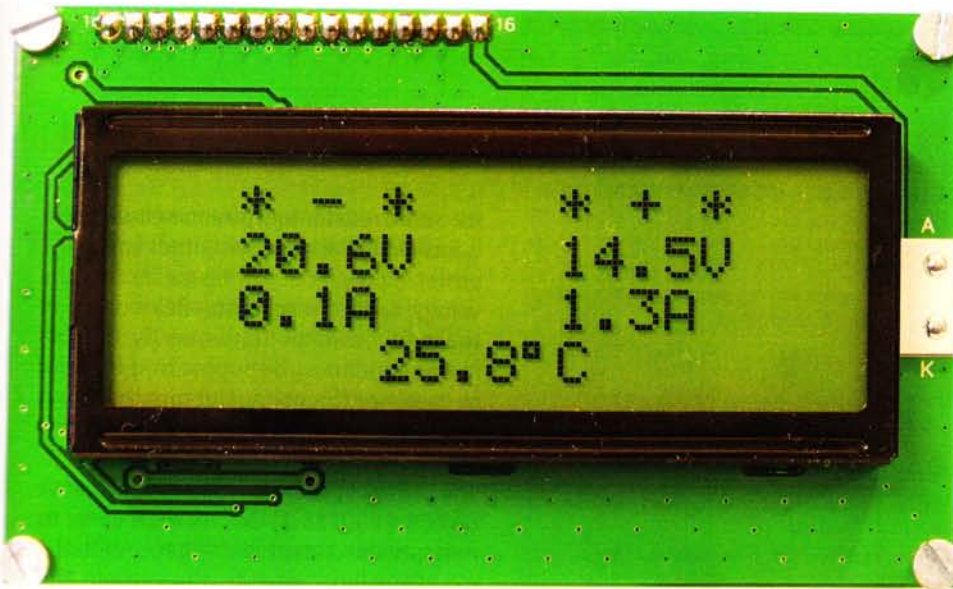
Features

- Readings shown on a large backlit LCD module with 4 lines of 20 characters

- Voltage and current of each supply output (positive and negative) displayed

- Internal temperature of PS enclosure also displayed

- Selectable °C or °F temperature indication

- Signal conditioning circuitry specifically designed for a dual Unilab power supply

- Software-aided calibration procedure

- Powered by a separate transformer (2 × 8 V / 3.3 VA)
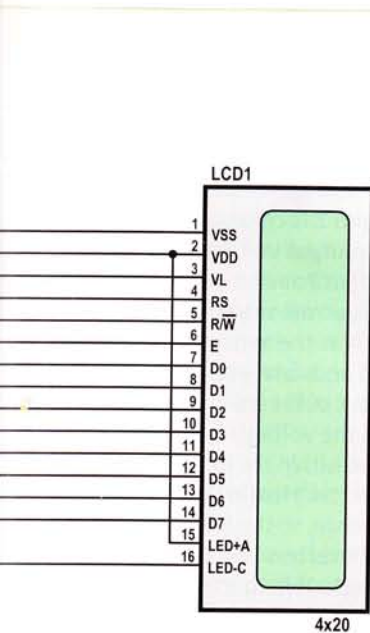
Figure 1. Schematic diagram of the dual voltage/current meter for a Unilab power supply. Naturally, it can also be used for other applications.

cially cheap, which means you're looking at a tidy sum if you need four of them. The second is that connecting modules of this sort to a power supply is always difficult. The amplitude and polarity of all of the measured quantities must be adapted to obtain optimal resolution with the selected modules, and in many cases it's necessary to make certain measurements with the meter floating relative to the actual power supply. As the Unilab module is highly suitable for putting together a symmetrical DIY lab power supply, we decided to develop a separate display unit for this purpose with a large LCD module that shows the voltage and current of both halves of the supply, along with the temperature inside the enclosure. All in all, this solution is certainly no more expensive than four separate modules, and what's more it is much more attractive and guaranteed to work properly. The latter reason is probably the best argument for building your own display module.

Design

Before we explain how the circuit works in detail, it's a good idea to describe the overall design (**Figure 1**). The 'core intelligence' of the circuit is provided by an ATmega168, an Atmel microcontroller with (among other things) six A/D converters. Here we use four of them to measure the two load currents and the two output voltages. The clock signal for the microcontroller is generated using an external 10-MHz crystal. The portion of the circuit around opamps IC1a–IC1d provides signal conditioning for the current and voltage sense signals before they are fed to the A/D converter inputs of

IC3 (ADC0–ADC3). This portion is discussed separately below. Resistors R17–R20 and diodes D1–D4 protect the A/D converter inputs of the microcontroller against negative voltages. Only the positive inputs need to be protected against excessive positive voltages, since the other measurement signals never exceed +5 V.

The LCD module that displays all the readings has 4 lines of 20 characters and is driven directly by the microcontroller. The contrast can be adjusted using P4. Transistor T1 (which is driven by port PB1) allows the brightness of the backlight to be adjusted or switched off. The software causes the backlight to blink if the power supply is overloaded (output current greater than 3 A). The supply voltage line to the LCD module is strongly decoupled by R14, C7 and C8 to avoid any interference on the A/D inputs.

As a bonus, we included an LM35 (IC2) that can be used to monitor the temperature inside the power supply enclosure. The temperature sensor provides an output voltage of 10 mV/K and is powered from the +5 V rail. If you want to have the temperature displayed in degrees Fahrenheit, simply fit jumper J1. In this case the 'C' on the display is replaced by 'F' and the Celsius temperature signal from the LM35 is converted to Fahrenheit using the formula

$$T [°F] = T [°C] \times 9/5 + 32.$$

K3 is the ISP connector. This is handy if you want to modify the software to suit your own wishes. K4 is a debug connector. An Elektor USB/TTL adapter cable (# 080213-71) can be connected directly to K4.

COMPONENT LIST

Resistors
(0.25 W unless indicated otherwise)
R1 = 499Ω 1% 0.6W
R2 = 1.00kΩ 1% 0.6W
R3,R15,R21,R22,R23 = 10kΩ 5%
R4,R6 = 100kΩ 5%
R5,R16 = 68kΩ 5%
R7 = 3.3kΩ, 5%
R8 = 47kΩ 5%
R9 = 12kΩ 5%
R10 = 45.3kΩ 1% 0.6W
R11,R12 = 24.0kΩ 1% 0.5W
R13,R14 = 6.81kΩ 1% 0.6W
R17,R18,R19,R20 = 100Ω 5%
R24 = 4.7Ω 5%
R25 = 470Ω 5%
R26 = 15Ω 5 %
P1 = 200Ω trimpot, e.g. Vishay Spectrol type
  M64W201KB40
P2 = 5kΩ trimpot, e.g. Vishay Spectrol type
  M64W502KB40
P3 = 500Ω trimpot, e.g. Vishay Spectrol type
  M64W501KB40
P4 = 10kΩ trimpot, e.g. Piher type
  PT10LV10-103A2020

Capacitors
C1,C4,C5,C6,C8,C9,C10,C13,C14 = 100nF ceramic, lead pitch 5mm
C2,C3 = 15pF 2%, ceramic, lead pitch 5mm
C7 = 100µF 25V, radial, lead pitch 2.5mm
C11,C12 = 10µF 63V, radial, lead pitch 2.5mm
C15,C16 = 220µF 25V, radial, lead pitch 3.5 mm
C17,C18,C19,C20 = 47nF, ceramic, lead pitch 5mm

nductor
L1 = 10µH 10%, axial, Rs = 0.29Ω, e.g. Epcos
  type B78108S1103K

Halfgeleiders:
D1,D2,D3,D4,D5 = BAT85
T1 = BC639
IC1 = LMC6464BIN
IC2 = LM35CZ
IC3 = ATmega168-20PU, programmed, contained in kit # 100166-71
IC4 = 7805
IC5 = 79L05
B1 = 80V, 1.5A bridge rectifier (pinout ~ + ~ -), e.g. Semikron SKBB80C1500L5B

Miscellaneous
K1,K2,K3 = 6-pin (2x3) pinheader, lead pitch 0.1 in. (2.54mm)
K4 = 6-pin pinheader, lead pitch 0.1 in. (2.54mm)
K5,K6 = 2-way PCB terminal block, lead pitch 5mm
J1,S1 = 2-pin pinheader, lead pitch 0.1 in. (2.54mm)
LCD1 = 16-pin pinheader, lead pitch 0.1 in. (2.54mm)
LCD1 = 16-pin SIL connector for display, lead pitch 0.1 in. (2.54mm)
X1 = 10MHz quartz crystal, HC-49/S, 50ppm, C_L = 12pF
LCD, 4x20 characters with background lighting, dim. 60 x 98 mm, e.g. HTDISPLAY HC200401C-YF62L-VA
PCB # 100166-1
Software en PCB design files: free download *
Kit of parts incl. PCB, programmed controller and LCD: Elektor order # 100166-71 *
* Details at www.elektor.com/100166
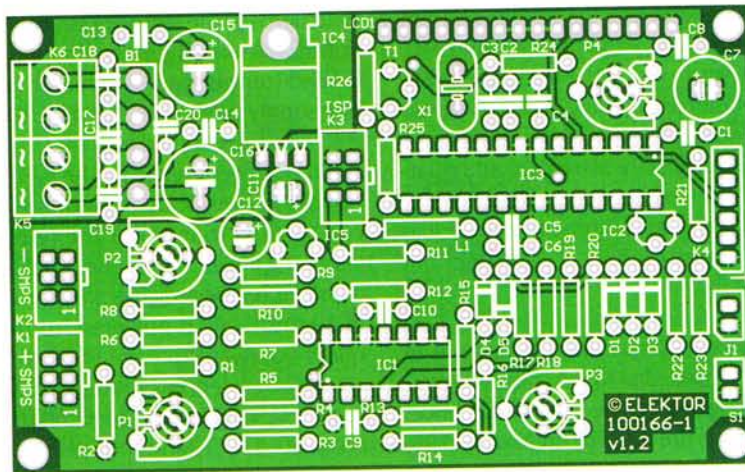
Figure 2. The double-sided PCB is exactly the same size as the display module.

Tweaking and shifting

To build a symmetrical lab power supply, you connect two Unilab power supply modules (# 090786-1 or 090786-71) in series, with the junction of the plus output of the one module and the minus output of the other module forming the neutral terminal of the overall power supply. Unfortunately, this makes it difficult to measure the output current of the negative supply, since the sense resistor for current measurement is located in the negative output line of the power supply module. This means that the voltage on the sense resistor floats on top of the output voltage. Consequently, the output voltage must be suppressed in order to measure the voltage across the sense resistor.

The Unilab circuit board has a connector (K1 on PCB # 090786-1) that makes it easier to measure the current and voltage. With the positive supply module, the voltage and current can be measured using pins 1 and 4 of this connector. Pin 2 is connected to the minus output terminal on the PCB and thus serves as the ground connection point (the neutral terminal of the symmetrical power supply) for the display PCB. With the negative supply module, pins 2 and 4 are used to measure the voltage and current, respectively. Here pin 2 is connected directly to the negative output voltage, and the voltage between pin 2 and pin 4 is proportional to the voltage across the sense resistor. In order to allow the various quantities to be measured accurately using the internal A/D converters of the microcontroller, the full range of the voltage or current sense signal must lie within the full-scale range of the A/D converter. Here we use the internal voltage reference, so the full-scale range of the 10-bit converters is 1.1 V. This means that it is not possible to measure negative voltages.

Measurement setup for the positive supply

As the positive output voltage can range from 0 V to approximately 30 V, all we need in order to measure this voltage is a voltage divider. It is located on the power supply board (R15 and P7), but it must first be adapted to our requirements. By reducing the value of R15 on the positive supply module board to 5.6 kΩ (preferably using a 0.6-watt metal film resistor), we can use P7 to adjust the voltage on pin 26 of the microcontroller to less than 1.1 V at the maximum no-load output voltage. The exact value is not especially important, since everything is sorted out by the calibration procedure.

Measuring the current is a bit more compli-

Test Circuit

We designed a simple current source (see the schematic diagram) to help you align the signal conditioning circuitry around IC1 and perform the subsequent calibration. To allow this test circuit to operate at the lowest possible voltage, it must have its own source of 5 V power. We assume that everyone will have a suitable supply available.

This is a conventional current source design using an opamp that compares the voltage across the emitter resistor of a transistor with a set value, which in this case is taken from the wiper of P1. Here the transistor consists of two devices connected in parallel (T1 and T2), which can jointly handle a substantial current. This approach also distributes the heat load more evenly over the heat sink. If you use three standard 0.25-watt resistors with a value of $0.15\ \Omega$ for the emitter resistor, this circuit can easily supply currents as high as 4.A. The usual practice when transistors are connected in parallel is to provide each transistor with its own emitter resistor. This is not possible here, and a specific voltage drop across each emitter resistor is also necessary for proper operation. To nevertheless ensure a reasonable degree of current sharing, we use individual base resistors here. However, this only works well if the gains of the transistors are roughly the same, so you must pay attention to this. C3 and R2 stabilise the feedback loop.
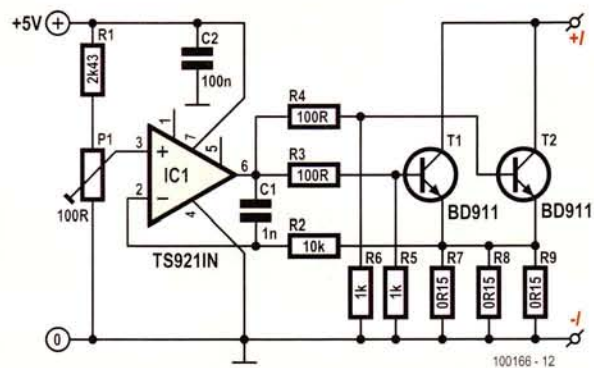
The maximum amount of power that must be dissipated is 120 watts (for example, $3.5\ \text{A} \times 34\ \text{V}$). This is rather a lot and calls for a substantial heat sink. We solved this problem by using a fairly small heat sink together with a hefty fan to provide strong forced-air cooling.

The transistors are ST types in a TO-220 package. We chose the TS921IN rail-to-rail device for the opamp. It can supply enough current (80 mA) to drive the two transistors.

In theory, it should be possible to set the current to just under 4 A with voltage divider P1/R1, but the tolerance of the potentiometer may cause the actual value to vary considerably. If the maximum current is insufficient, you can increase the supply voltage a bit (but do not exceed 12 V) or adjust the value of R1.

cated. On the power supply board, the voltage across the sense resistor is first fed to amplifier (IC3a) with a gain of approximately 4. The output signal from this stage is available directly on pin 4 of connector K3 of the power supply module. Due to the design of the Unilab module, the output voltage of this amplifier stage is negative, but it has the same slope as the voltage across the sense resistor. If the voltage across the sense resistor increases, the output voltage of the amplifier becomes less negative. This means that we need a circuit on the display board to shift the voltage into the input range of the A/D converter and amplify it a bit more to obtain maximum precision. This task is handled by opamp IC1a, with an additional gain of 1.68. With the gain of IC3a in the power supply module and a maximum signal level of 150 mV across the sense resistor, this yields a voltage of approximately 0.97 V at the A/D converter input. As the 0 A setting can be adjusted reasonably close to 0 V with P1, the circuit also has some headroom to display higher currents (due to supply overloading), up to a maximum of around 3.4 A. The –5 V supply rail is used as the reference voltage for voltage divider R1/R2/P1.

