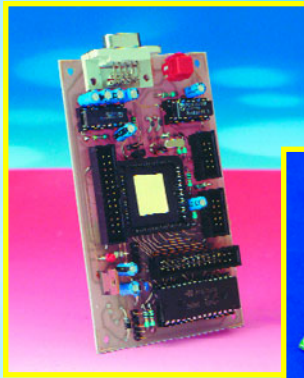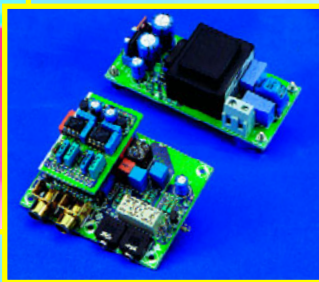# ELEKTOR ELECTRONICS

PC TOPICS:
make your own music CDs
the new ST7 microcontroller
EEPROM for BASIC control computer

68HC11 evaluation system

DIY: from vinyl to compact disc

weather-satellite decoder
from radio signal to computer display

EEDTS Pro
our new model train control system

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The ST family of microcontrollers is used a lot in Elektor Electronics projects. The well-known ST6 controller has now acquired a sturdy younger brother in the form of the ST7 processor. In this article, we take a closer look at the new member of the family.

new ST7 controllers

faster, more versatile and with more features

Microcontrollers play a steadily increasing role in a large number of products. The major growth markets are the automotive industry and mobile communications. Monitor manufacturers also build a microcontroller into practically every unit, frequently using an ST7 type. In all of these systems, there is a need for a large processing capacity and high reliability. In addition, low energy consumption is a factor in mobile communications.

In order to better meet the changing demands of the market, ST Microelectronics has recently expanded the range of the ST7 family with versions that include new interfaces. The CAN bus, for example, makes these controllers a good choice for use in the automotive industry and other environments that are subject to strong interference. Support for the SCI interface,

by contrast, makes these controllers particularly well suited for use in some other market segments. A new, programmable interrupt system introduces the possibility of flexible configuration of the priorities of the various interrupts. The controllers are available in six different types of packages, including types with pins and SMD versions. Depending on the features included, the packages have 28, 32, 42, 44, 56 or 64 connections. An extensive mix of internal memory configurations and hardware options is available. This allows the user to make an optimum choice for the cost/performance trade-off. All devices are available with various options for the sizes of the data and program memory. A ROM version is also available for large production runs, and an OTP (one time programmable) version can be used for small produc-

tion quantities. Some versions also have an internal EEPROM. All in all, this results in a product palette of around 30 different versions of the processor.

The available I/O functions are as follows:

- 16-bit timer
- watchdog timer
- A/D converter
- D/A converter
- asynchronous serial interface
- SPI (serial peripheral interface)
- I²C
- USB
- CAN

The family tree

ST Microelectronics has divided the ST family into four main branches.
The well-known ST6 is an energy-efficient 8-bit processor with 1.2 to 8kB of ROM and a limited number of I/O functions. Typical applications for this processor are simple control circuits.
The new ST7 is also an 8-bit processor. It uses an industry-standard instruction set, has from 256 bytes to 3 kB of RAM, 4 to 40 kB of ROM and numerous I/O functions. Typical applications for this controller are to be found in the automotive industry, remote controls for television sets and (for example) RDS decoders.
The powerful ST9 is an 8/16-bit processor with an instruction time of 250 ns for a 16-bit word. It supports many addressing modes, includes DMA and has an advanced interrupt system and many I/O functions. In addition, it has 16 to 128 kB of ROM and at least

256 bytes of RAM. The automotive industry is a large consumer of ST9 controllers.

The last branch of the ST family is the ST10. This 16-bit processor has an instruction time of 100 ns. It has 72 kB of flash memory and at least 10 kB of RAM, and a number of integrated I/O functions. This processor is an outstanding choice for demanding control applications, such as energy management and air bag controllers.

Controller design

The ST7 processor complies with the Von Neumann architecture, which means that all system components (such as memory, timers and I/O) are located in a single memory space.

The processor has an 8-bit internal data structure. It has six registers (A, X, Y, PC, SP and CC), including a 16-bit program counter (PC) and a 16-bit stack pointer (SP). The ST7 is considerably faster than its predecessor, the ST6. This is due to the fact that it works with a parallel internal bus, while the ST6 has a serial internal structure. The ST7 processor uses 63 instructions with 17 different addressing options. The instruction set includes several powerful instructions, such as an unsigned 8 ⇔ 8 multiplication, bit manipulations, various bit/byte conversions and powerful branching options. The peripheral hardware is controlled via specialized interrupts and registers. **Figure 1** shows the overall organization of the processor.

Since the address bus is 16 bits wide, the processor can address up to 64 kB. This is more than adequate for the intended applications of the ST7. To make the code especially efficient, a special 256-byte block (0–0FF$_H$) is reserved as 'page 0'. This means that the memory space is divided into two regions. The reserved portion can be accessed using an 8-bit address, while the remainder of the memory can only be accessed using the 16-bit address mode. All I/O ports are located in page 0 in a reserved block between 000$_H$ and 080$_H$. The memory structure is depicted in **Figure 2**.

The processor works at a clock rate of 16 MHz. It can be put into a WAIT, SLOW or HALT state under software control. This allows its energy consumption to be drastically reduced.

Register usage

As already mentioned, the ST7 has six registers.

The **accumulator** (A) is naturally the best-known register in the processor. It
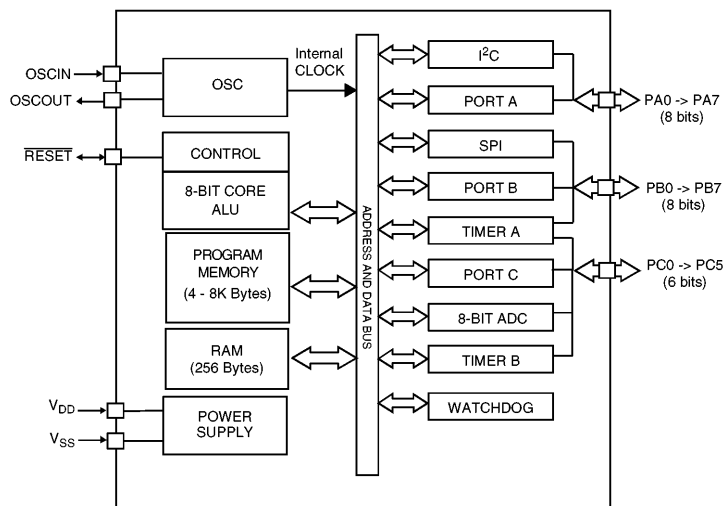
Figure 1. The ST7 is a powerful microcontroller, which combines a lot of processing power with numerous I/O functions. Some of the I/O functions are not included in certain versions.

is used for logical and arithmetic instructions. Instructions that consist of only two operands and also use two operators are only possible if one of the two operands is held in the accumulator. Instructions that employ only one operand, such as increment, decrement, complement, compare, test for negative or zero, bit compare and so on, can either directly reference a memory location or use the accumulator.

The **condition code register (CC)** holds flag bits that are more or less independent of each other. For example, if the result of an addition is zero, the Z flag will be active, but it will be reset for any other result. If the result of an operation is negative, the N flag will be activated, and it will similarly be reset for any other result. The CC register saves the flag bits after each instruction has been executed. They can subsequently be used by a conditional jump instruction. The following flags are contained in the CC register:
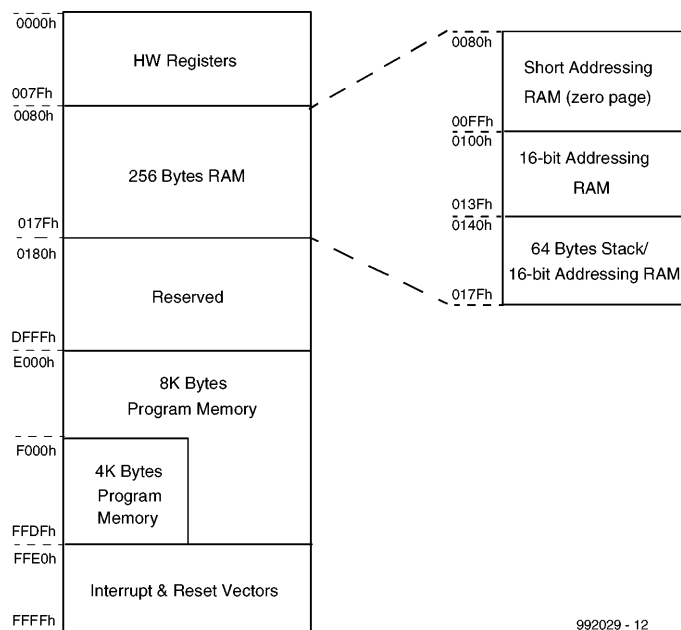
Figure 2. As expected, an 8-bit processor has 64 kB of address space. This diagram shows how the memory space is allocated.
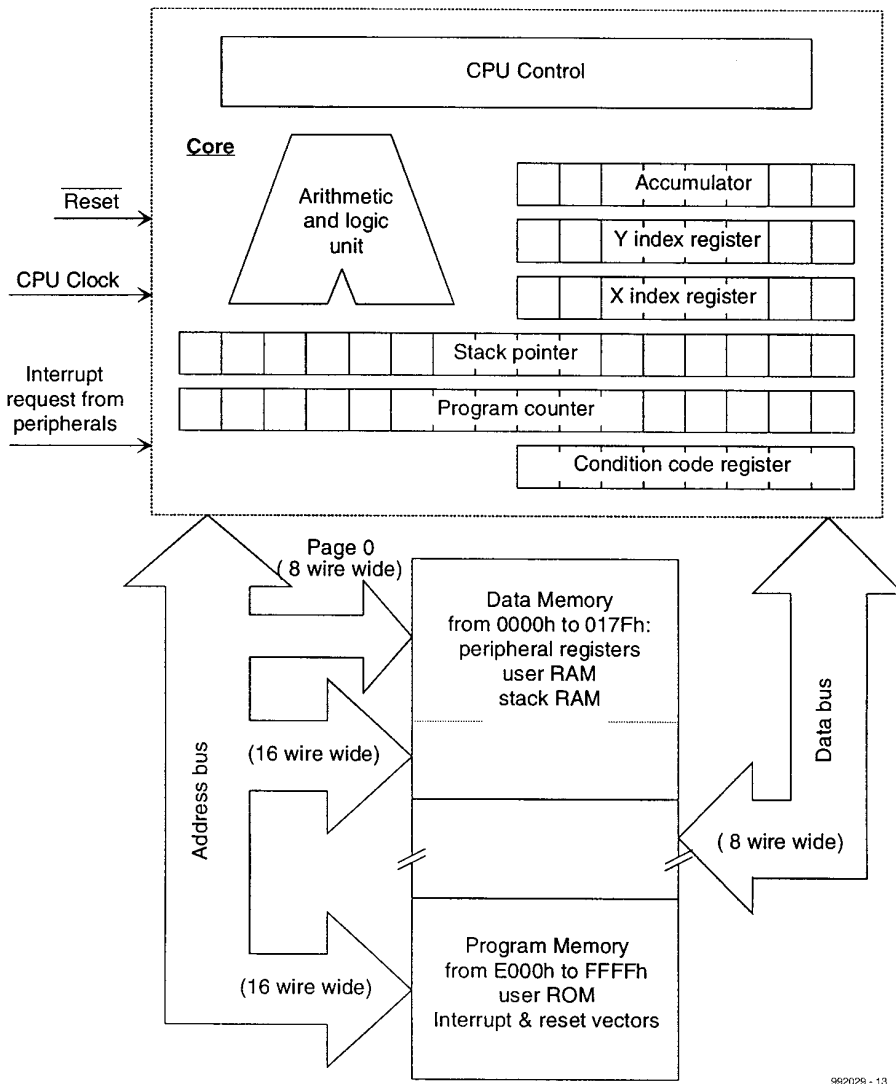
Figure 3. The core of the microcontroller consists of the registers, memory, ALU and a number of other functions, all interconnected by several busses. This structure forms the heart of all ST7 processors.