Measurement setup for the negative supply

The only thing we need to measure the negative power supply output voltage is an inverting amplifier (IC1d), which attenuates the output voltage (maximum value 30 V or more) to match the input range of the A/D converter. An output voltage range of 0–33 V can be measured with the specified values of R6 and R7 ($-3.3\ \text{V} \div 100 \times -33 = 1.089\ \text{V}$).

In order to convert the voltage on pin 4 of K2 into a usable current reading, we need a bit more than what we use for other signal processing stages. Although the current sense signal has the same form as the signal for the positive output voltage, here it floats on top of the output voltage. Since we're already sensing the output voltage, we can use a differential amplifier (IC1c) to suppress this voltage as a common-mode signal. Here we implement a classic differential amplifier using only one opamp. As

we need to be able to handle voltages up to more than 30 V and the IC supply voltages are only ±5 V, this stage also attenuates the signal. This avoids applying voltages to the IC that exceed its input voltage range.

Component dimensioning around IC1c requires careful attention. For example, we must bear in mind that the voltage may be

as high as 33 V. At 33 V, the voltage divider formed by R10, R11, R12 and P2 causes the voltage at the inputs of IC1c to be slightly less than –4 V. To allow the signal level to be adjusted by P3 in the following stage (IC1b) so it lies within the allowable range, this stage inverts the output of the differential amplifier. This means that the voltage across the sense resistor must also be
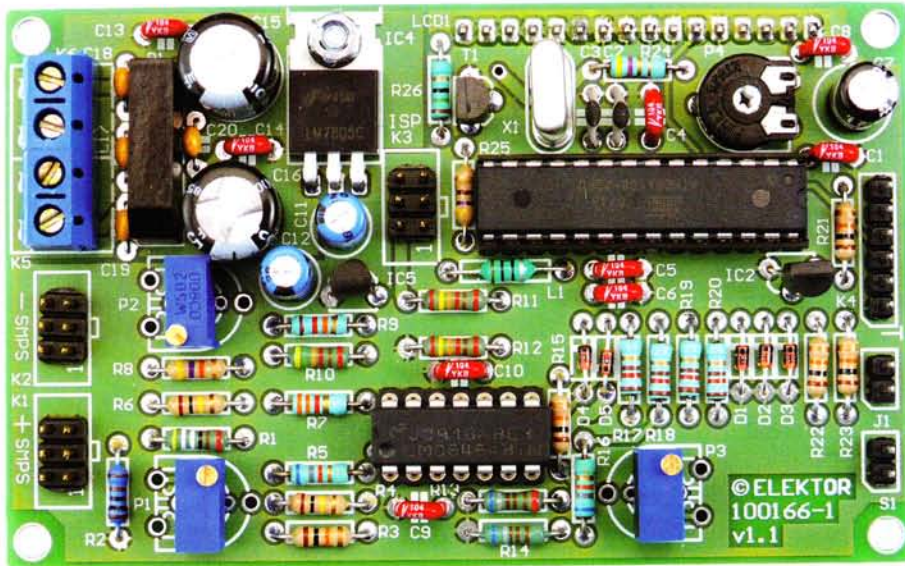
Figure 3. The fully assembled prototype. It is fitted with the trimpots specified in the components list, but there is enough space available to fit other types.

inverted by the differential amplifier, so the signal from pin 4 of K2 is fed to the inverting input of IC1c. The voltage divider ratio (which can be adjusted with P2) must match the ratio of R9 and R8.

For the opamp, we chose a National Semiconductor micropower CMOS device with rail-to-rail input and output capability. The supply current, speed and bandwidth are not important here because we are only processing DC signals. However, the rail-to-rail properties are important, along with the low bias current (typically 150 fA, so extra compensation is not necessary) and the excellent temperature coefficient of the input offsets (only 1.5 µV/°C). The various input offsets (a few millivolts) are compensated by the calibration procedure.

Power supply

It's a good idea to galvanically isolate the power supply of the display board from the Unilab modules in order to eliminate potential ground loops. The implementation here is straightforward. A transformer with symmetrical secondary windings, a bridge rectifier and smoothing capacitors form the basis for the symmetrical 5-V supply for the display board. We use a standard 7805 for the +5 V supply voltage. To improve heat dissipation, this voltage regulator is fitted flat on the PCB without any insulation (the PCB has copper planes on both sides). Note that the +5 V supply must also provide the current for the backlight of the LCD module, which

can draw more than 100 mA (47 mA in our prototype). The analogue supply voltage for the microcontroller is separately decoupled by a choke and a capacitor (L1 and C5). All we need for the –5 V supply is a 79L05, since it only has to power the analogue circuitry of the input stage. The current consumption on this supply rail is quite low.

A standard short-circuit proof transformer rated at 3.3 VA is a reasonable choice for the supply transformer, preferably a type with dual 8-V secondary windings (or alternatively dual 9-V windings). However, we took a different approach for our prototype. The transformer for the two Unilab modules is a toroidal type, and it's easy to add a new winding to it. To determine how many turns are needed, you first have to make a winding with a few turns (such as 5 or 10) and then measure the voltage. After this you can simply calculate the number of turns needed for 8 V, since the transformer has a uniform magnetic field. You can calculate how much wire you need by measuring the circumference of the core and adding some more for the leads to the PCB. For our prototype, we needed approximately 5.5 metres (18.5 feet) of wire for each winding (32 turns). We used bifilar winding (two wires together) to obtain two identical windings. Use relatively thick stranded wire with robust plastic insulation for the windings.

Assembly

Assembling the double-sided PCB shown in **Figure 2** should not present any problems, since only standard components are used (i.e. no SMDs). As already mentioned, the 7805 is mounted flat on the PCB and secured by a screw. With this arrangement, the PCB provides adequate cooling. You can choose from various types of trimpots here, either horizontal or vertical. **Figure 3** shows the prototype board fitted with the types specified in the components list. The 16-way headers are fitted on the back of the PCB.

The display module has a 16-pin connector on the back of its circuit board, and it can be plugged into the header of the assembled PCB. The two boards can be fastened securely together using standoffs (12 mm height).

Suitable software must be loaded into the microcontroller before the board is put into service. You can do this yourself by using the ISP port to download the software (the C source code and hex file are available free of charge on the Elektor website under product # 100166-11), or you can order a pre-programmed microcontroller from the Elektor Shop (# 100166-41).

When fitting the two Unilab modules and the display board in an enclosure, be careful to ensure that the plus terminal of connector K2 on the negative power supply board is connected to the minus terminal of connector K2 on the positive power supply board by a thick wire. This connection forms the ground reference for the display board. Be sure to replace R15 on the positive Unilab board by a 5.6-kΩ resistor. In addition, connector K3 on each Unilab board must be connected to connector K1 or K2 (respectively) on the display board using individual 6-way IDC connectors and 6-way flat cable. Be sure to get the connections right! With regard to the remaining wiring of the Unilab supply modules, we suggest that you consult the article in the April 2010 edition. The only other thing you need to do is to connect the two 8-V supply inputs of the display board to the supplementary power transformer or the additional windings on the main transformer.

(100166-I)

Alignment and calibration

After assembling the boards and fitting them in the enclosure, you can start the alignment procedure for trimpots P1 to P3. Set the positive Unilab module to its maximum output voltage and measure the voltage on one of the leads of R20 of the display board. Adjust P7 to obtain a reading less than 1.1 V (you may be able to simply leave P7 at its maximum setting).

The next task is to align the current signal conditioning stage. Adjust the output of the positive Unilab supply to exactly 0 V and ensure that no load is connected. Adjust P1 to obtain a voltage of exactly 0 V (or slightly higher) at the output of IC1a. Then check whether the voltage on R19 is less than 1.1 V when the output current is 3 A. This does not have to be done at the maximum output voltage; it can be checked just as well at an output level of a few volts. This way you can use a relatively small load resistor, such as 1 Ω / 10 W with an output voltage of 3 V.

No alignment is necessary for the negative output voltage. Unfortunately, the alignment of the signal conditioning circuitry for the negative output current is distinctly more complex.

First the effect of the negative output voltage must be suppressed. This is adjusted with P2. In order to do this properly, we recommend that you build the current source described in the inset (tit also comes in handy during the subsequent software calibration procedure). Now connect an ammeter (with a range of 10 A or so) in series with the current source and connect this arrangement to the negative Unilab output to serve as a load. Set the current to exactly 3 A and vary the output voltage back and forth over the range of 2 V to 25 V. If everything is OK, the current through the current source will remain constant. In this case you can disconnect the meter and then use it as a voltmeter. Measure the voltage at the output of IC1b or IC1c. Adjust P2 until this voltage remains constant when the output voltage is varied. Bear in mind that the signal is filtered by IC3a in the power supply module, so you must always wait a few seconds (or longer) after each change to allow the voltage to stabilise. Disconnect the load. With the current now zero, adjust P3 so the voltage on R18 is 0 V (or slightly higher). After this you can perform the calibration procedure.

The calibration routine allows for three pos-

sible situations when the supply voltage is switched on:

1. An Elektor-programmed microcontroller by is fitted in the circuit.

2. A user-programmed microcontroller is fitted in the circuit.

3. Calibration has already been performed.

In the first case, provisional parameter values have already been stored in the EEPROM memory by Elektor, but calibration is still necessary. In this case, a message requesting you to press S1 is displayed for 5 seconds after power is switched on. If this time expires without S1 being pressed, the display will show the voltages and currents, but the readings may not be correct. As a reminder of this, '[uncal]' is shown at the start of the bottom line. This reminder disappears after calibration has been performed.

If you programmed the microcontroller yourself using the ISP port, data is also stored in the EEPROM (this is part of the available download) and the same course of events occurs as in the first situation. However, if you programmed the microcontroller in some other manner, there is probably no data in the EEPROM. In this case the display indicates that calibration is necessary after the unit is switched on, since the display cannot present meaningful readings without calibration data.

If calibration has been performed, the display shows the correct readings after first presenting a welcome message.

Calibration

Button S1 is used for the calibration procedure. It is not present on the PCB; you must connect an external pushbutton for this purpose. After S1 is pressed once, the voltage, current and temperature screen is replaced by the message screen for the first step of the calibration procedure:

```
CALIBRATION     STEP 1
set outputs to 0V
then press S1
```

Pressing S1 stores the values of the first our parameters, which are used to determine the four zero levels. They are labelled U1L, I1L, U2L and I2L in the software, where 'U1' represents the negative supply module.

Press S1 again to display the message screen for step 2 of the calibration procedure:

```
CALIBRATION     STEP 2
set outputs to 25V
draw 3A from -PSU
then press S1
```

The second step determines the values of the two parameters (U1H and U2H) for the maximum output voltages that the supply modules must be able to deliver (25 V max.) and the parameter (I1H) for the maximum specified output current (3 A). If you have built the current source described in the inset, you can put it to good use here. Otherwise you must provide the right load resistance (such as twelve 100-Ω, 10-W resistors connected in parallel) to draw a current of 3 A from the negative supply output. To ensure that the output voltage is still exactly 25 V, measure it once again with the load connected before pressing S1 again.

Pressing S1 now takes you to the third and final step of the calibration procedure:

```
CALIBRATION     STEP 3
draw 3A from +PSU
then press S1
```

In this step you determine the value of the eighth and final parameter (I2H) by connecting a 3 A load to the positive supply output. The output voltage does not matter as long as the output current is 3 A. If you use the current source, always set the current to 0 A before connecting it to the supply or disconnecting it (i.e. do not make or break the connection at 3 A). Thanks to this third step, only one multimeter and one current source or load resistor are necessary for the calibration procedure. Press S1 again to save this parameter value. The following message screen is displayed to conclude the calibration procedure:

```
CALIBRATION DONE!
press S1
```

After you press S1 the last time, the microcontroller executes a restart and then presents the calibrated readings on the display. You can check them with a multimeter. We recommend that you check the readings after the modules have been fitted in the enclosure and repeat the calibration procedure if necessary.

After calibration has been completed, the display shows the following message for one second after the power supply is switched on:
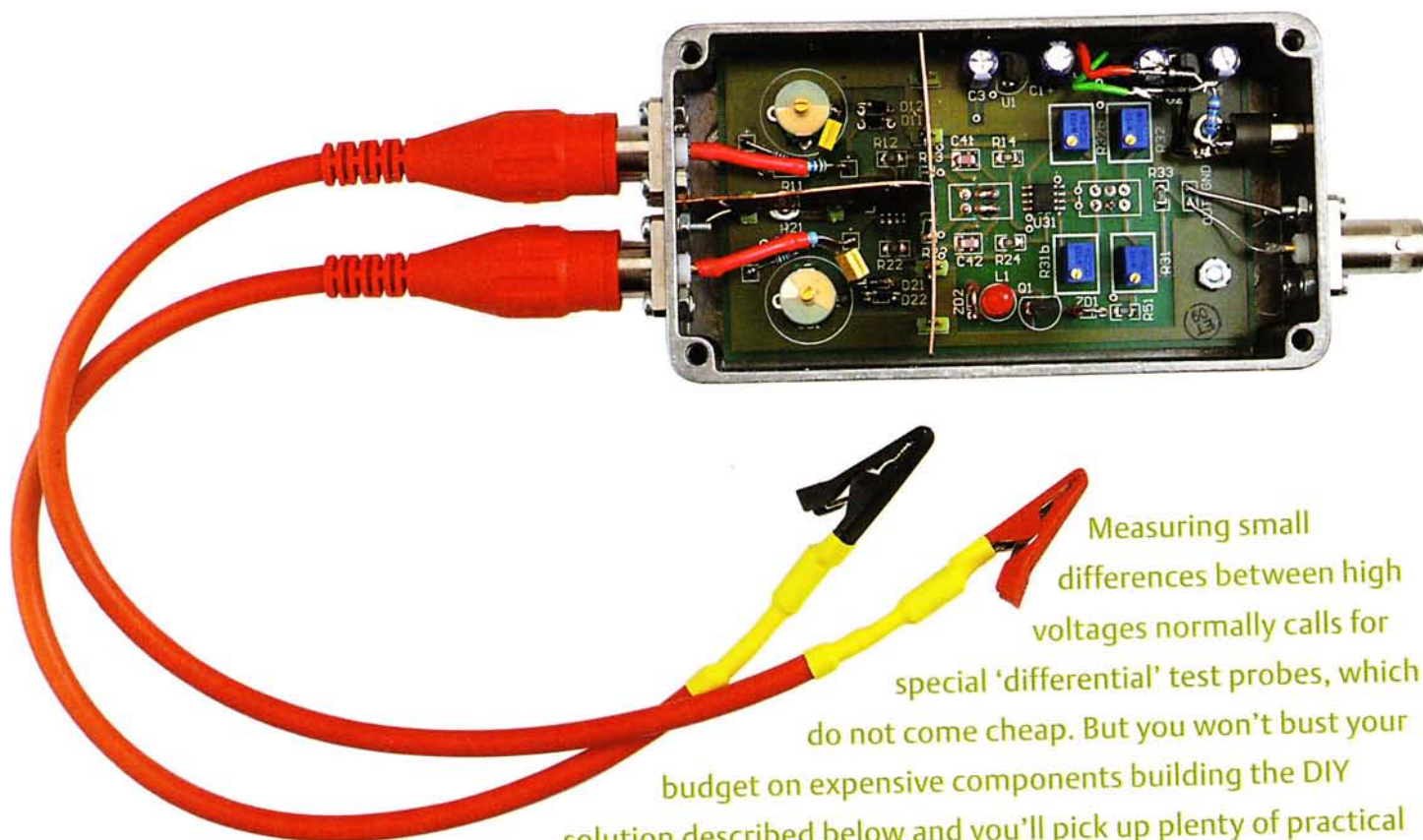
```
SMPS display v1.0.0
```

Note that the actual version number may be different.

High-voltage Probe
Home made and differential, too

By Alfred Hesener (Germany)

Measuring small differences between high voltages normally calls for special 'differential' test probes, which do not come cheap. But you won't bust your budget on expensive components building the DIY solution described below and you'll pick up plenty of practical know-how in the process.

Many circuits employ high voltages. The two best-known examples are switch-mode power supplies and vacuum tube circuitry. A relatively new application is found in hybrid and electric automobiles, which operate with high battery voltages (and dangerously high currents) in order to reduce voltage drop and the cross-sectional diameter of cabling.

Whilst any decent multimeter is adequate for measuring high voltages, it's not so handy for measuring small fluctuations in high DC or AC voltages overlaid on these. Frequently, moreover, we are not interested so much in the absolute value of a high voltage as much as in the difference between two separate high-voltage levels, such as (for example) the differing anode volt-

ages of a push-pull amplifier or at switching nodes in a phase shift full-bridge topology transformer (within high output level switch-mode power supplies).

Starting point

One solution would be to use two standard high-voltage test probes (not too expensive, with serviceable characteristics) together with a digital oscilloscope so as to calculate the signal difference using mathematical functions. This method has three disadvantages:

1. This means using two channels of the oscilloscope, which makes it harder to observe several signals simultaneously.
2. Both signals are digitised at the resolu-

tion of the oscilloscope (generally 8 bits maximum), so that errors add up. Subtracting two large, almost identical voltage values is always problematic, increasing the risk of measurement errors.
3. Since the timing correlation between the two measurement channels is related to factors such as cabling and earth (ground) loops and mathematical functions within the oscilloscopes can throw up random and deterministic fluctuations, the timing information of the signal is not very trustworthy, particularly at higher signal frequencies. Remedy can be found in the so-called 'high-voltage differential test probe'. This is a probe set enhanced with a differential amplifier able to accept very high voltages on its inputs and amplify only the voltage

Characteristics

- Differential attenuation switchable in two stages (–20 dB/–40 dB)
- Bandwidth 1 MHz, switchable limit at 500 kHz
- Maximum input voltage ±1000 V (peak value)
- Maximum output voltage ±10 V (at min. terminating impedance 1 kΩ)
- Common-mode suppression 55 dB at 6 kHz, 35 dB at 600 kHz

difference between the two input connections, whilst at the same time suppressing common-mode signals (signals of exactly the same level on each input).

It might sound as if a simple differential amplifier with voltage dividers on its inputs could do this job but life is seldom this simple. For proof just see how expensive commercial differential test probes are for this kind of measurement task. A good example is the Tektronix P5200. On the Tek website [1] you'll find an extremely readable application note [2] on using high-voltage probes. The maximum input voltage differential is given in the data sheet as ±1300 V and the bandwidth as 25 MHz. Digging deeper into the data, a value of particular interest is the CMRR or Common-Mode Rejection Ratio, which is the ability of a differential amplifier to reject the portion of the signal common to both the + and – inputs. The impressive value of 80 dB at 60 Hz drops off rapidly at higher frequencies, for instance to 50 dB at 100 kHz, which is pretty good nevertheless. A small differential signal becomes progressively more difficult to measure as the frequency of the common-mode signal rises. A frequency sweep of the common-mode signal will indicate that the output signal increases with the frequency of the common-mode signal but this is an illusion. At higher frequencies a greater part of the common-mode signal reaches the output by means of (parasitic) capacitive coupling, making it very difficult to improve the CMRR for high frequencies.

Test probes and oscilloscopes

Although the oscilloscope is a very handy instrument for taking measurements, it can easily lead you astray when you misinterpret what's indicated on the display. A whole load of measurement errors can also occur, so using a differential test probe means keeping your wits about you.

The specification of our low cost, easy-to-build differential test probe is shown in the panel 'Characteristics'. The switchable attenuation feature is a boon when the differential signal being measured is small and the common-mode signal is very large. Here we need to ensure that the test probe remains operating linearly and is not overdriven. An overload indicator of the

kind provided with commercial differential test probes was omitted for reasons of simplicity.

Bandwidth limitation is important for measurements with spurious high-frequency signals, as in switch-mode power supplies. The frequency sweeps are shown in **Figure 1**. The upper of the two curves indicates the output signal at –20 dB attenuation both with (in orange) and without (in green) bandwidth limiting. The blue line is the –40 dB setting with the 500 kHz filter switched in. As expected, the frequency response is very linear, with a maximum cut-off frequency of approx. 1 MHz. The red line shows the common-mode output signal relative to the frequency. By calculation this gives a CMRR of about 55 dB at low frequencies, which falls to about 35 dB at higher frequencies. This fall-off starts at around 6 kHz and is caused mainly by parasitic coupling inside the test probe. It is further influenced at higher frequencies by parasitic coupling within the op-amp.

The maximum input voltage of about

±1000 V was soak-tested in the lab with constant voltage over an extended period. BNC connectors and suitable HV cables are specified for these tests. The input impedance of the circuit must likewise be laid out for high voltages. The resistors used (R1 in **Figure 3**) should be rated for 1600 V (standard resistors are usable up to only 250 V). Alternatively you can wire several resistors of lower voltage rating in series to divide the voltage (assuming all of these resistors have the same value).

One more tip concerning the measurement cables: the cabling on the two differential inputs should be as far identical as possible, since any disparity will unbalance the setup and cause measurement errors. For our prototype we cut a short coaxial test lead with BNC connectors into two halves of equal length, with insulated croc clips fixed to the free ends and the transition insulated with heatshrink tubing (see **Figure 2**). Because these cables represent a capacitive load to the measurement setup, they need to be as short as practically possible.
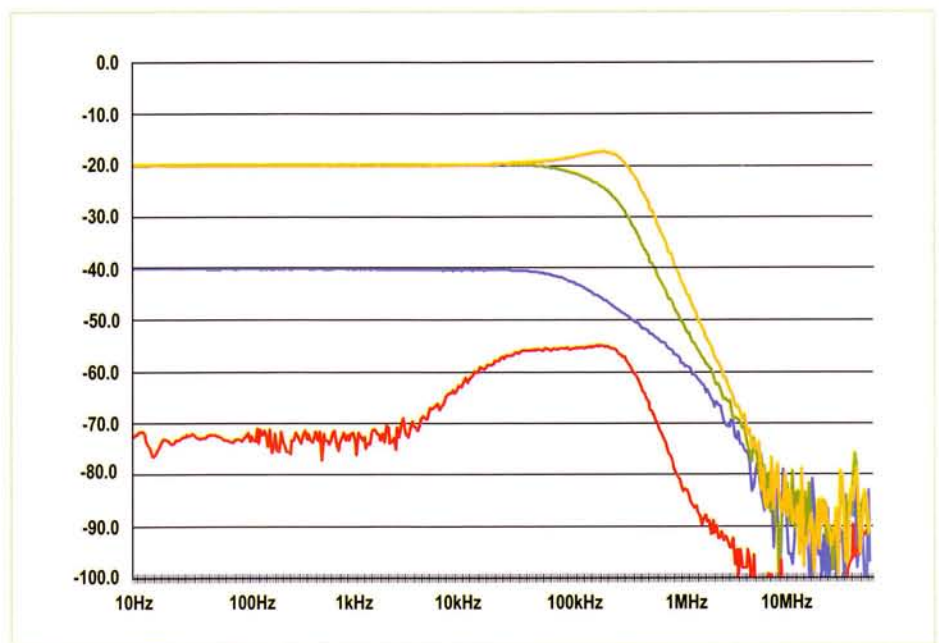

Figure 1. Frequency response at both amplification settings, each with and without the 500 kHz filter. The lower curve indicates the common mode suppression.

Figure 2. Measurement cable with fully insulated croc clips.

Caution: lethal voltages!

Handling high voltages demands setting up properly, proceeding with caution and taking all necessary safety precautions — even if you're in a rush and it seems too much of a hassle. Your life is worth more than rapid results. Before testing a circuit with the type of high-voltage test probe shown here make sure you are familiar with the safety regulations. For devices operating at voltages above 50 VAC or 120 VDC testing live circuitry is allowed only if there are valid reasons why the power cannot be turned off (such as for taking voltage measurements). Under German law these operations may only be carried out by qualified electrical technicians, not by trainees and apprentices.
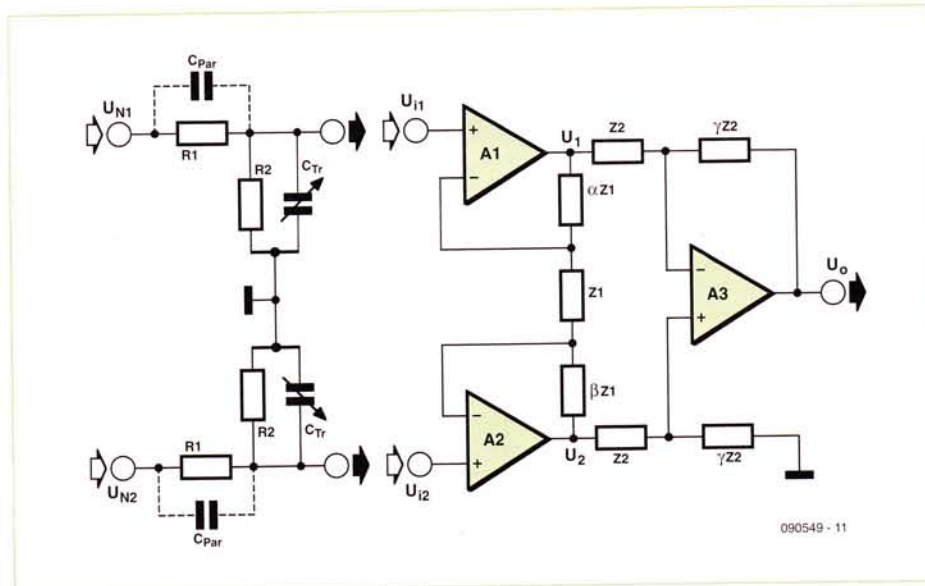

Figure 3. Block diagram with input attenuator (left) and instrumentation amplifier (right).

Input voltage division ratio: $DR_{1,2} = \dfrac{R_2}{R_1 + R_2}$

Output voltage (first opamp): $U_1 = (1 + \alpha + \beta) \times U_{in1} \times DR_1$

Output voltage (second opamp): $U_2 = (1 + \alpha + \beta) \times U_{in2} \times DR_2$

Output voltage (test probe): $U_{out} = (U_1 - U_2) \times \gamma$

Circuitry and input design

The circuit (**Figure 3**) is basically a differential amplifier using three op-amps, sometimes described in the literature as instrumentation amplifiers. A1 and A2 act as high-impedance inputs and amplify differentially, whilst A3 is an actual (classic) differential amplifier. Our circuit offers an interesting advantage over the classic differential amplifier, however: whilst the CMRR in a simple differential amplifier depends on the matching of the resistors Z2, it is here greater by the factor

$\gamma \times (1 + \alpha + \beta)$

This, however, creates amplification and consequently the danger of overload when a large common-mode signal is present. For this reason we must configure the amplification and input voltage divider to produce maximum CMRR in the linear operating range of the op-amp. In the circuit we selected 1 for the value of   and ß, whilst   in the two attenuator stages is 0.657 (at −40 dB) and 6.57 (at −20 dB). Z1 and Z2 are set relatively low at 1 kΩ but with the advantage that the influence of parasitic capacitance in the circuit is reduced.

The formulae for calculating the values are set out in the panel 'Formulae for Figure 3'. In your calculations ensure that the op-amp output voltages remain in the range ±12 V.

R1 and R2 form the input voltage divider. R1 must have high resistance for a high input voltage range and low loading on the signal under measurement. Because the parasitic capacitance C_{par} reduces the division ratio at high frequencies, a trimmer capacitor is fitted in parallel to R2 for compensation. We selected 10 MΩ for R1 and 51 kΩ for R2, allowing a 5-30 pF trimmer to provide compensation. The division ratio is around 198. Alignment of the trimmers is the same procedure as for other test probes, applying a square-wave signal and obtaining the optimum curve shape. Many oscilloscopes have a square-wave generator built-in for this purpose.

Construction

Figure 4 shows the construction of the input stage. The two input resistors (for minimising leakage currents) should be

insulated with heatshrink tubing. Additional ceramic capacitors are wired in parallel with the two trim-caps, to extend the compensation range.

The complete circuit (apart from the two 10 MΩ input resistors) is shown in **Figure 5**. On the left-hand side the two extra ceramic capacitors can be seen in parallel with the trimmers, also the second divider resistor. The diodes at the inputs (D11/12 and D21/22) are for over-voltage protection; they operate much faster than Zener diodes and their capacity is far lower. The high input resistors (10 MΩ) reliably clip any over-voltages from the diodes plus any current leakage from the power supply. For good frequency response it is vital to minimise parasitic capacitance at nodes in the layout.