- C flag: represents the carry bit;
- Z flag: active if the result is zero;
- N flag: active is the result is negative;
- I flag: used to enable all interrupts;
- H flag: half-carry bit, used to pass the carry between two nibbles.

The **index registers** (X and Y) are used for storing addresses during instruction execution. They complement the accumulator, which is used to hold the data that will be processed by an instruction. The **program counter** (PC) is the register that controls the execution of a program. It contains the address of the next instruction to be executed. The content of this 16-bit register can be modified by instructions, such as conditional jumps.

The **stack pointer** (SP) is the final register. Its function should be obvious. In plays an important role in the use of the stack memory. A memory block from 0140_H through $017F_H$ is reserved for the stack.

Figure 3 shows the core of the processor, consisting of the registers, memory, ALU and a number of other functions, and illustrates how these elements are interconnected.

Integrated I/O functions

The following summary describes the numerous I/O functions that can be included in the processor. Depending on the selected processor version, certain of these functions may not be present.

Parallel I/O. The bidirectional I/O lines are grouped in sets of eight, as usual. The number of available I/O lines depends on the package type. The I/O lines may be used by a number of I/O functions in common (alternative applications), or by an I/O function defined in software. Data can be read in and output via these connections. The Data Direction and Option registers can be used to customize the characteristics of each I/O pin.

16-bit timer. The 16-bit timer can be used for a large variety of timing functions. It consists of a free-running 16-bit counter and a prescaler. Each timer has two inputs for count pulses and two outputs. Pulses can thus be measured or generated using the timer. Timer overflows and other events associated with the timer are logged in the status register, and can be optionally used to trigger an interrupt.

Watchdog timer. The built-in watchdog timer contains a 7-bit reloadable register that initializes a processor reset as soon as a previously-defined value is reached. During normal operation, the software must reload the register within a certain interval, to prevent the reset condition from occurring. If the program hangs, the counter will no longer be reloaded and the system will reset itself. A software reset is normally sufficient to free the software and let the system start again with what it was designed to do. The 'software watchdog' is activated by software, while the hardware watchdog is permanently activated by the hardware.

A/D converter. The built-in A/D converter has eight analogue inputs. It converts analogue values to 8-bit digital values by successive approximation. The analogue voltage to be digitized must not be higher than the processor's supply voltage, since this is used as the reference voltage.

D/A converter. The built-in D/A converter generates a 10-bit pulse-width modulated signal whose duty cycle is determined by software. This signal can be converted to a control voltage by an external RC filter, and the resulting voltage can be used to replace a potentiometer or an analogue control voltage. Some versions of the ST7 have a 12-bit converter.

Asynchronous serial communication. The Serial Communication Interface (SCI) provides the user with a flexible full-duplex serial connection. Data can be exchanged via this interface according to the standard serial data format.

Thanks to the built-in double baud rate generator, a wide range of baud rates is available. The transmitter and receiver work independently and can be set to different rates.

Serial Peripheral Interface (SPI). This synchronous serial interface is ideal for interconnecting master and slave devices. Systems with single as well as multiple masters can be constructed. This interface can be used for communication with I/O chips as well as with other processors. Advanced registers and interrupts allow transactions to be dealt with under software control, and also support user-defined protocols.

USB interface. The USB interface allows PCs and peripheral equipment, such as monitors, keyboards, multimedia devices and scanners, to be interconnected via a standardized interface. An important advantage of this interface is that it is 'hot pluggable', which means that peripheral devices can be connected or disconnected without having to switch off the power or reset the system. The USB interface implemented in the ST7 is suitable for the low-speed version of the USB. Data transfers take place via DMA. The USB function has its own integrated 3.3 V supply and a transceiver. The processor supports the Suspend and Resume commands.

I2C interface. The I2C bus allows serial data communication between components to take place using only a data line and a clock line. The I2C interface of the ST7 supports both multi-master and slave modes, and it can handle speeds (signal rates) up to 400 Hz. Standard events such as 'bus busy' and 'slave address', as well as error conditions, are automatically detected and stored in an associated register. An interrupt can optionally be triggered as a result of such events. The interface supports 7-bit and 10-bit addressing.

CAN bus interface. The CAN (Controller Area Network) protocol allows the user to take advantage of developments in Europe and other regions, in which this bus is being used more and more in cars and industrial settings. The CAN bus is an excellent choice for environments that are subject to strong interference signals.

The CAN interface that is integrated in the processor supports the '2.0 active' and '2.0B passive' CAN standards. It contains three 10-bit transmit/receive buffers and two 12-bit message acceptance filters. The baud rate is programmable, with a maximum value of 1 Mbit/s.

Getting started is easy

There are several development systems available for creating applications for the ST7 processors. They range from simple starter kits, consisting of a prototyping board with documentation and simple development software, to a very powerful C compiler with debugging tools. Existing development environments, such as the Actum Realizer (a package that allows software to be developed by describing its structure graphically), have also been adapted to support the ST7 processor. For beginners, the ST7 Development Kit from ST Microelectronics is recommended. This package contains all relevant information, with most of the documentation located on a CD-ROM along with the software. The accompanying hardware makes it easy to develop relatively simple applications.

Conclusion

If this article has aroused your curiosity, it is recommended that you visit the web site *http://www.st.com*. ST Microelectronics has put a lot of information about the ST processors and associated development systems at this address. Application notes, data sheets and other interesting information can be downloaded free of charge from this site.
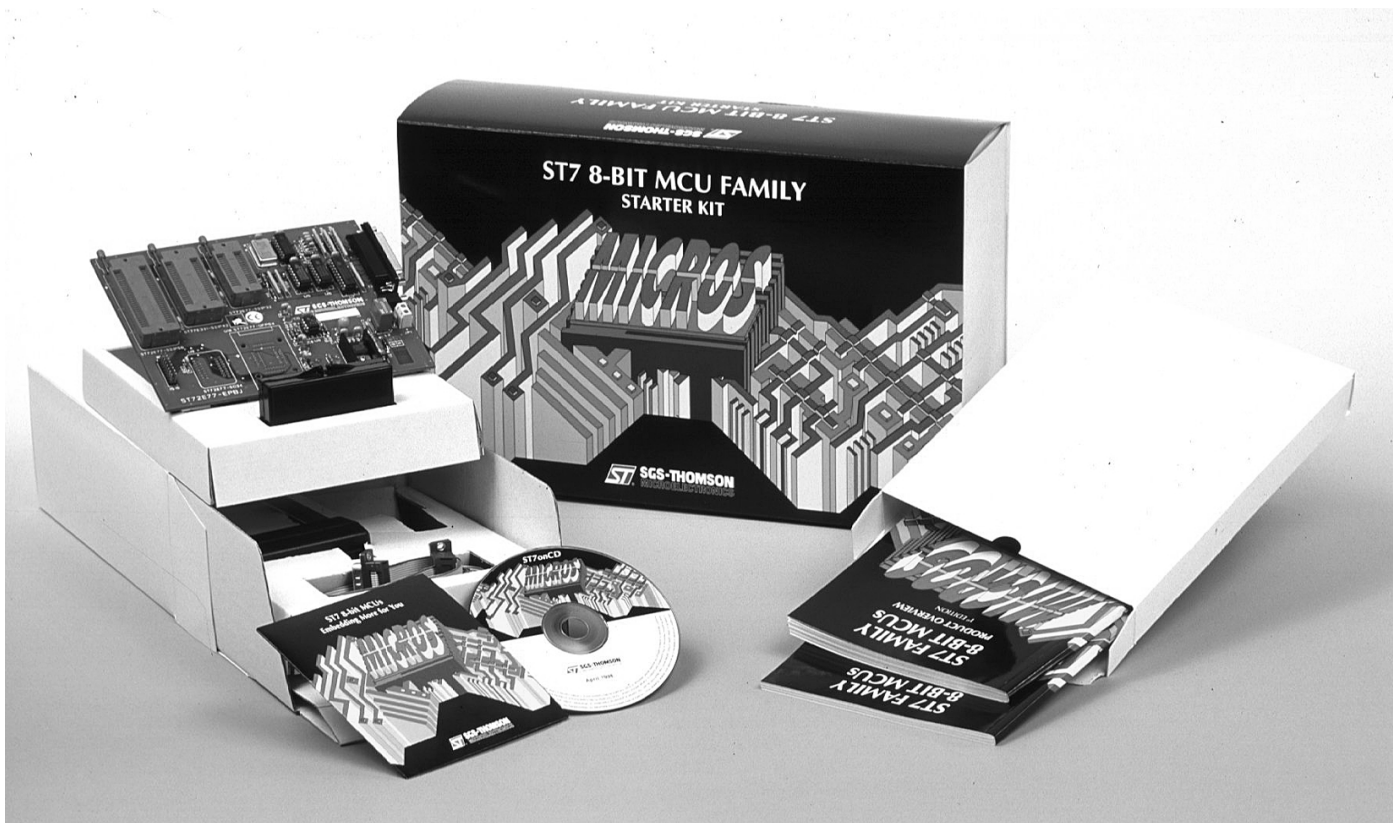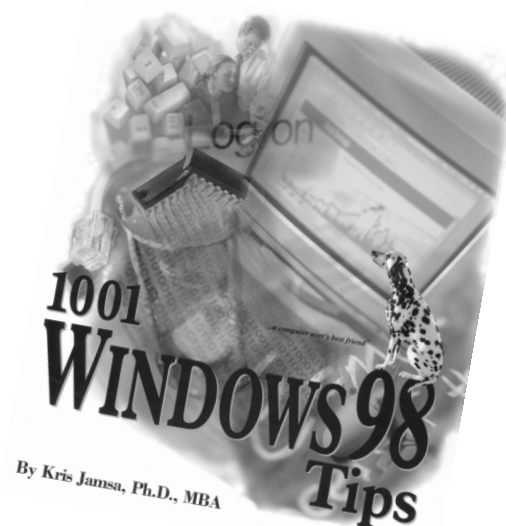
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Figure 4. The ST7 Development Kit from ST Microelectronics is a good starting point for getting acquainted with these processors.

This month we present a new selection of tips, tricks, hints and kinks for all users of the Windows 98 operating system. Take your pick!

By Chris Jamsa, Ph. D., MBA

Windows 98 Tips & Tricks

NFS is the Unix Network File System

Windows 98 uses the FAT32 file system to store files on your disk. In a similar way, the Unix operating system uses the NFS or Network File System. In general, FAT32 and NFS define how the operating systems store files on disk. Windows 98 and Unix store files differently. As a result, using the FAT32 file system, Windows 98 could
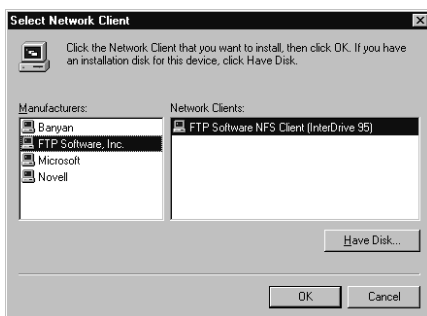
Figure 7. To access files on a Unix disk, you may need to install NFS client software on your system.

not read a Unix disk. Likewise, using NFS, Unix could not read a Windows 98 disk. If you connect to a Unix computer that resides on your network or to which you dial in, you may need to install software support within Windows 98 that lets your system access files stored on an NFS disk. In most cases, your network administrator will install and configure the NFS software for you. **Figure 7**, for example, shows the Select Network Client software dialog box that shows FTP Software, Inc.'s NFS client.

For Optimal Performance, Ensure that Your System is Using 32-Bit Device Drivers

To improve performance, Windows 98

makes extensive use of protected-mode 32-bit device drivers. Unfortunately, depending on your hardware (or age of your hardware), Windows 98 may have to use real-mode 16-bit drivers for specific hardware devices. Unfortunately, real-mode device drivers are much slower than their 32-bit protected-mode counterparts, which will slow down your system performance. As a rule, you should use 32-bit drivers whenever possible. By using the Windows 98 Update Wizard, you can locate and install new device drivers much easier than users could in the past. To determine if a device driver is using a real-mode 16-bit or a virtual 32-bit mode device driver, perform these steps:

1. Click your mouse on the Start menu and then select the Settings menu Control Panel option. Windows 98, in turn, will open the Control Panel window.
2. Within the Control Panel window, double-click your mouse on the System icon. Windows 98 will display the System Properties dialog box.
3. Within the System Properties dialog box, click your mouse on the Performance tab. Windows 98, in turn, will display the Performance sheet.
4. Within the Performance sheet, look for the message "Your system is configured for optimal performance." If you find the message, your system is using all available virtual 32-bit device drivers. If you see a different message, click your mouse on the Device Manager tab. Windows 98, in turn, will display the Device Manager sheet. Within the Device Manager, examine the various devices and check if they are using 32-bit drivers. If you find an older driver,

use the Windows Update feature to locate and install a newer device driver.

Optimizing the Windows 98 File System for your Hard Disk

To improve your system performance, Windows 98 takes advantage of disk caches which reside in your PC's fast electronic RAM. Windows 98 can perform read and write caching. As your learned, when Windows 98 performs write-behind caching, Windows 98 will temporarily place the information that your programs write to disk into its disk cache. Later, when Windows 98 has a free moment, it will write the cache's contents to disk. Because your program does not have to wait for the slow disk-write operation to complete, your program's performance improves. Unfortunately, if the short interval of time between when Windows 98 places the information into the cache and when Windows 98 records the information on your disk, your system experiences a power outage or disk error, you may lose the information in cache. Worse yet, your program thinks the information was correctly written to disk. Because of this potential data loss, users often turn off write-behind caching.

Depending on your willingness to trade off risk of data loss for improved performance, you may want to let Windows 98 perform write-behind caching. If you are using write-behind caching and you are experiencing disk errors on a removable drive, you can perform these steps to disable write-behind caching for that drive:

1. Select the Start menu Settings options. Windows 98, in turn, will dis-

play the Settings submenu.
2. Within the Settings submenu, click your mouse on the Control Panel option. Windows 98 will open the Control Panel window.
3. Within the Control Panel, double-click your mouse on the System icon. Windows 98, in turn, will display the System Properties dialog box.
4. Within the System Properties dialog box, click your mouse on the Performance tab. Windows 98, in turn, will display the Performance sheet.
5. Within the Performance sheet, click your mouse on the File System button. Windows 98, in turn, will display the File System Properties dialog box.
6. Within the File System Properties dialog box, click your mouse on the Removable Disk tab. Windows 98, will display the Removable Disk, sheet as shown in **Figure 8**.
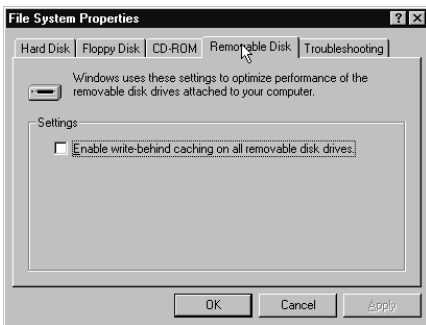

Figure 8. The File System Properties dialog box Removable Disk sheet.

7. Within the Removable Disk sheet, remove the check mark from the Enable write-behind caching on all removable drives checkbox and then click your mouse on the OK button.
8. Within the System Properties dialog box, click your mouse on the Close button to close the dialog box.

Changing a Disk Drive Letter Assignment

When you install software on your system, Windows 98 will often create shortcuts and menu options that correspond to the software's location on your disks. If you later remove a hard drive from your system, Windows 98 may reassign your disk drive letters, which causes your programs not to run from the menu option or shortcut. In such cases, you may be able to change the disk drive's letter assignment (which may you may find convenient for changing your CD-ROM drive letter) by performing these steps:
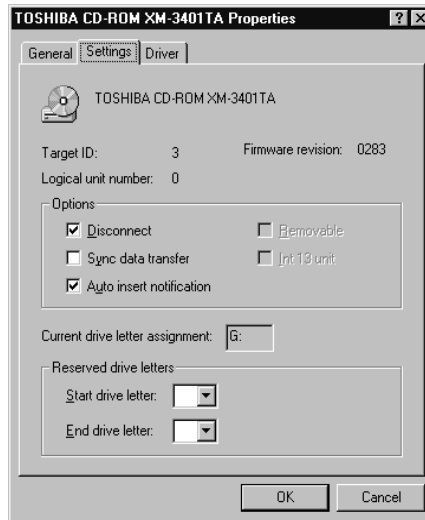

Figure 9. The Properties dialog box Settings sheet.

1. Select the Start menu Settings options. Windows 98, in turn, will display the Settings submenu.
2. Within the Settings submenu, click your mouse on the Control Panel option. Windows 98 will open the Control Panel window.
3. Within the Control Panel, double-click your mouse on the System icon. Windows 98, in turn, will display the System Properties dialog box.
4. Within the System Properties dialog box, click your mouse on the Device Manager tab. Windows 98, in turn, will display the Device Manager sheet.
5. Within the Device Manager sheet, expand the drive list, and then double-click your mouse on the disk drive. Windows 98 will display the drive's Properties dialog box.
6. Within the Properties dialog box, click your mouse on the Settings tab. Windows 98 will display the Settings sheet, as shown in **Figure 9**.
7. Within the Settings sheet, use the Reserved drive letters field to select the drive letter you desire. Then, click your mouse on the OK button.
8. Within the System Properties dialog box, click your mouse on the OK button.

Forcing Windows 98 to use a Permanent Swap File

When you run several programs at the same time, Windows 98 may, depending on the number and the size of your programs, run out of RAM. In such cases, Windows 98 will use a *swap file* into which it will move one program out of memory to disk, in order to make room for a different program in memory. The more programs you run

simultaneously, the more Windows 98 may have to swap programs between RAM and your swap file on disk. As a rule, the more RAM your computer has, the less Windows 98 will have to swap to and from the disk.
Usually, Windows 98 will reserve space on your hard disk for the swap file. In previous versions of Windows, you had to configure your Windows swap file settings yourself. In contrast, Windows 98 uses a dynamic swap file that it can grow or shrink as its needs require.
Although Windows 98 configures its own swap file settings, you can take control and configure your own swap file, as your needs require (most users won't have a need to configure the swap-file settings). To force a permanent swap-file size within Windows 98, perform these steps:
1. Click your mouse on the Start menu Settings option and choose Control Panel. Windows 98 will open the Control Panel window.
2. Within the Control Panel window, double-click your mouse on the System icon. Windows 98 will display the System Properties sheet.
3. Within the System Properties sheet, click your mouse on the Performance tab. Windows 98 will display the Performance sheet.
   Within the Performance sheet, click your mouse on the Virtual Memory button. Windows 98 will display the Virtual Memory dialog box, as shown in **Figure 10**.
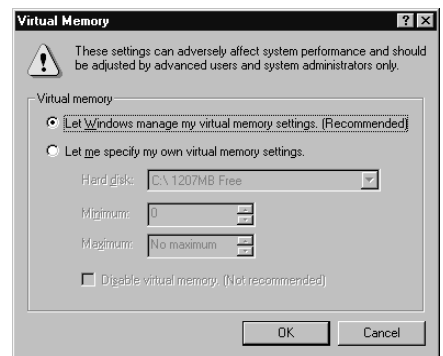

Figure 10. The Virtual Memory dialog box.

5. Within the Virtual Memory dialog box, click your mouse on the Let me specify my own virtual settings and then enter a minimum and maximum size value for your swap file. Next, click your mouse on the OK button.
6. To put your changes into effect, you must restart your system.

(992028-II)

For recording you own CDs or editing audio files on the computer, it can be a good idea to have an S/PDIF input and output available, for example on a sound card. Unfortunately, the capabilities of such interfaces on inexpensive sound cards are fairly limited, which you should take into account before making a purchase. First of all, though, what do you need to know about such connections, and are they really necessary?

S/PDIF connections for the PC
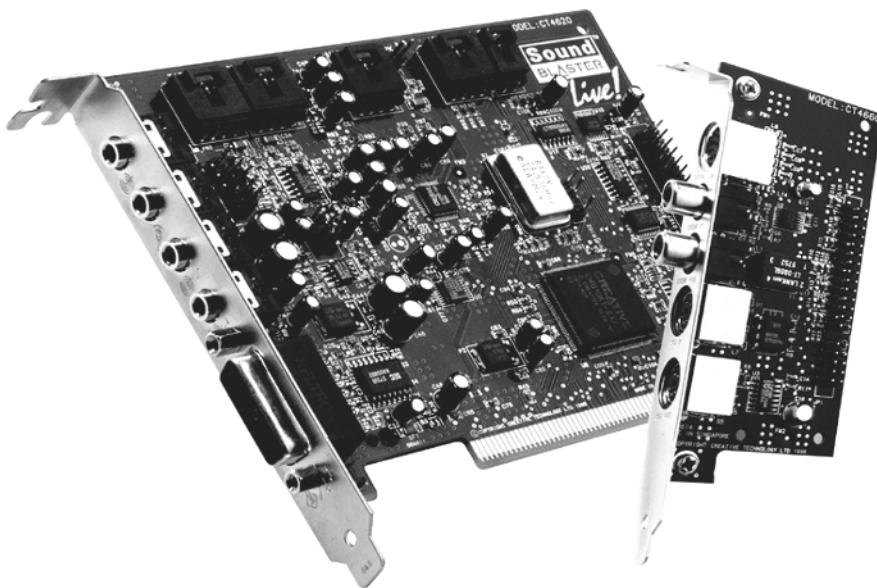
useful, or an unnecessary luxury?

Figure 1. The Soundblaster Live! has two connector brackets. The digital connectors are located on the second bracket (photo: Creative).

lowly but surely, affordable sound cards with an input and/or output for S/PDIF signals as a standard feature are becoming available in increasing numbers. The S/PDIF interface take the form of coaxial (Cinch) or optical-fibre connectors. With this interface, it is possible to transfer digital audio signals from the computer to a recorder (or the other way around) without having to pass through an intermediate analogue stage.

Only 48 kHz?

One of the currently most popular sound cards, the Soundblaster Live!, has a digital input and a digital output as standard features. However, before you start using them, you should under-

stand what happens to the digital signal inside the card.

The heart of this sound card is a DSP chip that performs all digital processing at a fixed sampling rate of 48 kHz. All digital input data are converted to this sampling rate by means of a sampling rate converter. The advantage of this is that all types of processing, such as adjusting the volume via the Windows mixing panel, can be performed by the DSP without the need for any extra calculations to convert between different sampling rates.

The digital output signal also works with a fixed sampling rate, which is the same 48 kHz. This is excellent for recording data from the PC on a DAT recorder, since this is the standard sampling rate for such recorders. However, if you feed the digital signal from a CD player to the sound card, the sampling rate converter will change the signal from 44.1 kHz to 48 kHz. Although this process is practically loss-free, it still amounts to an extra conversion step. And if you want to output the digital signal from the sound card to a device that uses a sampling rate of 44.1 kHz, you are simply out of luck, since that is not possible with this card.