The differential amplifier formed by U31A follows after the amplifiers U11A and U11B with their negative feedback resistors. To achieve the best match between amplification and CMRR, the negative feedback
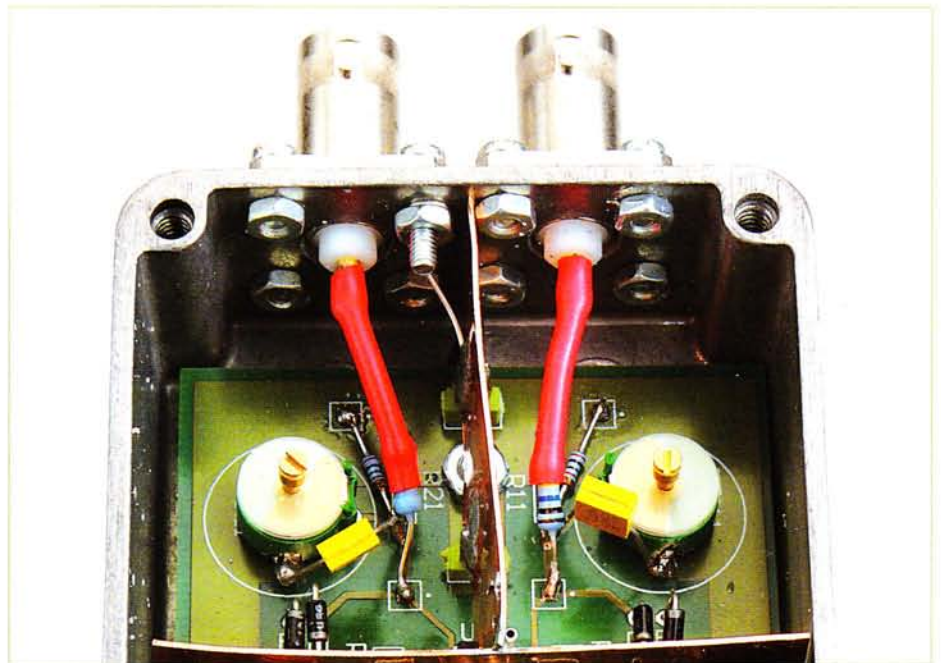

Figure 4. Constructing the input stage.

resistors for U31A take the form of precision trimpots. The output is terminated with a 50 Ω resistor.

Switch S1 enables the amplification factor and the attenuation of the test probe to be toggled. S2 switches the bandwidth limiting

using a 330 pF capacitor in parallel with the negative feedback resistors in the first stage. The only special feature of the power supply using voltage regulators U1 and U2 is the little module using Q1 and LED L1 for monitoring the two voltages. Close to the

Calibration

Calibration is carried out as follows:

1. Connect power supply, observe red LED is lit. Check operating voltages (±15 V) and op-amp output voltages (a few mV).

2. Connect oscilloscope to the output (output terminated in 50 Ω).

3. Set R31 and R31b in the middle of their range and connect a 1-kHz square-wave signal with an amplitude of around 10 V_{pp} to one of the two inputs – the exact value is not so important.

The square-wave signal should be visible on the oscilloscope. Adjust the trimmer capacitor on the relevant input for best square-wave curve shape. The best method is to apply the square-wave signal to trigger the second channel of the oscilloscope and use this for comparison. If the adjustment range

is too small you can connect some more small ceramic capacitors in parallel with the trimmer capacitor. If the trimmer capacitor is too large and the input voltage divider behaves like a low-pass filter, you can give the voltage divider a lower impedance (e.g. using 3.3 MΩ for R1 and 16 kΩ for R2). If you find it impossible to achieve a proper square-wave signal you will need to change the physical construction (the parasitic capacity is too large).

4. Adjust the second input in the same way (square-wave signal on this input only).

5. Now adjust R31 and R31b so that the output voltage corresponds exactly to the input voltage attenuated by the factor you have set. For an input signal of 10 V_{pp} the output signal should be exactly 1.0 V_{pp} with S1 set for –20 dB and 100 mV_{pp} in the –40 dB setting.

6. Now apply the same signal to both inputs and adjust R32 and R32b to make the output signal as small as possible (the smaller the better).

7. Repeat steps 5 and 6 several times, as they are mutually interactive. Take care to achieve maximum common-mode suppression with R32/R32b — this is more important than producing the precise attenuation preset by R31 and R31b.

It is important to complete steps 3 and 4 accurately before proceeding with the remaining alignment. Without good frequency response at the input voltage divider stage all the following steps are meaningless. Signal measurement first on one and then on both inputs should be carried out repeatedly at various frequencies from DC to HF. This will also give you a good understanding of how the test probe behaves with differing signals.
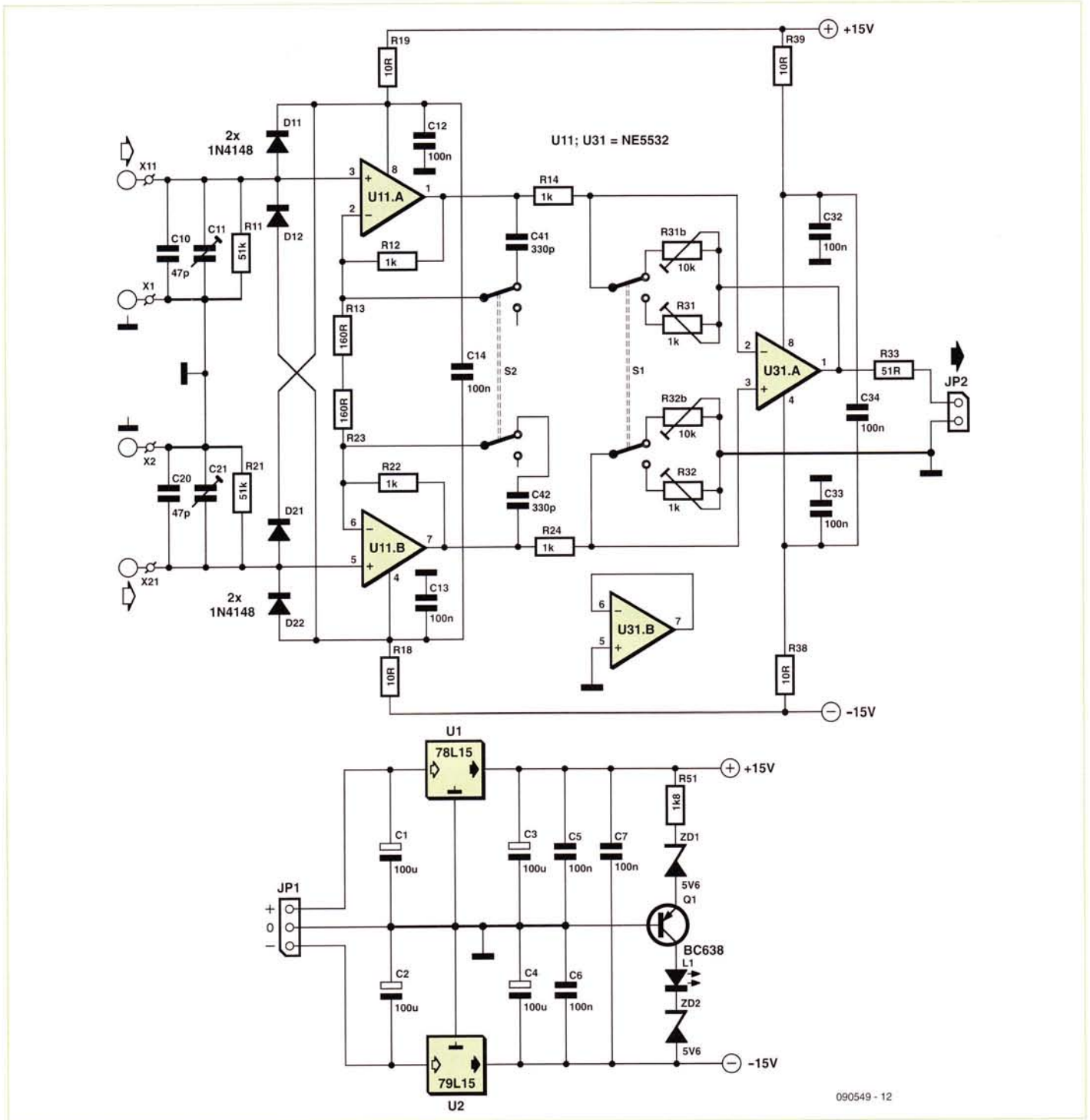
Figure 5. Full circuit (omitting the two 10 MΩ input resistors).

op-amps the voltages are decoupled once more using 10 Ω and 100 nF.

The PCB layout can be downloaded from the Elektor website, where you will also find the component list.

Figure 6 shows how the circuitry is constructed on a PCB built into a diecast aluminium box. The input stage (left) has copper foil laid below it and shielded from the main differential amplifier on the right, so as to minimise capacitive coupling. The copper strips are soldered to the PCB using earthing (grounding) pins. Take care around the inputs to provide adequate insulation and space between the high voltage conductors. For this reason the 10 MΩ input resistors and their connecting leads should be

made safe with heat-shrink tubing.

The two switches are mounted on the underside of the PCB and are operated from the lower side of the case housing. At the upper right-hand corner of the PCB are the power supply components and on the right the BNC output socket and power supply connector (around ±18 to ±20 V).

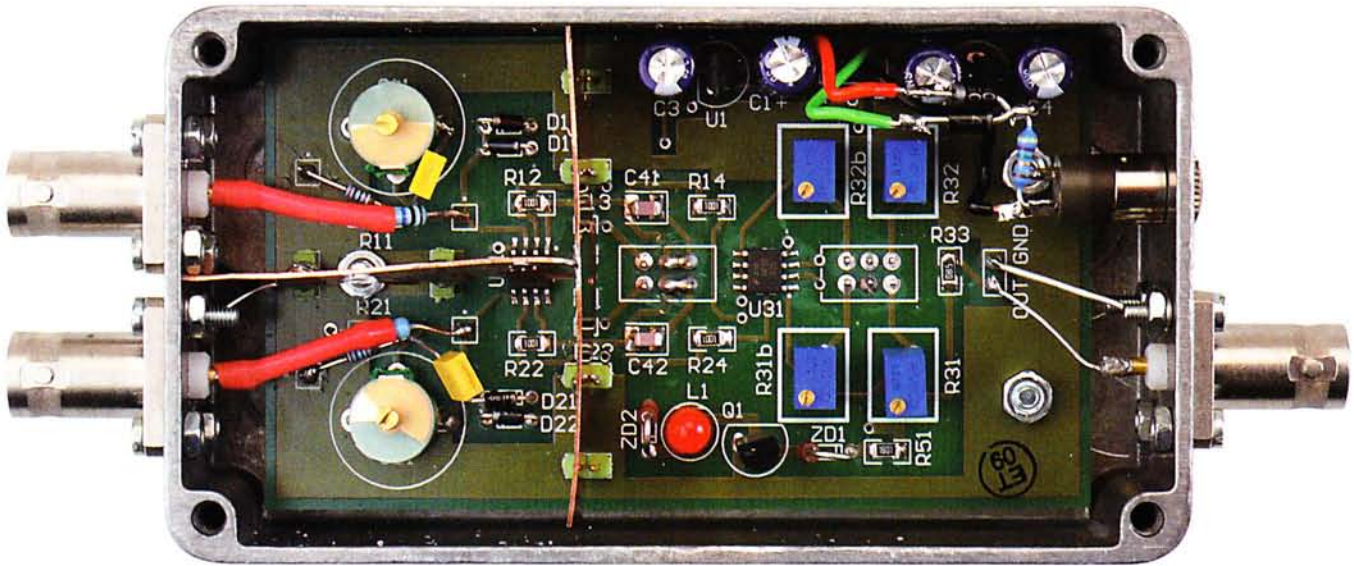Parasitic capacitance and inductance are

Figure 6. Completed circuitry with the printed circuit board inside the diecast aluminium box.

minimised by the use of surface mount (SMD) resistors and capacitors. The disadvantage of the wider tolerances is offset by precise adjustment of the trimmers. Our lab sample, using 5 % tolerance components, was successfully aligned to a CMRR of more than 60 dB at low frequencies, which corresponds to a resistor tolerance of below 0.1 %.

Applications and options

Commissioning and calibration are explained step by step in the panel 'Calibration'. Remember constantly when taking measurements that high voltages are lethal! Before altering or touching anything on the test bench, switch off the volts and make sure everything is completely dead (capacitors can retain their charge for a long time!). Always work using one hand alone (keep the other one in your trouser pocket!).

The high input impedance simplifies use of the test probe and minimises signal loading at the point of measurement. Signal level readings taken from the oscilloscope display should always take into account the attenuation level selected (–20 or –40 dB). The selected bandwidth (0.5 or 1 MHz) determines the extent to which high frequency oscillations will be visible on the oscilloscope. The input capacity of the test

probe is certainly small but can nevertheless, like any capacity, lead to signal degradation or even oscillation at particularly critical points in a circuit.

Should the signal under measurement appear to be flattened or smoothed off above and below, you need to check whether the test probe is still operating linearly or is already being overdriven, because the attenuation is inadequate and/or the input signal is too large. Meaningful measurement is impossible with overloading.

The effective range of the differential test probe can be adjusted by altering the division ratio, for instance to ±100 V for lower voltages. For higher voltages the mechanical design must be changed and the insulation improved (e.g. voltage rating of the input connectors).

With a capacitor in series with R14 and R24 the DC component of the signal is removed entirely, which would enable greater amplification for measuring small signals overlaid on high DC voltages. Capacitors at this stage influence the CMRR very little.

You can also add capacitors between the input divider node points and the op-amp inputs but then you will need an additional resistor to ground at the op-amp input for the bias current, which increases the outlay

and worsens the CMRR. A capacitor in series with the 10 MΩ input resistor must possess adequate high-voltage rating, which means bulky form factor and large inductance.

Overall this differential test probe is straightforward to construct and produces a cost-effective alternative to expensive commercial products that will serve you well at frequencies up to 1 MHz. The test probe finds regular use in the author's lab and is particularly useful for audio applications.

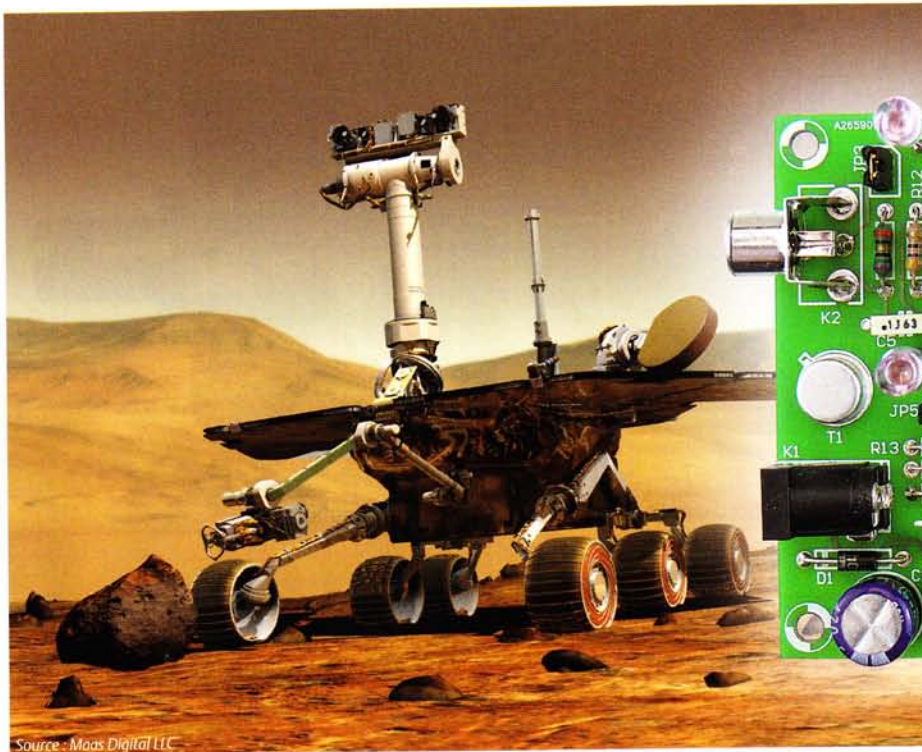(090549)

[1] www.tek.com

[2] www2.tek.com/cmswpt/tidetails. lotr?ct=TI&cs=apn&ci=2343&lc=EN

The author

Alfred Hesener is a chartered engineer employed by Fairchild Semiconductor as Application and Marketing Director for the European market.