Still, these restrictions are not as bad as they may seem at first glance, as long as you first carefully think through what you want to do with the S/PDIF connection.

Applications

For most hobby applications, digital connections on the sound card will rarely be necessary. Anyone who wants to make compilations of music from CDs can simply read in the tracks digitally using his or her CD-ROM drive or CD burner, and then put the modified selection on a new CD. Digital inputs and outputs on the sound card are thus unnecessary for this task.

However, anyone who wants play back data from a MiniDisc or DAT cassette to the computer does in fact need a sound card with a digital input. If in addition the data are to be transferred back to a DAT or MiniDisc recorder after being edited on the computer, a digital output is also necessary. As already noted, the standard sampling rate for DAT is 48 kHz. In a MiniDisc recorder, the data are always first passed through a sampling rate converter internal to the device and then compressed. In this case we do not have to be concerned about the sampling rate used by the sound card — 48 kHz is just fine!

On the other hand, if you want to copy your own music from a DAT to a CD via the computer, then you will have to somehow arrange for the data to be converted from 48 to 44.1 kHz. This can be done using software on the PC.

Some sound cards have only a digital output, with no digital input. Such a card is good for fanatical game players (among others) who want to feed the sound signal to the digital input of their stereo system in order to get the best possible results.

Nonetheless, in most cases the best advice is to take a good second look at your specific application. Frequently, you will conclude that there is no specific requirement for a digital input or output.

Note well that this recommendation applies to typical hobby applications. For professional applications, the requirements are naturally quite different.

Other cards

The Soundblaster Live! is fully adequate for the vast majority of PC users, for 99% of all applications. Another general-purpose sound card that has digital inputs and outputs is the Audiowerk2 card from Emagic, at around £200.

However, if you do a lot of work with digital audio signals, it's better to use a card that is specially designed for such use. A well-known and affordable example is the Digi32 card from RME, which has both RCA and Toslink connectors and supports several different
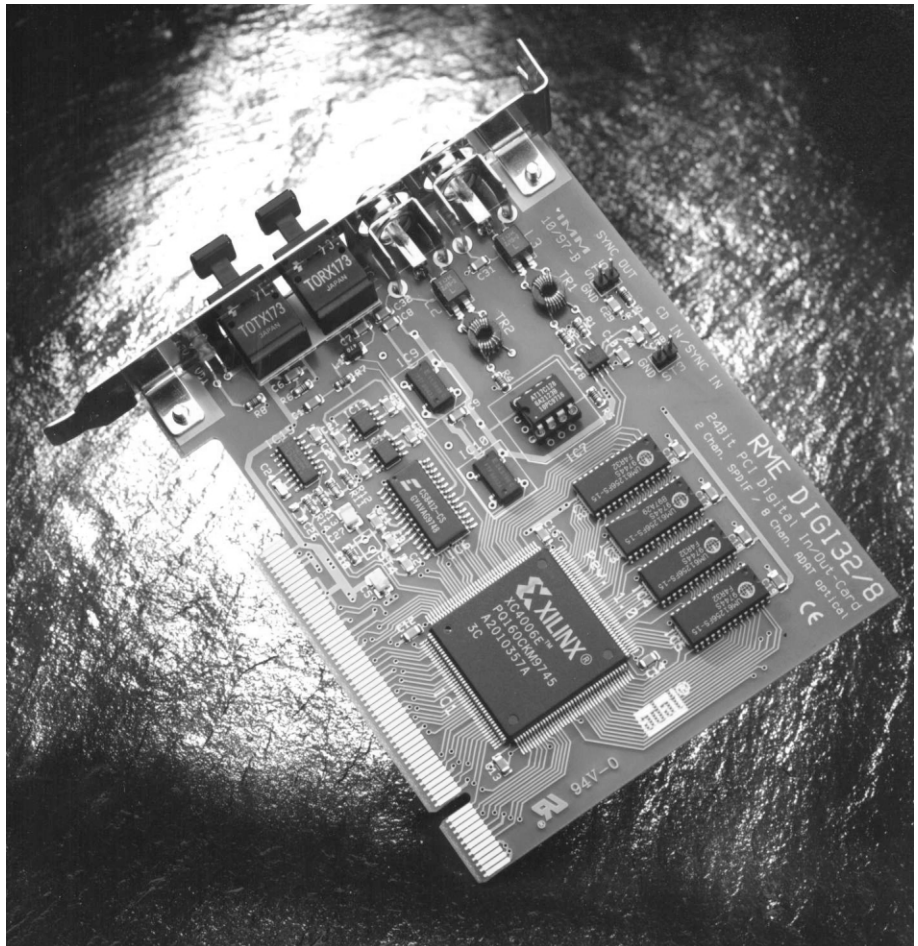

Figure 2. If you work a lot with digital audio signals, you are better off to buy a card that is specially designed for such signals, such as the Digi32 from the German firm RME (photo: RME).

sampling rates. The price of this card is around £240.

For the true audio enthusiast who does a lot of audio editing with the computer, it is convenient to have all the audio connectors in a more accessible location than at the rear of the computer. The Guillemot Maxi Studio Isis (currently around £300) is in this case an outstanding solution, since it has all connectors located in a separate box,

along with the converters.

If you want to be well prepared for what's coming next, it's a good idea to buy a card that supports 96 kHz sampling with 24-bit resolution. This is the new standard for DVD audio. RME offers cards with this capability, but there are others available as well, such as the Terratec EWS 88 MT card (at around £400).

(992030-1)

Figure 3. This card from Guillemot provides a handy setup for the true audio hobbyist, since it has all connectors and the converters located in a separate enclosure (photo: Guillemot).

One of the most frequently asked questions on the Internet site for the 80C32-BASIC control computer is, 'Can it burn EPROMs?' In fact, it can, and this article explains how.

Design by H. J. Böhling

EEPROMs and the BASIC control computer

FAQs for the BASIC computer

The 8052AH-BASIC V1.1 processor uses port 0 for programming EPROMs. Since this port is also used for the external data and address bus, using it to burn EPROMs is only possible because the BASIC interpreter is stored in the internal memory of the 8052. It thus does not need the external data/address bus while it is programming an EPROM. Unfortunately, an 80C32 processor, with MCS-51 BASIC located externally, cannot burn EPROMS or EEPROMS! If you need to do this, you will have to have either the original 8052AH processor or an 87C52 processor (with 8 kB of internal EPROM) that has been programmed with the BASIC interpreter. Since the first of these controllers is no longer manufactured by Intel, it is likely to be difficult to obtain. The second controller, manufactured by Philips, is available and even has a lower current consumption, but it must be programmed with MCS-51 BASIC before it can be used. If you are not able to do this yourself, you can obtain a preprogrammed controller from Elektor Readers Services.

In addition, you will need some hardware that switches the programming voltage to the EPROM and controls the low-address register during the programming process. Both of these functions are found on the *Elektor Electronics* 80C32 BASIC Computer, for example (see the Elektor book *Short Course 8051/8032 Microcontrollers and Assembler*, which is available along with the circuit board 910042-1 through our Readers Services).

On the other hand, if you can do without burning EPROMs and can limit yourself to 8 kB EEPROMS, it's also possible to work with the 80C32-BASIC control computer. This still requires a processor with an internal BASIC interpreter, plus a number of modifications to the hardware and the circuit board (980002-1), as follows:

✓ Remove the 80C32 CPU (IC1) from its socket.
✓ On the top side of the circuit board, cut the track at pin 31 of the CPU socket (IC1).
✓ Connect a 47 kΩ resistor (R2) between pin 31 of the CPU socket and pin 40 of the CPU socket (+ 5 V).
✓ On the bottom side of the circuit board, cut the track that connects pin 30 of the CPU (ALE/P) to pin 11 of the address register (IC2). Solder a 1N4148 diode (D1) across the break, with the cathode on the CPU side and the anode on the IC2 side.
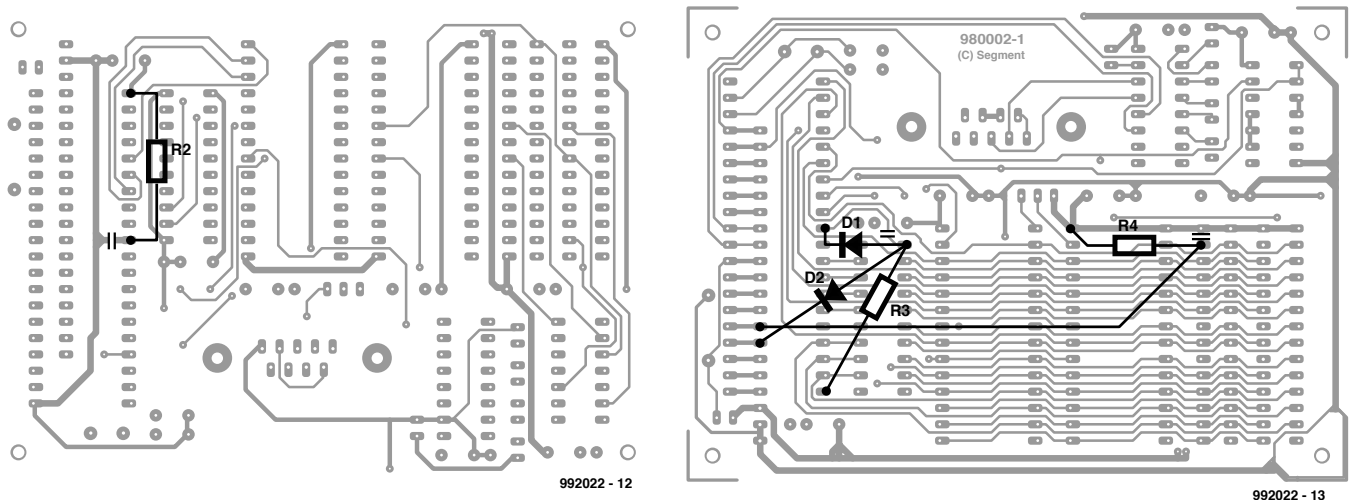✓ Connect a 47 kΩ resistor (R3)

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Figure 1. Modifications to the top and bottom sides of the printed circuit board.

With the present prices of CD burners and CD-R discs, making your own audio CDs with your own choice of music is an attractive possibility. You can for example make a compilation of tunes from existing CDs, or you can copy your favourite tracks from LPs onto a CD. In the latter case, the sound quality can be improved in numerous ways with the help of special software. In this article, we provide some general guidelines and technical details for those of you that want to start making your own audio CDs.

make your own
music CDs

Digitizing LPs and making your own compilations

clean up the audio files, you will have to resort to a computer with a CD burner.

Currently, one of the CD-recording programs Easy CD Creator (from Adaptec) or Win On CD (from Cequadrat) is most often supplied with the CD burner. We can use a few screen shots from the latter program to illustrate what you have to pay attention to when making an audio CD.

Copying CDs

The easiest task is simply making a 1:1 copy of an audio CD.

Before you start copying a CD, the CD reader must first be tested. This is because the error correction for audio CDs is much less extensive than for CD-ROMs. As a result, modern 32 or 40-speed CD drives can typically read audio CDs at a maximum speed of 4 or 8 times the normal rate. Most CD recording programs have a test option to determine the maximum rate at which the drive can read the audio data without errors. Note however that the reliability of the test results can vary, depending on the testing method used. It can thus be better to use a separate 'ripper' program (as it's called in trade jargon), which is specially designed to read the data from an audio CD. Some examples of such programs are Digital Audio Copy (*www.windac.de*) and Easy CD-DA Extractor

Although nowadays everybody in the computer world is talking about MP3 audio, the normal audio CD is still the most important recording medium for the true audio fan, as far as quality is concerned. The data on an audio CD are not compressed and are recorded

at a sampling rate of 44.1 kHz with 16-bit resolution.

To put together an audio CD, you can use a special audio CD recorder, such as the one made by Philips. However, the features of such a recorder are rather limited. If you want to edit or

(*www.polkasoft.com/cdda/*).

Win On CD also provides jitter correction for reading in audio data. With most modern CD drives, the quality of the read data is good enough that jitter correction is normally not necessary, but with older-model drives it can be essential for obtaining good results.

After you have determined the best data transfer rate from the CD-ROM drive, you can copy the desired tracks. With a 1:1 copy, you can either copy the tracks directly to the CD burner or you can first make an 'image' on the hard disk. An image is a replica of the content of the CD. It is usually written to the hard disk as a wav file, and then read out and converted to the normal audio CD format when the new CD is recorded. Keep in mind that you will need around 700 MB of free space on the hard disk for an image of an audio CD.

When putting together a compilation CD, you can first copy the individual tracks to the hard disk and then record the collected tracks on a CD in the desired sequence.

That's basically it, but there are a few things that you have to watch out for.

Track at once or disc at once?

Normally, in track-at-once mode, a 2-second gap is inserted between successive tracks by the CD burner. This is not exactly what normal audio CD players want to see. It can often completely disrupt their operation, since the laser beam is switched off while the gap is recorded.

When copying an entire CD, the CD burner often automatically transfers all the tracks and records them 'chained together'. If you are recording a number of tracks that you have collected yourself, you must take care that the option 'disc at once' is selected in the CD burner program. In this case pauses will still be inserted between the individual tunes, but the laser beam will not be switched off between tracks.

Pre-emphasis

You may not be particulary familiar with this term. In the early days of audio CDs, a form of pre-emphasis was sometimes used to improve the dynamic range for high frequencies. With pre-emphasis, all frequencies above 3kHz are recorded a few dBs louder on the CD. This setting is detected by the CD player during playback, so that a de-emphasis filter can be switched in to restore the original levels. Nowadays this technique is totally unnecessary and is not used. However, if you are copying tracks from old CDs you have
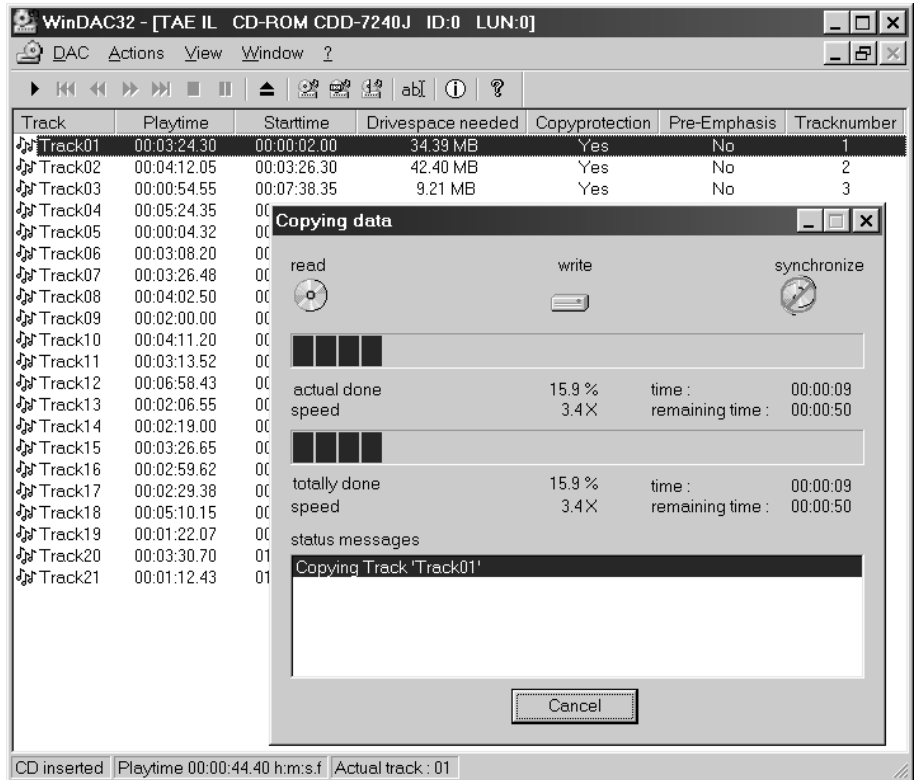
Figure 1. A 'ripper', such as Digital Audio Copy, can be used to transfer exact 1:1 copies of CD audio tracks to the hard disk.

to check whether they were recorded with pre-emphasis. The burner software will normally not indicate this, but you can always check it by simply inserting the CD into a normal audio CD player, which usually will indicate whether pre-emphasis is present. Some CD copying programs, such as Digital Audio Copy, also indicate whether the pre-emphasis bit is set in the original recording.
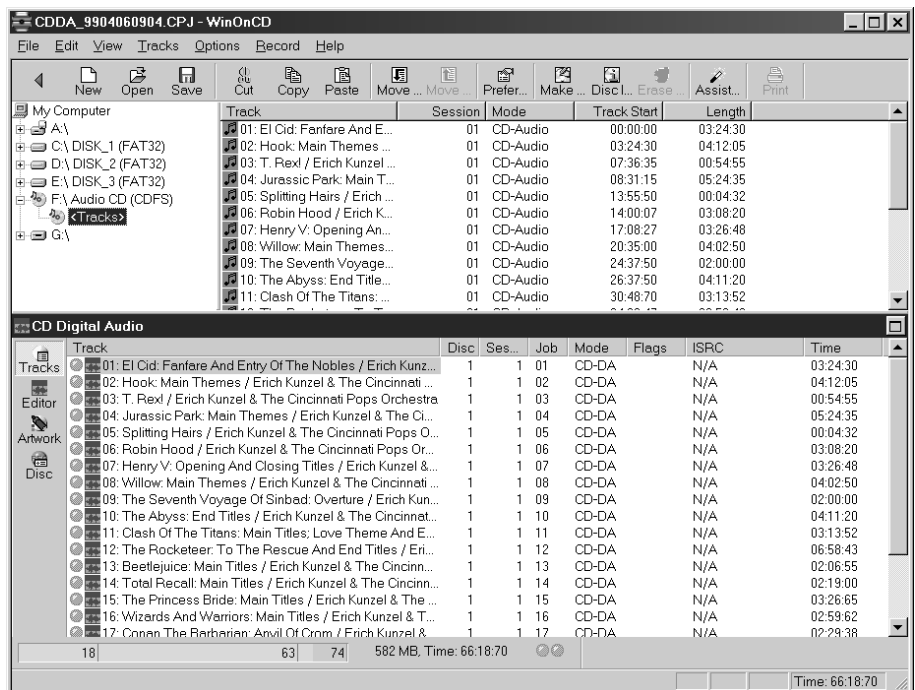
Figure 2. Most CD burning software is designed with two main windows, so that tracks can simply be dragged from the one window (the 'explorer' window) to the other window (the CD-R that is to be burned).
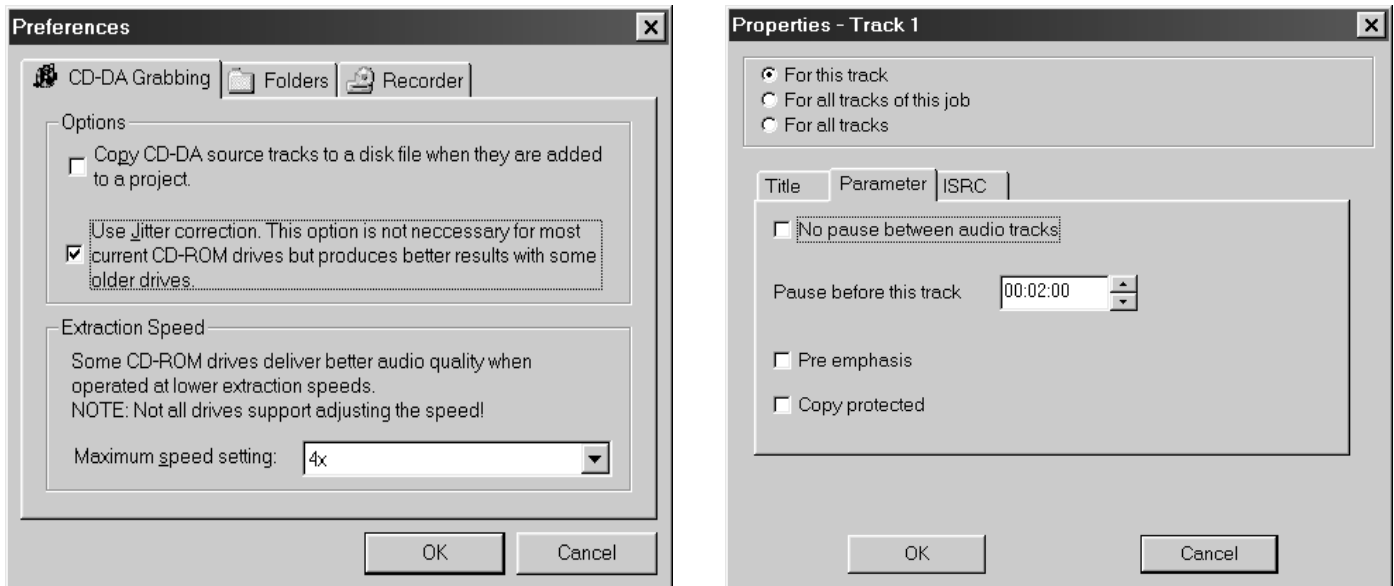
Figure 3. Win On CD 3.6 provides (among other features) jitter correction for reading in audio track data. Parameters such as the pre-emphasis bit and copy protection can be individually set for the CD to be recorded.

An enabled pre-emphasis bit is not automatically copied from the audio CD by most CD-recording programs. With Win On CD you can set it manually, which is a useful feature if you occasionally need it.

Copy protection
A number of copy-prohibit bits prevent an audio CD from being copied digitally more than once (that is, they prevent any further digital copying of a digital copy of a CD). The settings of these bits are also not necessarily copied intact by the recording software. Some programs allow copy protection to be manually enabled or disabled. If it is disabled, any number of secondary copies of the CD can be made (so that you can make a copy of the CD, and then a copy of the copy, and so on). This can be useful for recordings of your own material.

Analogue recordings

When you copy digitally recorded music you will seldom if ever need to do anything to the content, but if you make digital recordings from analogue sources, such as phonographs and cassette players, some editing of the music may well be useful or even essential. Who hasn't dreamed of being able to listen to some numbers from a rare LP (that has never been released on CD) without the usual noise? Digital technology provides quite a few possibilities in this regard.
What do you need to be able to make your own recordings?

First of all, of course, you need a computer with a CD burner. In most cases, the computer will be running Windows software.
In addition, you will naturally need a good-quality sound card. What does this mean? Almost all modern PCI sound cards provide good to outstanding sound quality, with a large dynamic range and little distortion. For recording material from phonograph records and cassettes, a sound card that costs on the order of £35 to £70 should thus be satisfactory. A cassette deck can be connected directly to the line inputs of the sound card, but a separate MD/MC phono preamplifier is needed for a phonograph (the specific type depends on the type of cartridge used). The best results are obtained when the preamplifier provides line-level outputs that can be directly connected to the line inputs of the sound card (since the microphone input of a sound card delivers inferior sound quality and is also only monophonic). Elsewhere in this issue you will find a design for a high-quality stand-alone MD/MC preamplifier, which is perfect for this application.
Finally, you naturally need software to read in the tracks from the records or cassettes and write the data to disk in wav format, following which the tracks can be cleaned up (if necessary) and used to 'burn' an audio CD.