Vision System for Small Microcontrollers
Few pixels – clear sight

Source : Maas Digital LLC

By André Thiriot (France)

The originality of the vision system described in this article lies in the method developed for acquiring video images – a method that is entirely compatible with the low calculating power of the microcontroller used and the slow sampling speed offered by its built-in analogue/digital converter.

The circuit combines a microcontroller and a miniature black-&-white video camera, and acquires and processes very low resolution video images in real time. This characteristic makes it possible to use a cheap microcontroller. Very low image resolution doesn't mean mediocre performance, and in fact proves amply sufficient in applications such as:

- monitoring the presence of an object by comparing an image with a reference image stored in EEPROM

- detecting movement by comparing two successive images stored iån RAM
- recognition of simple shapes by correlation with images stored in EEPROM
- determining the co-ordinates of a bright spot (sorting and selection of the brightest group of pixels) for system position steering or servo-control
- etc.

The black-&-white video signal

The video camera is a sensor that breaks the image down into 625 lines, or to be precise,

into two interlaced fields of 312.5 lines each – an arrangement that helps reduce image flicker on the screen. Each line is scanned by an electronic 'paintbrush' that modulates the amplitude of a voltage depending on the brightness. Synchronizing signals are added to indicate the start of a new field and the start of scanning a new line (Figure 1).

The amplitude of the video signal is 1 V. The voltage range between 0 and 0.3 V is reserved for coding the synchroniz-

ing signals (line and field sync). The range between 0.3 V and 1 V allows coding of the brightness level: a voltage of 0.3 V corresponds to black, while a voltage of 1 V corresponds to white.

The usable line scan duration is 52 µs, to which must be added the 5 µs duration of the line sync pulse and the durations of the front and back porches (5 µs and 2 µs) which frame the usable signal and make it possible to delimit each line. Hence the total duration of one line is 64 µs.

A succession of lines, separated by their synchronizing signals, form the fields, themselves separated by field synchronizing signals. These field synchronizing signals comprise a succession of line sync pulses, with the special feature of having a different mark/space ratio, making it possible to differentiate them (Figure 2).

Out of each 312.5-line field, only 288 lines are usable (lines 23–310 for odd fields and lines 336–623 for even fields), the other lines being used for encoding the field sync pulses.

A few calculations before getting started

The operation of acquiring an image consists of sampling the video signal, with the sampling sequence synchronized to the field and line syncs.

Acquiring, storing, and processing video images require substantial resources. Let's take as an example a VGA standard image (480 lines of 640 pixels) produced from a video signal. Capturing 640 pixels in 52 µs implies a sampling frequency of 640 / 52 µs, or over 12 MHz.

Storing a 16-colour VGA image coded using 4 bits requires memory space of 640 × 480 × 4 = 1228800 bits, or 153,600 bytes.

And finally, digital image processing in real time requires a high calculation speed. Thus, in order to perform a simple comparison between two VGA images (pixel-wise subtraction), we need to carry out 640 × 480 = 307,200 operations. If we want to perform this processing in less than 10 ms, for example, this implies a basic calculation speed of 10 ms / 307,200, or 33 ns

An image processing system normally requires resources that a simple microcontroller is unable to offer. However, if we can

make do with lower performance, a microcontroller like the PIC16F690 will do. Running at 8 MHz, it offers an instruction cycle time of 0.5 µs, and can perform an A/D conversion in 24 µs. It has 256 bytes of RAM, 256 bytes of EEPROM, and 4,096 words of
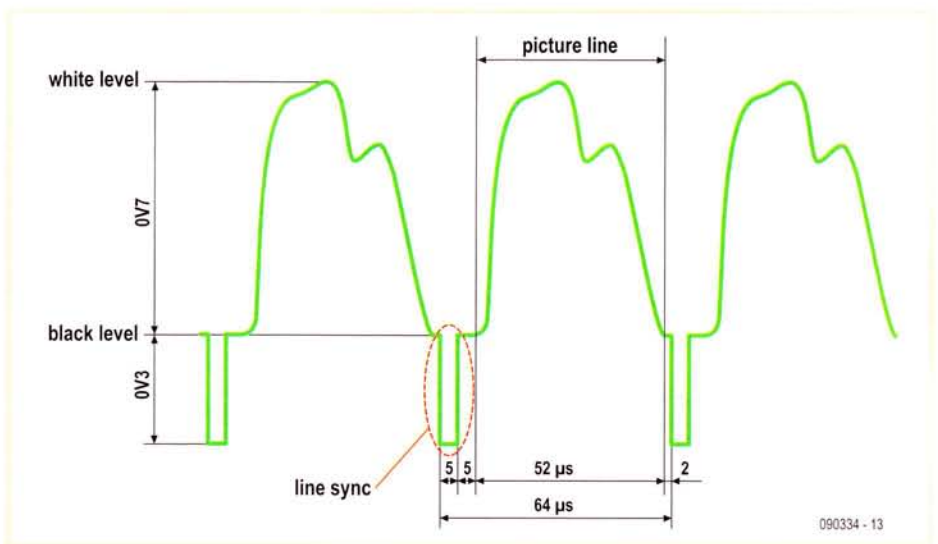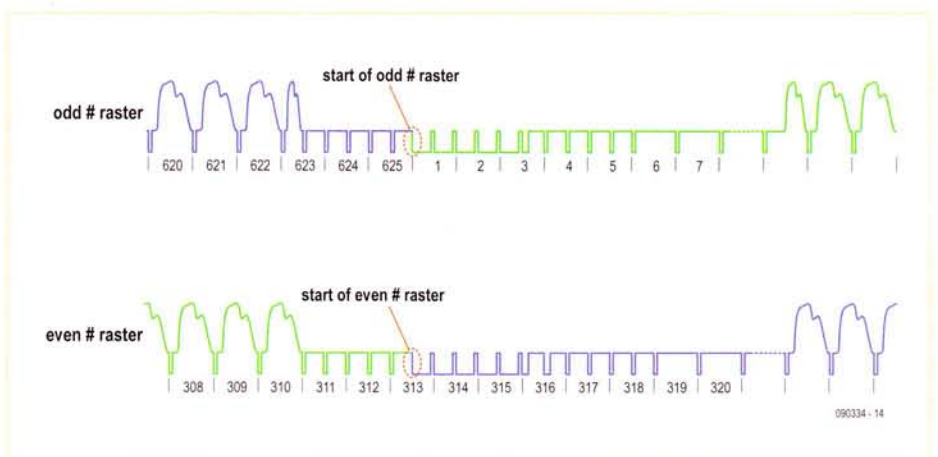

Figure 1. The B&W video signal.

Figure 2. The difference between two lines.

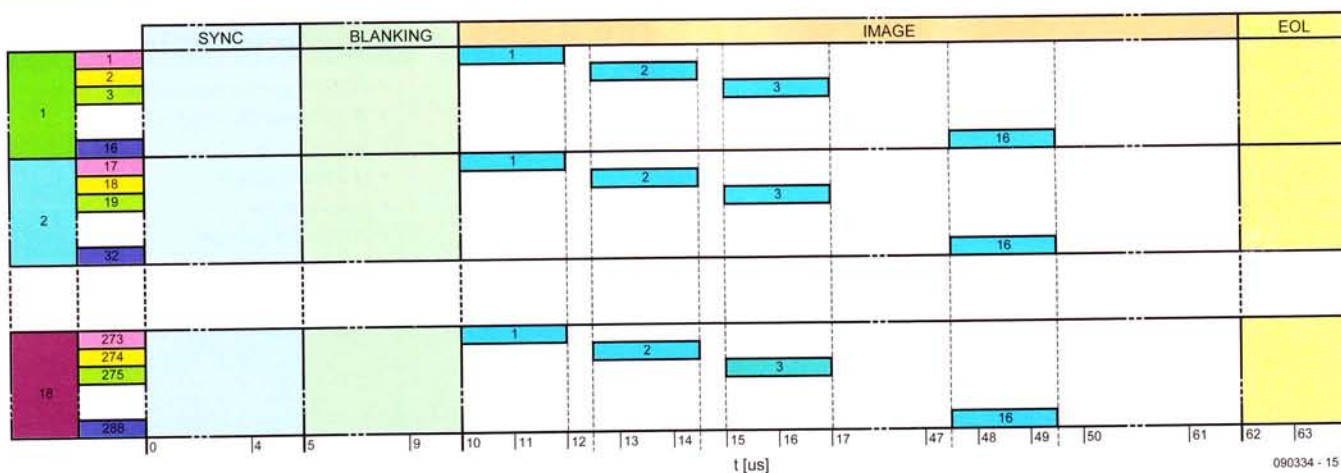Figure 3. Sampling timing diagram. Each cell corresponds to one of the points in the image. The light blue cells symbolize the duration of the analogue/digital conversion and thus represent the minimum time to be left between two successive samples. The under-sampled grid that will form the very low resolution image can be made out.
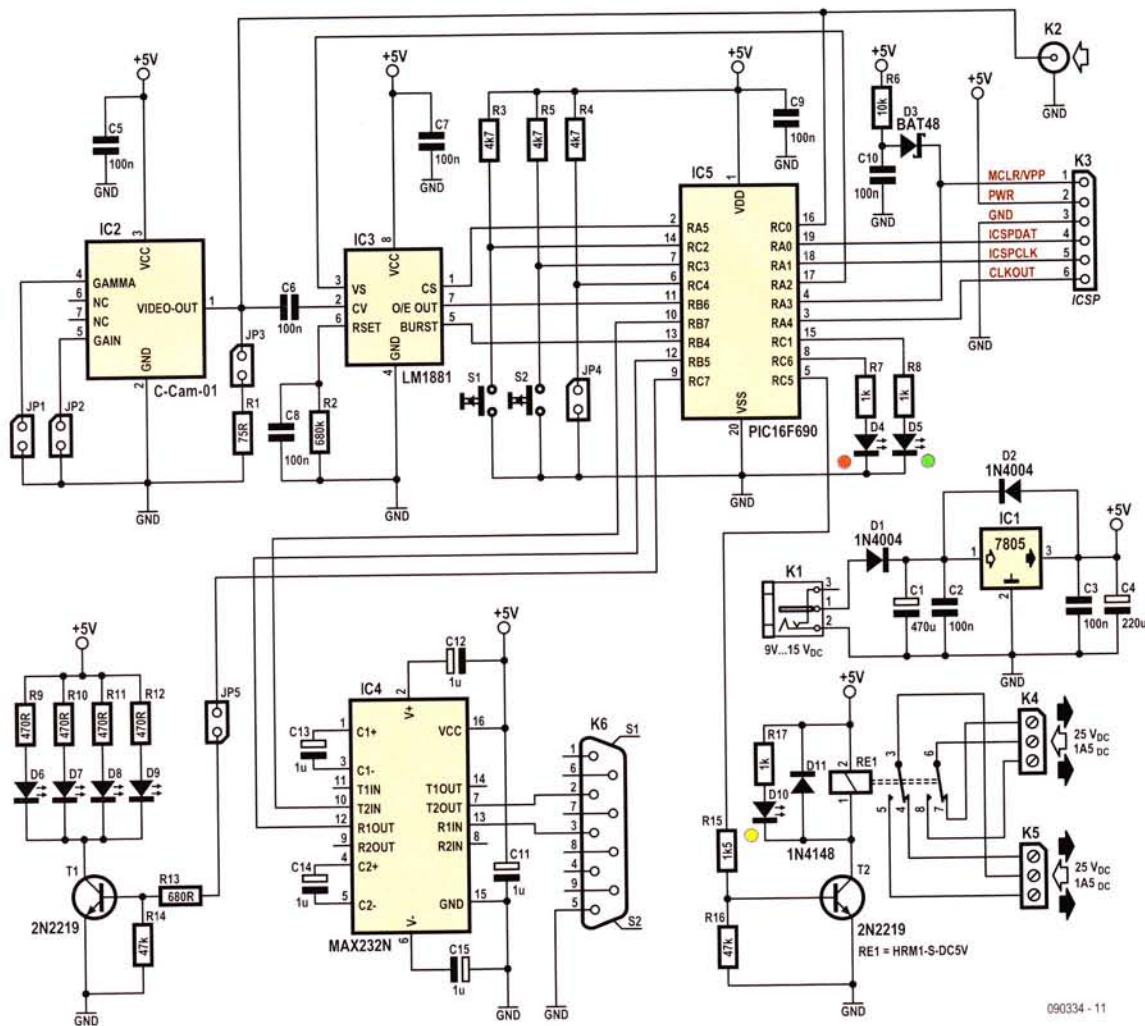

Figure 4. The circuit diagram of the project. A few suitable ICs and there you have it!

flash memory. Given these characteristics, we can see that the images are going to have to be of modest size for us to be able to process them using this microcontroller.

Suggested method

The 24 µs conversion time means sampling at a frequency of 12 MHz is not possible. However, this time is of the same order of magnitude as the 64 µs line period. So out of this simple observation comes the idea of acquiring only a single point per line, regularly offsetting the sampling moment with respect to the start of the line. This under-sampling then makes it possible to form a very low resolution image, compatible with the resources available to the microcontroller.

Sampling one point per line implies a number of points in the image less than or equal to the number of lines, i.e. 288 pixels. An image of 18 lines of 16 points (18 × 16 = 288) lets us maintain roughly the same proportions between image width and height (Figure 3).

The memory space required to store the image depends on the number of pixels, and also on the pixel brightness resolution. Using 2 bits to code the brightness level allows us to distinguish four grey levels, which are enough for the applications considered. Thus an image will occupy 2 × 288 = 576 bits, i.e. 72 bytes. This small size allows three images to be stored in the RAM memory and another three in the EEPROM.