Reading in audio signals
In principle, you can use the sound recorder function of Windows to read in analogue signals, but this is a rather primitive approach. It is better to use a

program with more features, such as the well-known programs Cool Edit (*www.syntrillium.com*), Goldwave (*www.goldwave.com*) or Wavelab (*www.steinberg.net*). With these programs, you can more conveniently set parameters such as the sampling rate and the recording level. Many of these programs also have built-in VU meters, so that you can closely monitor the recording level. Be sure to carefully select the recording level. Overdriving is bad for the sound quality, but a signal recorded at too low a level contains too much noise. This is actually the same as for making recordings with a cassette recorder, but you should keep in mind that the audible effects of overdriving are much more evident in the digital domain than with analogue recordings.
It is important to select the correct input (line) in the Windows audio mixer before you start, and to set all other inputs to zero or switch them off. Pay particular attention to the microphone input, since it will add a lot of unnecessary noise if it is not disabled!
A program such as Cool Edit also allows the lead-in and trailer of a piece of music to be removed, so that the resulting wav file contains only exactly what is necessary.
Also make sure that everything is read in with a sampling rate of 44.1 kHz if you plan to record it on a CD that is to be played back using an ordinary audio CD player. Although good audio editing programs allow the data to be resampled (for example, to change the sampling rate of a digitized audio file from 48 kHz to 44.1 kHz), it is naturally

better to avoid any unnecessary intermediate processing.

The indirect approach

As an alternative to digitizing the analogue signal using the sound card, you can first record the desired pieces of music using a DAT or MiniDisc recorder. This is often a more convenient and reliable approach, and the D/A converters in such recorders provide outstanding sound quality. After you have recorded the data, you can simply transfer the digital signal to the PC.

Of course, this approach makes other demands on the necessary hardware and software. First of all, you will need a card with a digital (S/PDIF) input in order to read the digital signals into the computer. This could for example be a Soundblaster Live card or a Terratec EWS64, or a special S/PDIF card such as the Digi32 from the German firm RME. Next, a translation step is often needed in the computer. A normal DAT recorder always works at a 48 kHz sampling rate. After the data that have been sampled at this rate have been read in, they must be resampled to 44.1 kHz with the help of a program such as Cool Edit. This problem is not present with a MiniDisc recorder, since it has an internal converter that always provides an output signal at 44.1 kHz. The disadvantage of using a MiniDisc recorder is that the data are compressed internally and possibly also resampled.

Editing

Various programs are available for editing wav files. For example, Sound Forge (*www.sfoundry.com*) and Steinberg Clean (*www.steinberg.net*) offer a lot of options. Nowadays, even CD recording programs such as Easy CD Creator de Luxe and Win On CD 3.6 include a number of editing modules that can for example remove noise and clicks.

Although audio enthusiasts have been very sceptical about such editing in the past, it must be admitted that the capabilities of modern editing software are amazingly good. It is truly possible to very accurately locate and eliminate noise and crackles at the digital level, without affecting the audio signal.

Figure 4. A good-quality sound card is essential if you want make recordings from records or cassettes.
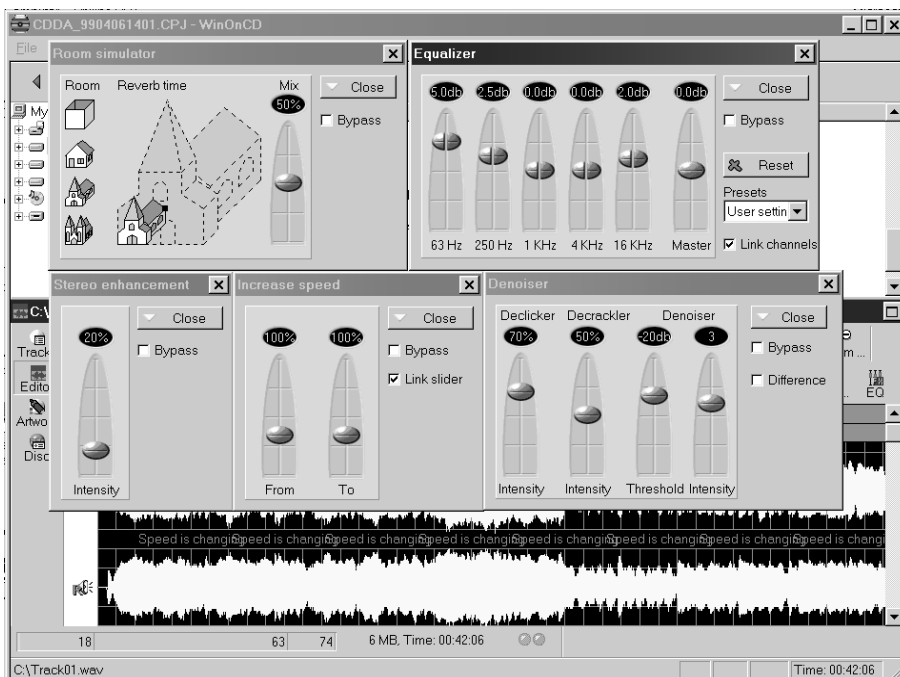

Figure 5. A program such as Cool Edit is very well suited to making recordings from phonograph records, and in addition it provides a large number of editing options, such as resampling.

CD databases

Making a complete list of the titles of all the tracks you have recorded on your CD, and putting the titles on a neat label or inlay for the CD case, can easily be a fairly time-consuming activity. Fortunately, there are also a number of interesting aids to help with this task. Everyone with access to the Internet can browse free of charge in the CDDB (CD Database), which is located at *www.cddb.com*. The track titles of several hundred thousand audio CDs are collected in this data base, and the number is increasing daily, since every user can make his or her own contributions to the data base. You can 'manually' look up the title of a CD in the CDDB and then copy the relevant information to your own program for further use, but there is also software available that makes automatic use of the CDDB. For example, there are programs for assembling your own CD database, CD players, CD jukeboxes and CD recording software that utilize the CDDB. Whenever the program or device needs the track information, it just makes a connection to the CDDB and downloads the necessary information.

In addition, there are programs available that have their own audio databases with track information. For example, the latest version of Win On CD (3.6 Power Edition) has its own database, with track titles from a number of well-known audio CDs. The exact number is not specified, but probably it will be possible to update this database as well via the Internet in the near future.
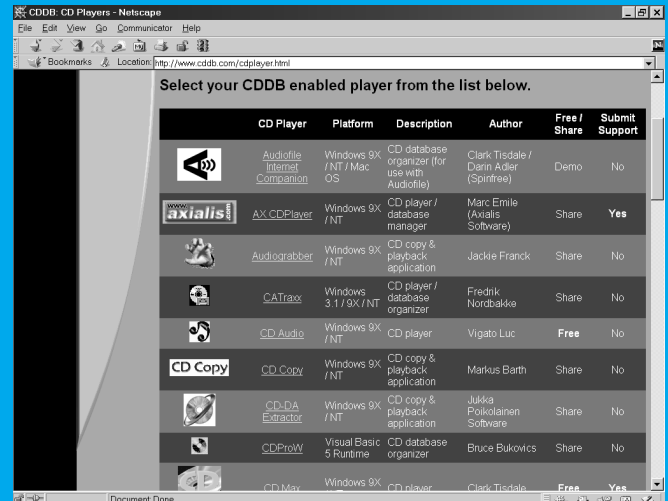
Figure 6. Win On CD includes various editing options, ranging from eliminating clicks to adding reverberation.

Exactly these sorts of defects plague LPs (especially older examples).

If we take Win On CD 3.6 again as an example, we see that there is a built-in audio editor that includes a 5-band graphic equalizer, a declicker for removing clicks, a decrackler for eliminating short ticks such as crackling or rattling sounds, and a 'denoiser' for removing broadband noise from a recording. There are also a number of optional 'effects', such as stereo enhancement, adjusting the playing speed and adding a certain amount of reverberation.

With Easy CD Creator de Luxe, a number of similar features are provided by the companion Spin Doctor program.

Burning the CD

After you have recorded and edited a number of wav files, you can put them all together onto a CD-R. Do not use a re-recordable CD (CD-RW), since the majority of audio CD players are absolutely unable to read such a CD. With most CD burning programs, you can simply drag the various wav files from the 'explorer' window to the CD-R window in the order that you want to have them placed on the CD. To stay as compatible as possible with normal audio CD players, you should not put multiple sessions on an audio CD and you should select the 'disc at once' option, as previously mentioned. This guarantees that the resulting audio CD can be played back by all CD drives without any problems. Some programs allow you to specify the length of the pause between successive tracks, but usually this is fixed at 2 seconds.

A normal audio CD player will always recognize the various track numbers of a DIY CD. Sometimes you also have the option of placing index markers before burning the CD. This can be handy for classical music, which cannot be easily split into a number of short pieces. Index markers provide a simple method for jumping to the start of specific passages or movements.

The final touches

As you can see, it's not all that difficult if you just have the right kit: a computer with a sound card, a CD burner, a turntable with a separate preamplifier or a cassette recorder, and of course the necessary software. All that's then left is finishing off your product. If you have the time, you can make an attractive inlay and label for your self-made CD. There are currently a number of commercial packages available for generating CD labels (from HP and Traxdata, among others).

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Copyright

If you make your own music CDs, you must give due consideration to copyright aspects. It is perfectly OK to make a copy of music from your private collection of CDs and LPs for your own use, but you are not allowed to give or sell such copies to others who do not already own the original(s). After all, the artists have to live from something as well, and they suffer the most from illegal copying.

RF marker generator

calibrate receivers, receiver dials and signal generators

Anyone involved in the construction of radio receivers and associated equipment will have felt the need for some means of producing an accurately calibrated dial. The ability to confidently set the tuning to a particular frequency is often half the battle when searching for a weak transmission, and greatly increases the pleasure of using a receiver of any kind.

The calibrator described here relies on a quartz crystal to determine and hold constant the frequency of the oscillator providing the marker signals.

Design by Raymond Haigh
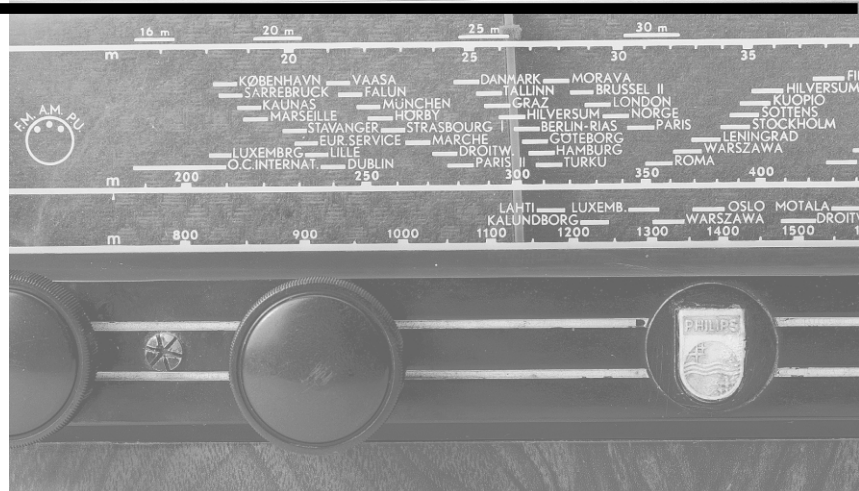
Early instruments of this kind often incorporated two crystals, one cut to resonate at 1 MHz, the other at 100 kHz, and the fundamentals and harmonics of these frequencies were used to calibrate receivers and test gear up to, and beyond, 30 MHz.

The development of the integrated circuit, in particular the production of binary and decimal counter ICs, made it possible to use only one crystal, cut to resonate at a higher frequency (often 2 or 4 MHz), and to divide the frequency down in order to provide markers for long and medium waves, and for the fine sub-division of the dial.

When attempting to calibrate the dial of a short-wave receiver using only a crystal standard, difficulty can be encountered in identifying the particular harmonic of a 1 MHz fundamental. If a 4 MHz crystal is used the position is much easier, but confusion can still arise as calibration progresses into the upper reaches of the HF spectrum.

In order to avoid this, the unit described here has been designed around an 8 MHz crystal. With harmonics at 16, 24 and 32 MHz, virtually unmistakable markers are produced throughout the HF bands, and it is a

comparatively simple matter to calibrate up or down from these points by means of lower frequency signals produced by the division of the crystal fundamental.

Most crystal markers make no provision for the modulation of the signal. High performance receivers incorporating a BFO (beat frequency oscillator) can produce an audible tone from an unmodulated carrier, but with domestic superhets there will be no clear audible indication that the marker signal has been tuned-in. Ironically, it is often the listeners who are trying to get the most out of simpler receivers who have the greatest need for some means of improving tuning accuracy. Accordingly, provision is made for the optional modulation of the marker signals.

THE CIRCUIT

The full circuit of the unit is given in **Figure 1**. It is based on low-cost, CMOS (complimentary metal oxide semiconductor) logic gate ICs. An inexpensive 8 MHz crystal determines the basic frequency of oscillation with a high degree of accuracy and stability.

IC5 contains four two-input NOR gates. Strapping the inputs of each

gate together produces four inverting amplifier stages. One of these, IC5a, is used as the RF oscillator, with the frequency of oscillation determined by quartz crystal X1. In this circuit, the crystal presents an inductive reactance at the operating frequency, and it resonates with loading capacitors, C1 and C2. C2 is adjustable, and the frequency of oscillation can be set, by reference to an external standard, at precisely 8 MHz. DC negative feedback, provided by R1, stabilises the operating conditions of the stage, and IC5b buffers the output.

IC5c and IC5d are connected as an R-C oscillator, the values of R5-R6 and C5 being chosen to give a frequency of oscillation in the region of 1 kHz. This audio tone is used to modulate the RF output of the marker generator.

The buffered output of the oscillator is applied to the clock input of IC1, a dual bistable flip-flop. This device allows alternate pulses to pass, thereby dividing the crystal oscillator frequency by two. The resulting 4 MHz marker at pin 1 is fed to the second clock input, and a 2 MHz signal is available at pin 13. This process is repeated with IC2 in order to produce 1 MHz and 500 kHz markers.

The 1 MHz signal is applied to clock input of IC3a, a dual decimal counter. The first stage of this IC divides the 1 MHz input by 10 to give 100 kHz. This signal is applied to the second stage for further division by 10 to produce a 10 kHz marker.

50 kHz and 25 kHz markers are useful when calibrating the long, medium and lower HF scales of a receiver, and to provide these the 100 kHz output from IC3a is applied to another dual bistable, IC4.

Transistor T1 acts as a modulator. Configured as an emitter follower, its base presents a moderately high impedance to the incoming signal from the divider ICs, the required marker being selected by rotary switch, S2. A low-impedance output is developed across emitter resistance, R2-R3. Socket K2 supplies a relatively low output level (20-50 mV) which may be safer for direct rather than inductive coupling to receiver inputs.

Switch S1 enables the 1-kHz oscillator when a modulated output is required.

With CMOS devices, propagation delay (the time taken for the output to change in response to a change of state at the input) is particularly dependent on supply voltage. IC5 and IC1, located at the start of the chain of dividers, are working fairly close to their maximum clock frequency, and the lowest supply voltage for reliable operation is 12 V.
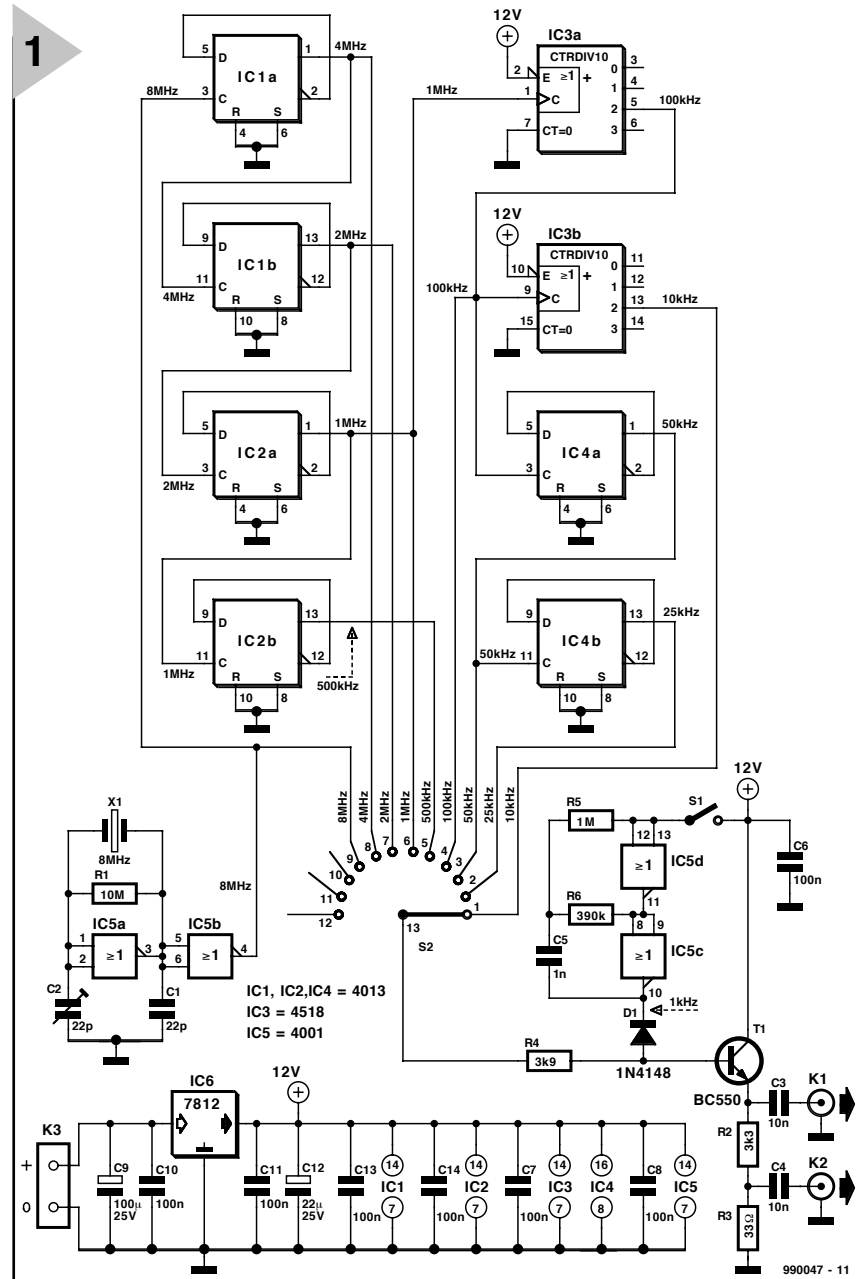
Figure 1. Circuit diagram of the RF marker generator.

Although crystal oscillators are renowned for their stability, operating frequency is shifted, to some small extent, by changes in the amplifier's supply voltage. Regulation of the supply voltage will therefore enhance the performance of the unit. Power is accordingly derived from two 9-volt PP3 batteries connected in series to give a nominal 18 V supply, which is dropped to the required 12 V as well as regulated by IC6. Bypass capacitor C12 shunts any electrical noise developed by the regulator to the ground rail.

Current drain with fresh batteries is in the region of 35 mA and this falls to around 25 mA when they reach the end of their useful life. Whilst this is not an excessive current demand on a PP3 battery, equipment of this kind tends to be operated for lengthy periods, and provision is made in the pro-

totype unit for powering it from a bench supply or an inexpensive (12-volt dc) mains adaptor.

CONSTRUCTION

All of the components, with the exception of switch S1 are mounted on a PCB. **Figure 2** shows the component mounting plan (overlay) and copper track layout. The board shown is not available ready-made.

Although the CMOS IC's are fitted with static protection diodes, these cannot be relied upon to prevent damage, and appropriate precautions against ESD should be taken when unpacking and installing the devices. Work on a grounded bench mat, and never insert or withdraw a CMOS IC, or work on the PCB, with the power supply connected.

IC sockets are fitted to permit rapid substitution checking (it is not

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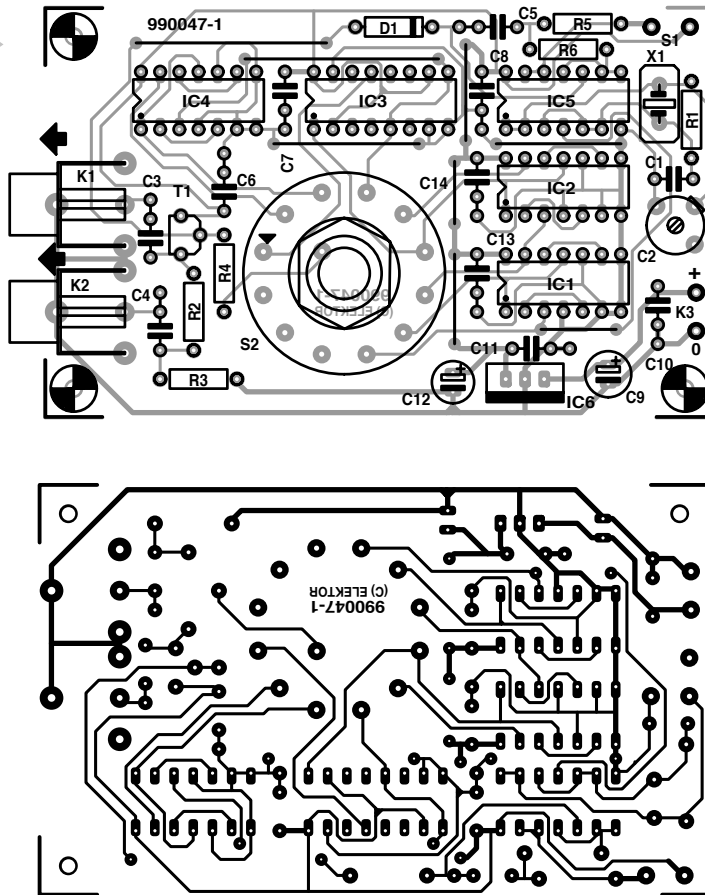
Figure 2. PCB artwork (board not available ready-made).