Circuit description

The circuit (Figure 4) is based around the PIC16F690 microcontroller (IC5) from Microchip [2], chosen for its low cost and the very affordable price of the Starter Kit PICkit 2 (or 3) used for developing the software. The 6-way connector K3 is provided for connecting up the development or microcontroller programming tool. Diode D3 protects the circuit against the programming voltage Vpp of 13 V. Before connecting up the programmer, uncheck the 'Power target circuit from MPLAB ICD 2' option (Figure 5) in the programmer configuration.

This lets you power the circuit normally and use the programmer at the same time.
The image detector IC2 is a miniature B&W CMOS camera, powered from 5 V. The signal produced conforms to the description given above. Video signal acquisition is performed by the microcontroller's internal ADC via pin 16 (AN4).
Jumper JP1 lets us enable gamma correction, while JP2 lets us select the camera's image amplification gain.
The phono socket (K2) lets us connect a video monitor so as to display the image produced by the camera, or to connect an external camera: this latter option makes it possible to evaluate the circuit very cheaply, by saving the cost of the CMOS camera, which in this case will not be fitted to the PCB. When K2 is not used, JP3 must be fitted in order to load the camera video output with a 75 Ω resistor.
IC3, a classic LM1881 [3], in conjunction with the network R2 / C8, has the task of extracting the field (VS) and line (CS) syncs from the video signal, along with the parity signal (O/E OUT) used by the microcontroller to synchronize the sampling data of the video image. These signals are connected to microcontroller ports RA2, RA5, and RB6 respectively. LEDs D6–D9, driven by port RC7 (if JP5 is fitted), make it possi-

Movement detection by image comparison

ble to illuminate the zone where the image is taken.
Advantage is taken of the asynchronous series interface available in the microcontroller to send the image information via a serial link. IC4 takes care of the electrical conversion to the RS-232 standard.
Relay RE1 offers the possibility of controlling an external system — for example, an alarm. It is driven by port RC5. LED D10 tells us if the relay is energized or not.
The circuit is powered from a mains power supply (9–15 V / 200 mA), connected to connector K1. The power from the PSU is
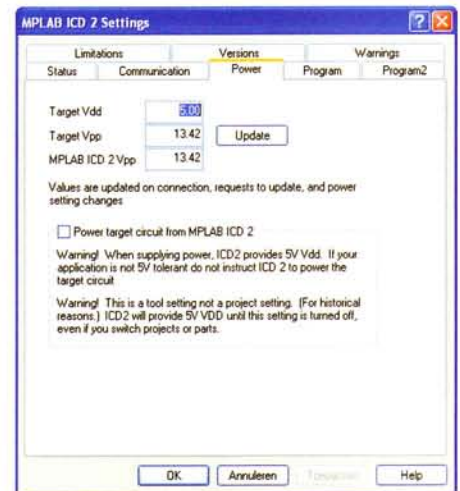

Figure 5. The programmer parameters. Uncheck the 'Power target circuit from MPLAB ICD 2' box.

smoothed then regulated by linear voltage regulator IC1.

Software

Using a microcontroller means there's bound to be software. For performance reasons, the software has been written in assembler throughout. The detail of the functions (parameters and procedures) is given in the program source code, which contains plentiful comments, available for free download from [1].
The main functions of the software are broken down into macros and subroutines: the use of subroutines was favoured in order to save program memory, but the use of macros (see Table 1) proved necessary where the execution time turned out to be critical, in particular during the image acquisition phase, where the grey level coding steps take a bit of time.
The video signal sampling and pixel grey level coding steps have been optimized: the grey level of each pixel is coded while the ADC is converting the next pixel, which saves the 24 µs wait time of the conversion operation.
The basic software illustrates the possibili-

Table 1. List of functions available for managing the vision system.

Type	Name	Function
Macro	MACRO_LineSync	Wait for line sync
Macro	MACRO_DPixelAcquisition2Bits	Launches acquisition of pixel n and grey-level conversion of pixel n-1
Function	Start	Main program comprising the initialization phase and the primary loop
Function	BinASCII	Decimal to ASCII conversion: returns the ASCII codes of the three digits of the decimal value to be converted (hundreds, tens, units)
Function	Bin2Colour2Bits	Converts a grey level (0–3) into the ASCII code used to represent it (32, 176, 177, 178)
Function	CharTrans	Transmits a character over the serial link
Function	PictureRamToEeprom	Copies the image stored in RAM to the EEPROM
Function	PictureEepromToRam	Copies the image stored in EEPROM to the RAM
Function	PictureCompare	Compares the two images stored in RAM and in EEPROM
Function	PixelGroupDistCalc	Calculates the distance (absolute value of difference in brightness level) between four groups of two pixels
Function	PictureAcquisition	Acquires an image of 288 points organized as 18 lines of 16 pixels coded using four grey levels
Function	DecimalValueTransmission	Transmits the three ASCII codes representing the three digits of a decimal value over the serial link
Function	PictureTransmissionBIT	Transmits the image stored in RAM over the serial link, with each pixel represented by the ASCII character corresponding to its grey level.

COMPONENT LIST

Resistors (5%, 0.25W)

R1 = 75Ω
R2 = 680kΩ
R3,R4,R5 = 4.7kΩ
R6 = 10kΩ
R7, R8, R17 = 1kΩ
R9–R12 = 470Ω
R13 = 680Ω
R14, R16 = 47kΩ
R15 = 1.5kΩ

Capacitors

C1 = 470µF 25V radial electrolytic, 10mm diam.
C2,C3,C5–C10 = 100nF, polyester (MKT), lead pitch 0.2 in.
C11–C15 = 1µF 16V, radial electrolytic, 5mm diam.
C4 = 220µF 25V, radial electrolytic, 8mm diam.

Semiconductor

D1, D2 = 1N4004
D3 = BAT48, Schottky diode, DO-35 case
D4 = LED, red, low current, 3mm
D5 = LED, green, low current, 3mm
D10 = LED, yellow, low current, 3mm
D6–D9 = infrared LED, QED222 (e.g. Farnell # 1652526)
D11 = 1N4148
IC1 = 7805, TO-220 case
IC2 = C-MOS IR camera module (CCIR*) e.g. Conrad Electronics # C-Cam-01 150001
IC3 = LM1881N/NOPB, DIP-8 case (e.g. Farnell # 1564700)

IC4 = MAX232N, PDIP-16 case
IC5 = PIC16F690-I/P, DIP-20 case (e.g. Farnell # 1103406)

Miscellaneous

RE1 = Relay, Multicomp type HRM1-S DC5V (e.g. Farnell # 9479937)
JP1–JP5 = 2-pin pinheader with jumper, 0.1 in. lead pitch
K1 = power supply connector, 2.1mm, horizontal, PCB mount
K2 = RCA (phono) connector, PCB mount

K3 = 6-pin pinheader, 0.1 in. lead pitch
K4, K5 = 3-way PCB screw terminal block, 5mm lead pitch
K6 = 9 way sub-D socket (female), horizontal, PCB mount
S1,S2 = pushbutton, e.g. Tyco Electronics type FSM4JH (Farnell # 1555982)
Sockets for IC3, IC4 and IC5 (8-pin, 16-pin, 20-pin respectively)
PCB, Elektor # 090334-1 (see [1])

* substitute NTSC version where applicable.
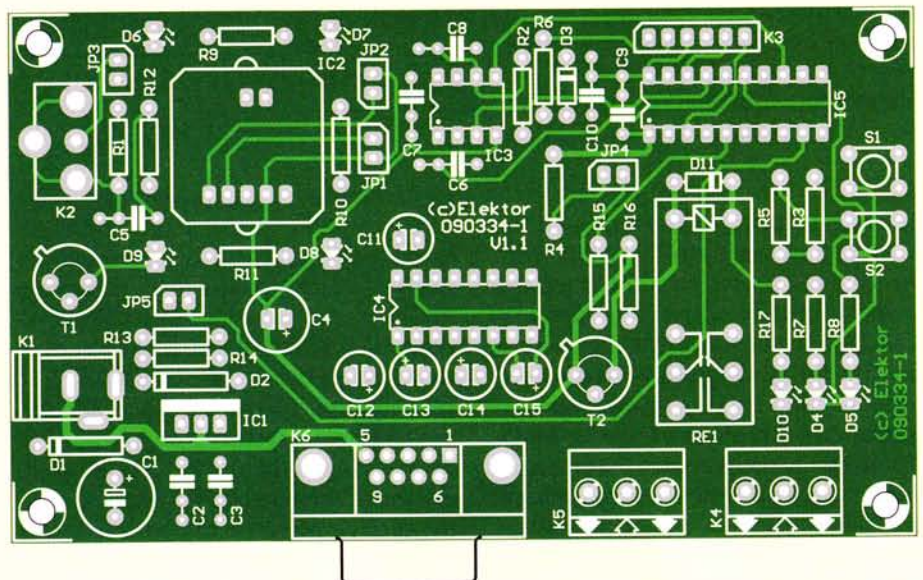
ties offered by the vision system. This software performs the following operations:

* acquires and stores a 288-pixel image in RAM
* transmits the image in ASCII format over the serial link
* compares the image stored in RAM with the image stored in EEPROM
* sends back the result of the image comparison over the serial link
* controls the lighting LEDs

The two images stored in RAM and in EEPROM are compared pixel by pixel. The comparison method used consists in accumulating the differences in the pixel brightness values. The sum total of these differences is then compared with a threshold in order to decide whether to set off an alarm or not.

The image is sent back over the serial link in ASCII format (19,200 baud, 1 start bit, 8 data bits, no parity, 1 stop bit) so that a simple terminal (like a HyperTerminal) can be used to view the image (Figure 6). The four grey levels that each pixel can have are coded using the four extended ASCII codes with values 32, 176, 177, and 178.

Pressing button S2 lets us store the latest image acquired and stored in RAM into the EEPROM; pressing button S1 lets us send the image stored in EEPROM back up the serial link. LED D5 flashes whenever either of the buttons is pressed, LED D4 indicates the start and end of image acquisition. D10 lights when the relay is energized.
JP4 allows the software to use the lighting LEDs. JP5 allows the hardware to use the lighting LEDs — the difference is a subtle one.

Conclusion

Despite its modest features, this circuit opens the door to a great deal of experimentation and even practical applications. For example, the author experimented with the object presence detecting software while looking for a way of detecting the presence of letters and packets in a letter-box, rather than the event of their being put into the box. To achieve this, a patterned background (checkerboard or concentric squares) is placed at the back of the letter-box, monitored at regular intervals by the vision system mounted in the top of the box. The image is compared with a reference image stored in EEPROM when the system is installed. There's no daylight in the letter-box, so the inside is illuminated by the circuit's LEDs while a new image is being acquired. Inserting a letter or packet modifies the image seen by the system, setting off an alarm which continues for as long as the object inserted remains in the box. This alarm goes off when the post is collected and the box is empty again.
Now it's your turn to dream up other applications, which you can share with Elektor readers!
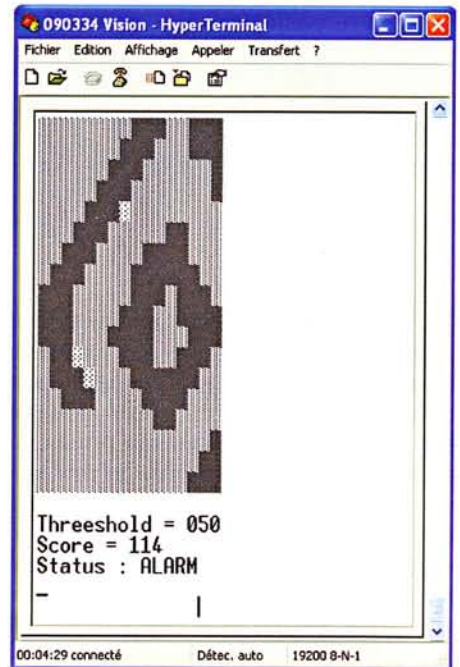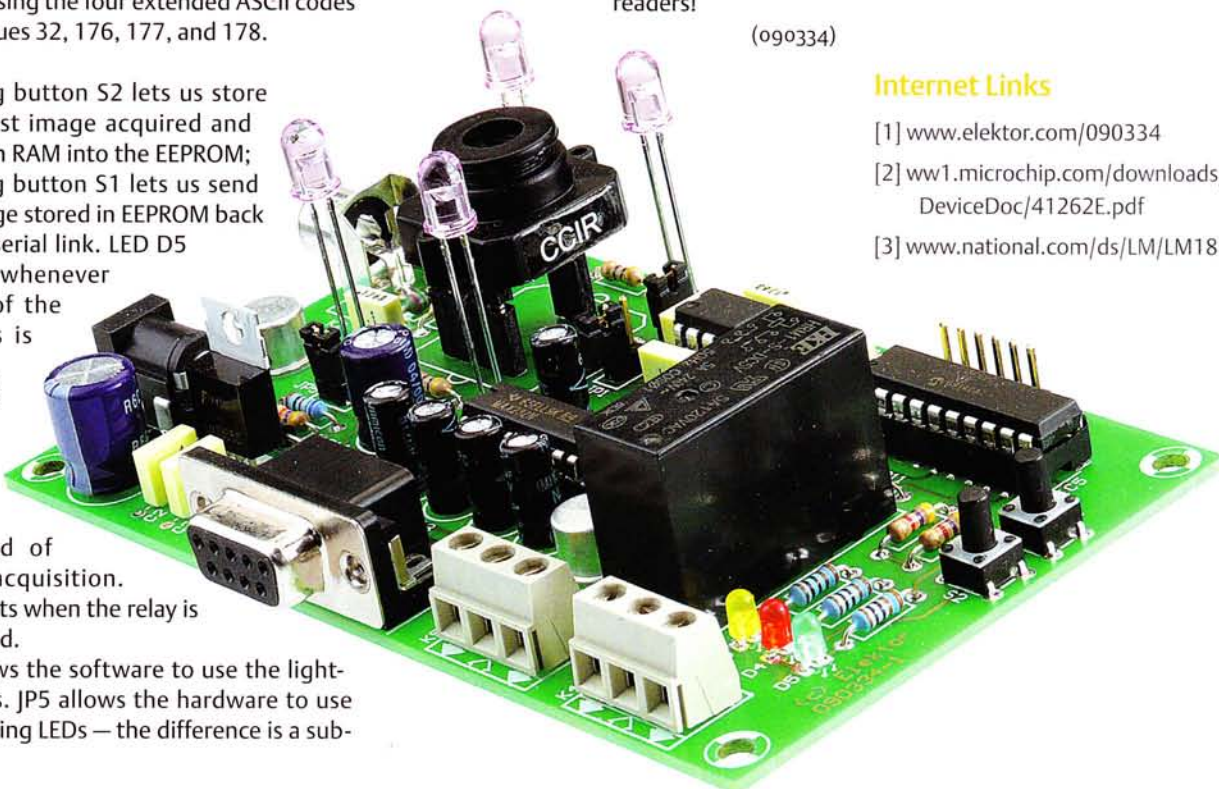
(090334)

Figure 6. The video image in HyperTerminal. Can you guess what it's of?

(A small square within a larger square.)

Internet Links

[1] www.elektor.com/090334

[2] ww1.microchip.com/downloads/en/DeviceDoc/41262E.pdf

[3] www.national.com/ds/LM/LM1881.pdf

Bascom 8051 Math Routines

By Darren Heywood (United Kingdom)

There are various ways to implement math on a given processor, some processors are optimised for math usage by way of their particular hardware design. For example, CORDIC methods are optimised for specific hardware. However when implemented on the good old 'jack of all trades' 8051 microcontroller one finds the CORDIC math algorithms are slow with little to no gain in accuracy. Help is on the way!

The routines presented here use the Padé algorithm and depending on what routine is utilised or run, the Newton/Rahpson root finder is sometimes implemented in a feedback loop. Some readers may never have heard of the Padé algorithm, it's the same idea as Taylor's expansion except Padé produces a quotient polynomial equivalent approximation of a math transcendental function which, as it turns out, is more accurate than Taylor's polynomial expansion. Both CORDIC and Padé algorithms work over a small range only — various software tricks have to be implemented to expand the ranges to a maximum. In this case, the math routines discussed here are limited only by the architecture of the IEEE 32-bit float.

Using the math routines

The general idea is to cherry pick the routines. The way to do this is to initially define an 8051.dat file with plenty of program code room, i.e., preferably 32 K or 64 K space, use the 8051 math routines as an initial template or platform to develop an application and then later on, remove the surplus math routines that are not required thereby releasing extra program code space. Care needs

to be exercised here though because some routines are nested for example, cos(x) routine calls sin(x) routine, so the particular user application software may not need a result from sin(x) but requires cos(x) only, in which case sin(x) routine must remain an integral part of the application, similarly, tan(x) utilises both sin(x) and cos(x). Therefore, to use tan(x) both sin(x) and cos(x) routines must be included. It is up to the end user to check for nested routines by skimming through a particular routine, which is easily identifiable by having the call statement embedded in the code routine.

There are two main variables used throughout, these have the variable names Arg1 and Arg2. If <call sinx Arg1> is compiled and executed then Arg1 value or contents will be passed to the sinx routine as an argument and upon execution the answer will be returned in Arg1. All trig functions use radians only.

Due to the interconnectivity nature of math transcendental functions in general, some routines are nested. Two software stacks have been implemented in software for Arg1 and Arg2 and each can go four deep and are common or global to all routines.

Bascom-8051 math routine function list

There are two main variables used with these routines these are Arg1 and Arg2, when calling a single function, such as Sinx, Arg1 would be preload with a radian number, the user then calls the routine and the result is returned in Arg1. Below is a full list of functions which is typically what you find on any scientific calculator.

Call Square Arg1 ; returns the square in Arg1.

Call Sqrx Arg1 ; returns the square root in Arg1.

Call Mag Arg1, Arg2 ; returns the square root of Arg1^2+Arg2^2 in Arg1.

Call Rect2polar Arg1, Arg2 ; converts a rectangular to polar x=Arg1 and jy=Arg2

Result returned in Arg1 and angle in Arg2 in degrees.

Call Polar2rect Arg1, Arg2 ; converts a polar to rectangular form, magnitude in Arg1 and angle in Arg2 in degrees, returns x+jy as Arg1+jArg2.

Call Cubr Arg1 ; returns the cube root in Arg1.

Call Expx Arg1 ; returns the natural anti log, base e, in Arg1.

Call Lnx Arg1 ; returns natural log , base e, in Arg1.

Call Alog Arg1 ; returns the anti log , base 10, in Arg1.

Call log Arg1 ; returns the log, base 10, in Arg1.

Trig functions

Call Sinx Arg1 ; returns the sine of a radian number in Arg1.

Call Asin Arg1 ; returns the arcsine of a number in Arg1.

Call Sin_2pix Arg1 ; returns the sine of a number with 2.pi.x, where Arg1=x, and has the range 0..to..1 to complete 360 deg or 2.pi radians.

Call Sin_a2pix Arg1 ; same as above except variable 'a' is used to denote harmonic number, wave number and is also known as sine circles.
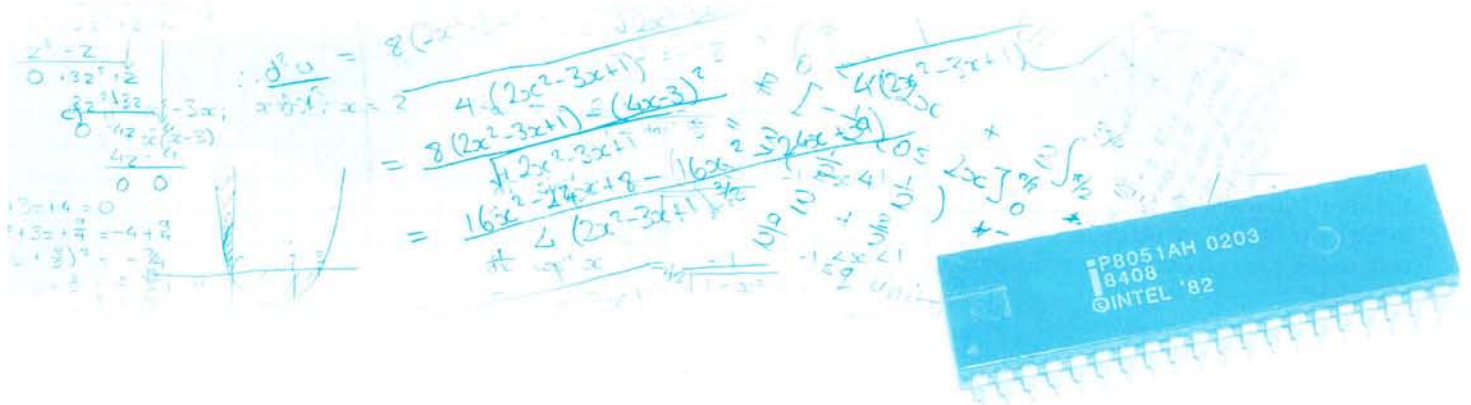
These math routines do not require or use any external hardware (except the LCD) and therefore all routines can run in the BASCOM simulator, the answers or results can be checked very easily either by directing Arg1 to the LCD screen simulator or by typing in the variable name (Arg1) as is allowed in the simulator, then verify or check the result is correct with a scientific calculator.

Examples — simple to complex

The majority of the routines are very easy to use, it's a simple case of using the BASCOM `Call` command, here are a few examples:

`Call sinx Arg1`, calls the sinx routine whereby Arg1 already contains the argument (pre-loaded) and on execution, returns the result in Arg1.

`Call cosx pi/4`, calls cosx routine, in this case, BASCOM automatically sets Arg1=pi/4 and returns result in Arg1.

`Lcd Arg1`, displays or prints Arg1 on LC display, useful for real hardware and the BASCOM simulator.

As can be seen, to use a particular math routine from your program code involves first pre-loading Arg1 with an argument and then simply calling the routine as shown above, after compilation and execution, the answer will be returned in Arg1, in some cases, depending which math routine is in usage, Arg2 as well.

When the routines are loaded into the BASCOM IDE system for the fist time, skim down to line 173, it can be seen that the command: `Call polar2rect 5, 53.13010235`

converts a complex number represented in polar form with a modulus of 5 (Arg1) and the angle of 53.13010235 degrees. As with all of the routines, the results or answers are returned in Arg1 and Arg2 or just Arg1.In this particular case the result is returned in rectangular form, i.e., Arg1+j.Arg2. Just so the user can get the initial hang of things, compile the code exactly as it stands, invoke the simulator including the LCD and execute the code using the

Call Cosx Arg1 ; returns the cosine of a radian number in Arg1.

Call Acosx Arg1 ; returns the arccosine of a number in Arg1.

Call Cos_2pix Arg1 ; returns the cosine of number with 2.pi.x where Arg1=x and has range 0..to..1 to complete 360 deg. or 2 pi radians.

Call Cos_a2pix Arg1 ; same as above except variable 'a' is used to denote harmonic number, wave number and is an integer.

Call Tanx Arg1 ; returns the tangent of a radian number in Arg1.

Call Atanx Arg1 ; returns the arctangent of a number in Arg1.

Call A2tanx Arg1, Arg2 ; returns the arctangent of numbers in Arg1=x and Arg2=y then arctangent(y/x) and returns result in Arg1.

Hyperbolic trig functions

Call sinhx Arg1 ; returns the hyperbolic sine of a radian number in Arg1.

Call coshx Arg1 ; returns the hyperbolic cosine of a radian number in Arg1.

Call tanhx Arg1 ; returns the hyperbolic tangent of a radian number in Arg1.

Call Asinhx Arg1 ; returns the arc hyperbolic sine of a radian number in Arg1.

Call Acoshx Arg1 ; returns the arc hyperbolic cosine of a radian number in Arg1.

Call Atanhx Arg1 ; returns the arc hyperbolic tangent of a radian number in Arg1.

Numerical differentiation and integration

The routines below are capable of performing both integration and differentiation numerically. The algorithm used for numerical integration is Simpson's rule.

Call Deriv func_1, func_1, inv, Arg1 ; will find the derivative at the point specified in Arg1, numerical dy/dx is returned in Arg1.

Call Simpson func_1, func_2, inv, lo, hi, strip_count ; will perform numerical integration and can find the area under a curve where hi and lo specify the integral limits and strip_count specifies how many strips to use.

simulator. The top line of the LCD should be 3.0000007 (Arg1) and the lower LCD line should be 4.0000000, note the acceptable error in Arg1, the 7th digit should be zero. If Arg1 and Arg2 are preloaded with values, then we can use the command:

```
Call polar2rect Arg1, Arg2
```

assuming Arg1=5 and Arg2=53.13010235 degrees will result in the same answer as before.

The command:
```
Call Simpson(func_1, func_2, product/quotient, Lo
integration limit, Hi integration limit, Number
of strips)
```

applies Simpson's rule, well known in mathematical text as a numerical method for performing integration, to two independent math functions selectable as a product=0 or quotient=1 with Lo and Hi integration limits, the number of strips must be specified 1 to 255. Each independent math function is assigned a specific number which can easily be looked up, there is a range for Func_1 (code start line 180) and the other range Func_2 (code start line 212). Here's an example: To find the area under a single function lnx curve between integral limits x = 1 and 2, with 10 strips. First find lnx from the list of functions under func_1 starting at code line 180, so func_1=3=lnx, next find func_2 and because it is a single function, func_2=1=1. The overall command structure becomes:

```
Call Simpson(3, 1, 1, 1, 2, 10)
```

note the product/quotient bit can be set either 0 means product or 1 means quotient, in this specific case, since lnx*1=lnx or lnx/1=lnx.

The above shows an example of finding the area under a lnx curve for single function between the integral limits 1 to 2, however, because the command allows product/quotient of two functions but more specifically, a product of two functions means that some complex math can be computed such as the Fourier series (as just one example), which is essentially the integration of two functions multiplied together between two limits.
Obviously, having integration, numerical derivative which can find any two function product or quotient is available, the command:

```
Call Deriv(func_1, func_2, product/quotient bit
0/1, X gradient position point)
```

Example: find the gradient of f(x)=x^2 when x=2.
```
Call Deriv(23 , 1 , 0 , 2)
```
, here we have func_1=23=x^2 and func_1=1=1 which means x^2*1=x^2 also x^2/1=x^2 is a single function as before, product/quotient bit set to 0 means 1*x^2 but if this bit was 0 it would also be correct as x^2/1=x^2. Next 2 means, what is the gradient of x^2 when x=2, the answer is 4.

There are some functions which maybe confusing like Call

Sin_2pix Arg1 or Call Cos_a2pix Arg1. When working with some areas of mathematics it better to represent or define Sinx with Sin_2pix which simply means 2*pi*x, so x can take on the values between 0 and 1, where 1 will represent one complete cycle i.e. 2.pi Cos_a2pix=cos_a*2*pi*x, where 'a' takes on integer values like 1,2,3,.... and 'a' can be any harmonic number.

Example: Find the average DC value of a half rectified cosine wave whose peak amplitude is 4 volts. To do this, we need to find the area under one half of a cosine wave and then divide the calculated area by length of base which will result in the average value for a half wave unity amplitude cosine wave. All that remains is to multiply the result by 4 for a final answer. Proceed with the command:

```
Call Simpson(18 , 1 , 0 , -0.25 , 0.25 , 16)
```

Here we have func_1=18=Cos_2pix Arg1, func_2=1=1, we are working with a single function and the bit inv=0 so func_1*func_2. The integration limits are from –0.25 to 0.25 representing ½ cosine wave exactly and lastly, 16 strips are used. The result is returned in Arg1 (as always) as 0.31831 so divided by length of base which in this case =1, the peak of the cosine wave is 4 so finally:
Vave = 4*0.31831=1.27324 volts.

Supposing, the user wants to find the derivative of function which is not available from given list of common transcendental functions, equations such as unique polynomials might be of interest. Consider, as an example,

$$f(x) = (1+x^{(2)})/(x-2).$$

There are two routines made available called Fract_1 and Fract_2, Fract_1 contains the equation for (1+(Arg1)^2) and Fract_2 contains (Arg1-2). So to find the numerical derivative of f(x) at point x=-1.25 proceed as follows:

```
Call Deriv(27 , 25 , 1 , -1.25)
```

As explained elsewhere, func_1=27=Fract_1 and func_2=25=Fract_2. The inv bit is set to 1 which tells the Deriv routine that f(x)=Fract_1/Fract_2 (a quotient) and finally what is the derivative at point x=−1.25? Run this routine to find the derivative at point –1.25, the answer should be 0.5266.

Where can I get this
The full set of Bascom math routines is available as a single .bas file for free downloading from the Elektor website [1].
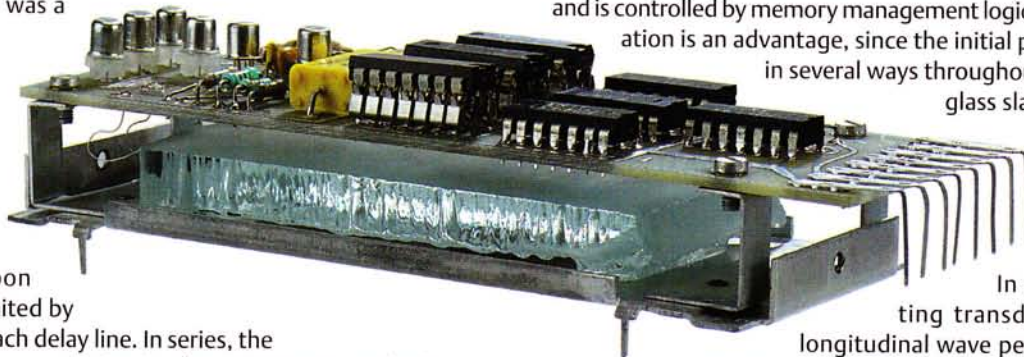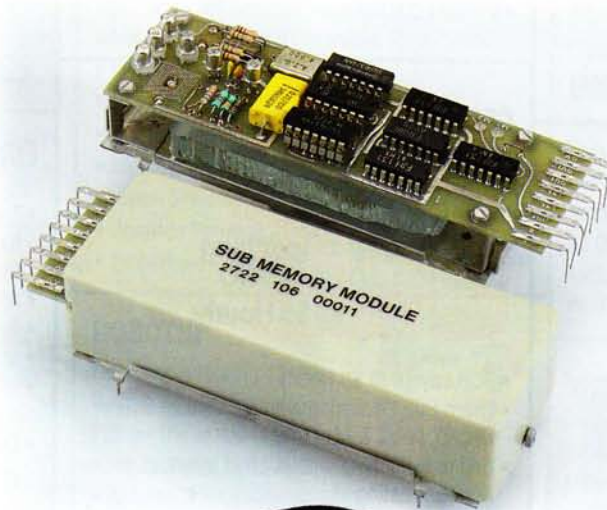
(100143)

Internet Link

[1] www.elektor.com/100143

Delay-line Digital Memory (ca. 1968)

By Jean Herman (Belgium)

In November 1968, the Belgian company MBLE (Manufacture Belge de Lampes et de Matériel Electronique) published an article in its "Special new techniques" magazine presenting a new device: a delay line capable of memorizing digital information for logic and calculation. The device was similar to the delay lines used in TV colour decoders, but complete with all the electronics needed to make it work. This was not a new discovery, as digital computers were already using bulky delay lines that employed a column of mercury or any other substance intended to delay information. But MBLE's innovation was to have made it into a reliable, stable, and quite accurate device, in a standard, interchangeable module. Three modules were available: the GDM11 with a capacity of 256 bits, a binary data rate of 0.5 MHz and a delay of 515 µs; the GDM12 (256 bits, 0.5 or 4 MHz, 515 or 64.5 µs); and the GDM21 (256 bits, 4 MHz, 64.5 µs). The GDM11 and GDM21 modules were master modules, the GDM12 was a slave module.

To increase memory capacity, several modules could be used in series. This was the simplest solution, but in practice, one soon found oneself limited by the accuracy of each delay line. In series, the delay time error is cumulative. So the commonest method was to use the modules in parallel, where there is no limit to the capacity. To achieve perfect synchronization of the modules, they used a common clock pulse from the single master module, making it possible to use just one bit-rate frequency converter for all the modules. This type of memory has been used in calculating machines or small computers, as an analogue/digital converter buffer, in digital machine tools, as a screen memory for CRT displays, etc.

The material used for transmitting and holding the information is a special type of glass that has a high lead content, to obtain the lowest possible acoustic (ultrasonic) propagation speed. Two ultrasonic transducers are bonded to the same 40 mm end of an 80 × 40 × 8 mm oblong slab of glass. Under the transducers, the glass is angled at 7.5° so that the transducers are perpendicular to the reflection paths. One piezoelectric transducer emits pulses and the other receives them.

Internal reflection takes place between the two 40 mm ends of the slab of glass, so the distance between the transducers is 2 × 80 mm = 160 mm. An ultrasonic bit travels this distance in 64.5 µs. Calculating back from this, the sound waves travel at a speed of 0.16 m/64.5 µs = 2,480.6 m/s.

Obviously, the long 160 mm path through the glass attenuates the signals. The voltage attenuation measured between the piezo transducers is around 6 dB. A BSX20 transistor amplifies the echo received and brings it up to TTL level.

The 256-bit word is continuously re-injected into the glass and is controlled by memory management logic. The signal attenuation is an advantage, since the initial pulse is propagated in several ways throughout the whole of the glass slab, but is no longer of sufficient amplitude to interfere with the working of the system. In reality, the emitting transducer generates a longitudinal wave perpendicular to the transducer, but also a transverse wave at a different speed (around half the speed) which might interfere with the main wave.

In 1968, one had to be very sparing with the precious memory capacity. I have worked with large computers that managed a whole great rolling mill and yet only used 16 KB of memory! I have also had occasion to repair Schneider calculating machines that used this delay line.

(100081-I)

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

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250 pages • ISBN 978-0-905705-91-0 • £29.50 • US $47.60

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50 PIC Microcontroller projects
FOR BEGINNERS AND EXPERTS
Bert van Dam

elektor

Bestseller!

For beginners and experts

50 PIC Microcontroller projects

This book contains 50 projects for PIC microcontrollers such as a laser alarm, USB teasing mouse, eggtimer, guarding a room using a camera, mains light dimmer, talking microcontroller and much more. You can use this book to build the projects for your own use, but also as a studybook or reference guide. Several different techniques are discussed such as relay, RS232, USB, pulse width modulation, rotary encoder, interrupts, infrared, analog-digital conversion (and the other way around), 7-segment display and even CAN bus.
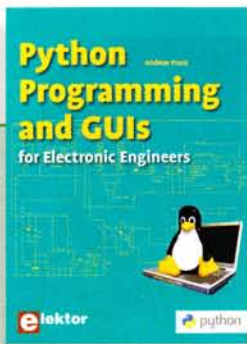
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PIC Cookbook for Virtual Instrumentation

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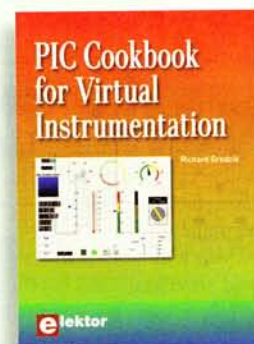
Several case studies included

PIC Cookbook for Virtual Instrumentation

The software simulation of gauges, control-knobs, meters and indicators which behave just like real hardware components on a PC's screen is known as virtual instrumentation. In this book, the Delphi program is used to create these mimics and PIC based external sensors are connected via a USB/RS232 converter communication link to a PC.

264 pages • ISBN 978-0-905705-84-2
£29.50 • US $47.60

Books

See the light on Solid State Lighting

DVD LED Toolbox

This DVD-ROM contains carefully-sorted comprehensive technical documentation about and around LEDs. For standard models, and for a selection of LED modules, this Toolbox gathers together data sheets from all the manufacturers, application notes, design guides, white papers and so on. It offers several hundred drivers for powering and controlling LEDs in different configurations, along with ready-to-use modules (power supply units, DMX controllers, dimmers). In addition to optical systems, light detectors, hardware, etc., this DVD also addresses the main shortcoming of power LEDs: heating. This DVD contains more than 100 articles on the subject of LEDs.

ISBN 978-90-5381-245-7
£28.50 • US $46.00
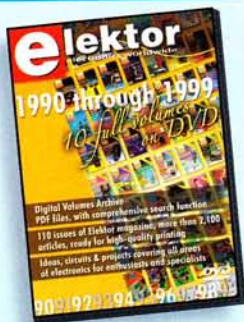
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This DVD-ROM contains the full range of 1990-1999 volumes (all 110 issues) of Elektor Electronics magazine (PDF). The more than 2,100 separate articles have been classified chronologically by their dates of publication (month/year), but are also listed alphabetically by topic. A comprehensive index enables you to search the entire DVD.

ISBN 978-0-905705-76-7
£69.00 • US $100.00

New!

The Elektor DSP radio

(July/August 2010)

Many radio amateurs in practice use two receivers, one portable and the other a fixed receiver with a PC control facility. The Elektor DSP radio can operate in either capacity, with a USB interface giving the option of PC control. An additional feature of the USB interface is that it can be used as the source of power for the receiver, the audio output being connected to the PC's powered speakers. To allow portable 6 V battery operation the circuit also provides for an audio amplifier with one or two loudspeakers.

PCB, assembled and tested

Art.# 100126-91 • £149.00 • US $240.40

InterSceptre

(June 2010)

In our March issue, we introduced Sceptre, a fast prototyping system fitted with a 32-bit microcontroller. Even on its own, this board will let you produce some great results, but if we add an extension board to make it easier to access all its peripherals, the Sceptre platform becomes downright powerful. What's more, if you fit this extension board into a suitable case, you'll be able right from the start to develop a prototype that you can use 'properly' in a installation, with no trailing wires or bits of sticky tape holding everything together. Now that's what you call fast, convenient prototyping!

Kit of parts, contains PCB and components

Art.# 100174-71 • £116.00 • US $ 187.10

dsPIC Control Board

(May 2010)

This control board has been designed for incorporation into typical industrial electronics applications like controlling motors or adjustment of static up- or down-converters. The objectives were to obtain a board with a large number of pulsewidth modulation (PWM) generators, which enables us to control several motors and static converters at the same time. The cost of the control board needed to be as low as possible too. In addition, it must be possible to construct the board using a soldering iron, without requiring use of a reflow oven.

PCB, populated and tested

Art.# 090073-91 • £140.00 • US $225.90

Bestseller!

Reign with the Sceptre

(March 2010)

This open-source & open-hardware project aims to be more than just a little board with a big microcontroller and a few useful peripherals — it seeks to be a fast prototyping system. To justify this title, in addition to a very useful little board, we also need user-friendly development tools and libraries that allow fast implementation of the board's peripherals. Ambitious? Maybe, but nothing should deter you from becoming Master of Embedded Systems Universe with the help of the Elektor Sceptre.

PCB, populated and tested, test software loaded (excluding Bluetooth module)

Art.# 090559-91 • £89.00 • US $143.60

September 2010 (No. 405)

	£	US $

Elektor Project Case
100500-71 .... Predrilled Lexan sheets with standoffs .... 14.90 ...... 24.10

Digital Multi-Effects Unit
090835-31 .... EEPROM 24LC32 .... 4.00 ...... 6.50
090835-41 .... ATmega8-16PU .... 8.30 ...... 13.40
090835-42 .... ATtiny2313-20PU .... 8.30 ...... 13.40
090835-71 .... Kit of parts including PCBs, programmed controllers
and EEPROM .... www.elektor.com

Dual Voltage/Current Display
100166-71 .... Kit of parts incl. PCB, item -41, LCD .... 62.00 ...... 100.00

July/August 2010 (No. 403/404)

The Elektor DSP radio
100126-41 .... ATmega168 PU .... 12.50 ...... 20.20
100126-91 .... PCB, assembled and tested .... 149.00 ...... 240.40

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080307-41 .... PIC16F628A-DIL-18, programmed .... 8.00 ...... 12.90

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090643-41 .... PIC16F72, programmed .... 8.00 ...... 12.90

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090940-41 .... ATtiny2313-20SU, programmed .... 8.00 ...... 12.90

Digital Thumbwheel Switch
090538-41 .... ATtiny2313 dip20, programmed .... 8.00 ...... 12.90

Whistler: Electronic Trainer/Coach
100203-41 .... PIC16F88 DIP18, programmed .... 8.00 ...... 12.90

Solar Cell Battery Charger/Monitor
090544-41 .... PIC16F877A, programmed .... 16.50 ...... 26.70

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090534-41 .... ATTiny2313, programmed .... 8.00 ...... 12.90

Tiny Timer
091044-41 .... ATtiny2313, programmed .... 8.00 ...... 12.90

Universal PWM Driver
090856-41 .... PIC16F628-1/P, programmed .... 8.00 ...... 12.90

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090187-41 .... PIC165F877-20/P DIP40, programmed .... 18.50 ...... 29.90

USB Tilt Sensor
070829-41 .... ATmega8-16AU (TQFP), programmed .... 8.00 ...... 12.90
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090863-41 .... PIC16F616-I/P, programmed .... 8.00 ...... 12.90

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090117-41 .... PIC16F690 DIP, programmed .... 8.00 ...... 12.90

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090444-41 .... ATTINY13-20P4, programmed .... 8.00 ...... 12.90

MicroMinimal Thermometer
090634-41 .... ATTINY13(A)dip8. programmed .... 8.00 ...... 12.90

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090834-41 .... Programmed controller ATtiny45 DIP-8 .... 8.00 ...... 12.90

Modeller's Clock
090023-41 .... PIC18LF1320 I/P DIL18, programmed .... 8.00 ...... 12.90

Astrolamp
090550-41 .... ATTiny45 DIL8, programmed .... 8.00 ...... 12.90

ATM18-DIPATM18-DIP
090896-1 ...... Printed circuit board .... 9.90 ...... 16.00

Underfloor heating controller
100318-41 .... ATtiny25-20p4DUO8, programmed .... 8.00 ...... 12.90

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ISBN 978-90-5381-159-7 .... £24.90 ..... US $40.20

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Art. # 090559-91 .... £89.00 ... US $143.60

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Art. # 100126-91 .... £149.00   US $240.40

3 — **InterSceptre**
Art. # 100174-71 .... £116.00 ... US $187.10

4 — **dsPIC Control Board**
Art. # 090073-91 .... £140.00   US $225.90

5 — **UniLab**
Art. # 090786-71 .... £64.00   US $103.30

Current Clamps Compared

Although a multimeter is an extremely useful and versatile instrument, the range of the ammeter section usually ends at about 10 A. For larger currents you should use a so-called clamp that can easily handle 100 A or even 1000 A. In the October 2010 edition we describe our experiences with twenty current clamp meters from different price ranges, varying from tens of pounds to about 500 pounds.
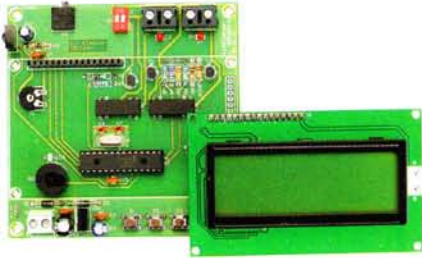
Photo Timer

Many modern digital cameras allow all features to be controlled remotely. Using the extensive photo timer published in the October 2010 edition, accurately timed photos can be made with Canon EOS or compatible cameras. A 4-line display shows all settings, which are also stored in an EEPROM. Optional pre-focusing is available and sound effects announce the actual 'shot'.
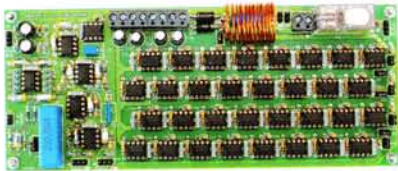
NE5532 Power Amplifier

In this article Elektor's 'off the beaten track' approach to electronics flourishes. Sure, the NE5532 opamp is widely used in audio applications but it's less well known that an interesting power amplifier can be made by connecting enough of them in parallel. For this project 32 NE5532s are bundled to make a quality amplifier with an output power of about 15 watts into 8 ohms. The first instalment of the article describes the design philosophy and the schematics of this remarkable audio power amp.

Article titles and magazine contents subject to change; please check the Magazine tab on www.elektor.com

Elektor UK/European edition: on sale September 23, 2010.   Elektor USA edition: published September 10, 2010.

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Elektor on the web

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