Figure 3. Finished PCB, ready for mounting into the enclosure.
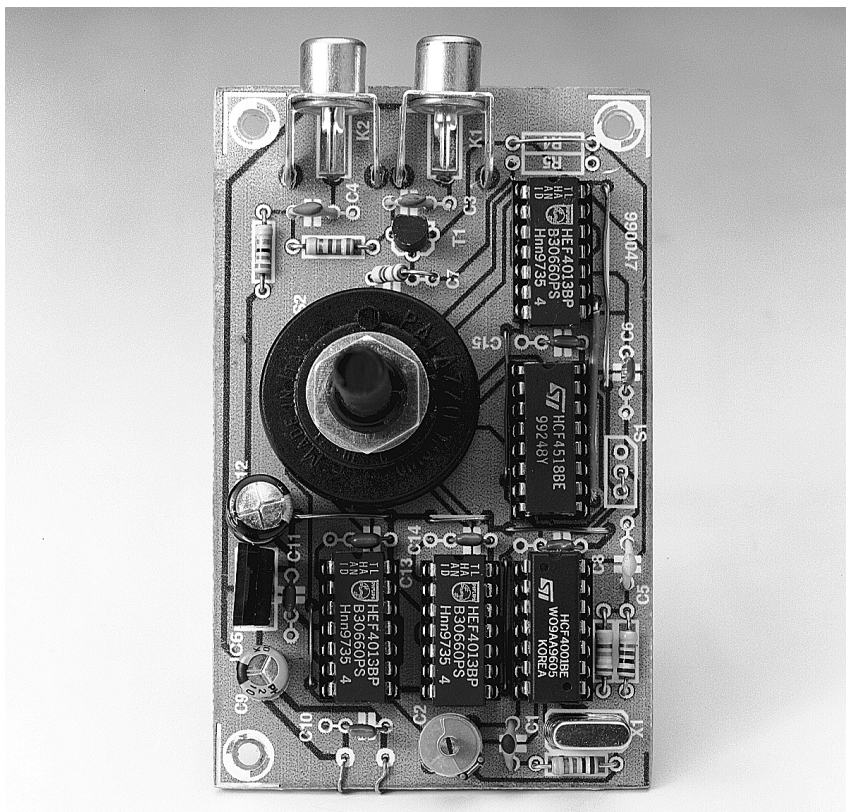
COMPONENTS LIST

Resistors:
R1 = 10 MΩ
R2 = 3kΩ3
R3 = 33 Ω
R4 = 3kΩ9
R5 = 1 MΩ
R6 = 390 kΩ

Capacitors:
C1 = 22 pF
C2 = trimmer, 22 pF
C3,C4 = 10 nF
C5 = 1 nF
C6,C7,C8,C10,C11,C13,C14 = 100 nF
C9 = 100 μF 25V radial
C12 = 22 μF 25 V radial

Semiconductors:
D1 = 1N4148
T1 = BC550
IC1,IC2,IC4 = 4013
IC3 = 4518 (see text)
IC5 = 4001
IC6 = 7812

Miscellaneous:
X1 = 8MHz crystal
K1,K2 = cinch socket, angled, PCB mount
S1 = on/off switch
S2 = 12-way 1-pole rotary switch, PCB mount
K3 = 2 solder pins

Components

All of the ICs and semiconductors, with the exception of the low current LED, are listed in JAB Electronics' catalogue (JAB, PO Box 5774, Birmingham, B44 8PJ). This firm can also supply the other components. Maplin retail all of the parts. They offer two variants of the 4518, and it is the device they list as HCF4518BEY dual BCD counter which is required for this project. Cirkit can supply everything except the 4518 IC.

unknown for ICs of this kind to be faulty when supplied) and minimise the possibility of damage during installation.

Finally, check your work against **Figure 3**.

TESTING

It is a good idea to check the functioning of the unit before the whole assembly is enclosed within a case.

First of all, check the PCB for poor soldered joints and bridged copper tracks. Check the orientation of the ICs and other semiconductors, in particular the voltage regulator IC: if this is connected into circuit wrongly, the full supply voltage could appear across the CMOS devices.

Connect the unit to batteries or a bench power supply. Current consumption with an 18 V supply to the regulator should be in the region of 35 mA. Check that the output from the regulator is 12 V.

If a frequency meter or an oscilloscope is available, use it to check the various spot frequencies. Otherwise, use a radio receiver to ensure that the circuit is functioning (guidance on injecting the signal into a receiver is given later).

ADJUSTMENT AND USE

If an accurate frequency counter is available, inject the 8 MHz marker (modulation switched off) and adjust C2 to 'pull' the crystal to the exact frequency.

If a radio receiver is to be used to set the calibrator, tune it to an appropriate transmission; e.g., Deutsche Welle, a 500-kW transmission on 6 MHz from Germany. Inject the highest possible marker spot frequency into the receiver, and adjust C2 for zero beat with the transmission. As the two frequencies draw closer, the pitch of the audible beat note emitted by the receiver will decrease to a slow fluttering and then cease (zero beat) when

the harmonic of the marker frequency exactly matches the frequency of the distant transmitter. The 1 MHz marker will, of course, have to be used for transmissions on an odd number of MHz, but 2 MHz can be used for the German station radiating on 6 MHz.

A short screened lead on K2 can be used to connect the output of the marker generator to the equipment to be calibrated, but a length of single flex on K1, placed near the receiver's aerial terminal, will usually be all that is required to secure adequate injection. Always keep injection levels as low as possible in order to minimise spurious responses within the receiver. Note that superhets generate images of the received signal, the most troublesome being spaced at twice the receiver's intermediate frequency from the transmission.

If the receiver is of the communications type, switch on the BFO and tune it to zero beat with unmodulated markers. This technique can also be used for the calibration of simple regenerative receivers if the regeneration or 'Q' multiplier control is advanced until the receiver is just oscillating. If the receiver is a domestic superhet or a TRF (Tuned Radio

Frequency) receiver without regeneration, the marker signals will have to be modulated in order to produce an audible tone.

Harmonics of all the markers extend upwards through the HF spectrum, and confusion will be avoided if the highest possible marker frequency is used to start the calibration process: e.g., begin by injecting a 1 MHz marker on medium waves (the band does not extend far enough to contain its harmonic on 2 MHz), and start with the 4 or 8 MHz markers as the calibration progresses through the short wave bands.

When these datum points have been established, work away from them using a lower frequency marker. On medium waves, 100 kHz would be appropriate, followed by 50 then 25 kHz in order to further sub-divide the scale. If the dial is large, 10 kHz could be injected, rather than 25 kHz, in order to produce a finer calibration.

The calibrator can, of course, be used as a means of accurately setting the tuning of a receiver with a poor or badly calibrated dial to the frequency of a particular transmitter.

WHEN ELECTRONICS WAS YOUNG (6)

After Morse had instigated the world's first telegraph line in 1843, practical men developed this new means of communications, and constructed improved and more reliable equipment. Before long overland lines were no longer sufficient and intercontinental lines were proposed. In 1850, a cable was laid between Dover and Calais. A similar operation failed when it was first tried in the Mediterranean (between Sardinia and Algeria), owing to that sea being much deeper in places than the English Channel. Only in 1857 did Newall & Co, with Werner von Siemens as adviser, achieve success and the first deep-sea cable was laid.
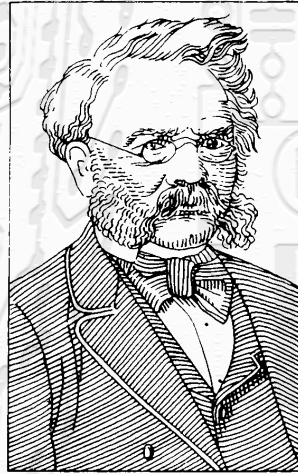
Werner von Siemens (1816...1892)

Following this success, the Agamemnon and Niagara, cable-laying ships of the London-based Atlantic Telegraph Company, laid 4,000 miles of cable linking Europe and America. When the work was completed in August that year, Queen Victoria and President Buchanan exchanged telegrams of congratulation. Three weeks later, the connection was suddenly interrupted and the telegraph machines stopped; it seems likely that moisture had penetrated the insulation of the cable. The fault was never found; to this day the cable lies at the bottom of the Atlantic.

It took many years before funds had been raised to attempt another trans-Atlantic cable. This was undertaken by the newly formed Telegraph Construction and Maintenance Company, using the biggest ship then afloat – the Great Eastern. This paddle steamer had a complement of 500 men, including 120 engineers and technicians of the cable company. It took five attempts, but in 1866 the permanent link between Europe and America was finally established.

By the end of the 19th century more than 250,000 miles of cable had been laid, establishing well over 300 links. Today, just over a century later, the cable link between France, Great Britain and the United States, finished in 1988, consists of a single fibre-optic cable, TAT-8, which is 4114 miles (6620 km) long, and carries most of the television, telephone, and data processing signals between these countries. An even later one (1992), TAT-10, is a direct 4436 mile (7320 km) long fibre-optic link between the USA and Germany via the Netherlands.

Of course, not everybody involved in physics and electrical engineering during the 19th century was concerned with cable-laying. In 1856, Ernst Werner von Siemens (1816–92) demonstrated a small, manually-operated dynamo that used a permanent magnet, and in 1866, the series dynamo. It was not until the Belgian-French inventor Zénobe Théophile Gramme (1826–1901) had built the first commercially practical generator for producing alternating current in 1867, however, that the world's first public electrical power plant could be taken into use (in Godalming, England, in 1881). The alternator used in this plant, manufactured by the firm of Siemens & Halske, developed 746 kilowatts.

In 1879, in America, Thomas Alva Edison (1847–1931) and in England, the British physicist and chemist Sir Joseph Wilson Swan (1828–1914) simultaneously introduced the first practical carbon filament lamp, which was a great improvement on the mercury arc lamp first demonstrated by Professor Wray on the Hungerford Suspension Bridge in London on 3 September 1860.

DIY: from vinyl to compact disk

with a PC and sound card

Nowadays, with the availability of personal computers and compact-disk (CD) writers, there is nothing in the way of transferring one's collection of vinyl records on to compact disks. All that may be needed in addition to the equipment already mentioned is a suitable preamplifier, such as the one presented in this article.

Brief parameters

Input sensitivity	
(moving-coil)	2 mV
(dynamic)	about 0.2 mV
Nominal output signal	200 mV
Signal-to-noise ratio	
(moving-coil)	78 dBa (750 Ω in)
	88 dBA (input short-circuited)
(dynamic)	70 dBA (25 Ω in)
	71 dBa (input short-circuited)

INTRODUCTION

The DIY making of compact disks is rapidly becoming a commonplace. One of the applications that is particularly attractive to many people is the digitizing of their valuable collection of vinyl records. There are, of course, other advantages than creating space (CDs take much less storage space than vinyl records): a compact disk has a longer life than a vinyl record (although it is not, as some people believe, infinite), and it becomes possible to select and shuffle sections of the recording if and as desired.

When a personal computer is available that incorporates a CD recorder (many modern ones are) and a good-quality sound card, the copying of vinyl records is straightforward. All that is then required is a means of linking the pickup output to the sound card. When the record player is placed next to the computer, the line outputs of the amplifier may be used. When this is not possible, there are a few difficulties. The output voltage of a dynamic pickup is about 3 mV and that of a moving-coil type around 0.3 mV. Clearly, these potentials are insufficient to drive the line input of the sound card. Moreover, the frequency response of the signal must be corrected.

RIAA CORRECTION

A vinyl record is cut tangentially, that is, the cutter traverses the disk in a straight line from disk edge to centre. The cutter response is called constant velocity, which means that its velocity is the same for all frequencies. Therefore, the amplitude increases as the frequency drops (at a rate of 6 dB/octave). It would thus be 16 times greater at 30 Hz than at 15 kHz.

Large low-frequency stylus excursions during playback are avoided by cutting the bass and boosting the treble frequencies to improve the signal to noise ratio. These contours roll off at either side of a short flat region centred on 1 kHz to form the RIAA (Recording Industry Association of America) characteristic. The playback amplifier or preamplifier has a frequency response that is a mirror image of the RIAA characteristic (see Figure 1).

DESIGN

The design of the preamplifier allows the output of dynamic as well as moving-coil pickups to be connected to its input.

Although the preamplifier is intended primarily for use as a converter between record player and personal computer, it is equally suitable for use with a hi-fi amplifier that has no integral phone input.

The block schematic of the preamplifier is shown in **Figure 2**. Each of the two stereo channels is linked to an input amplifier for moving coil pickups, which can be taken out of circuit by a wire bridge, followed by a standard amplifier for dynamic elements. The RIAA frequency-correction circuit is incorporated in this latter amplifier.

Note that for cases where the record player is linked to the computer for long periods, the line input is retained for other applications. To avoid the cumbersome changing of plug-and-socket connections, there is a change-over relay at the output, which ensures that when the moving-coil preamplifier is not used, the line input is connected to the relevant terminal(s) on the computer.

The power supply provides the ± 15 V lines for the operational amplifiers, as well as the single + 20 V line for the relay. It is preceded by a filter to eliminate any mains hum and interference.

Figure 1. RIAA recording and playback characteristics.

Figure 2. Block schematic of the preamplifier.
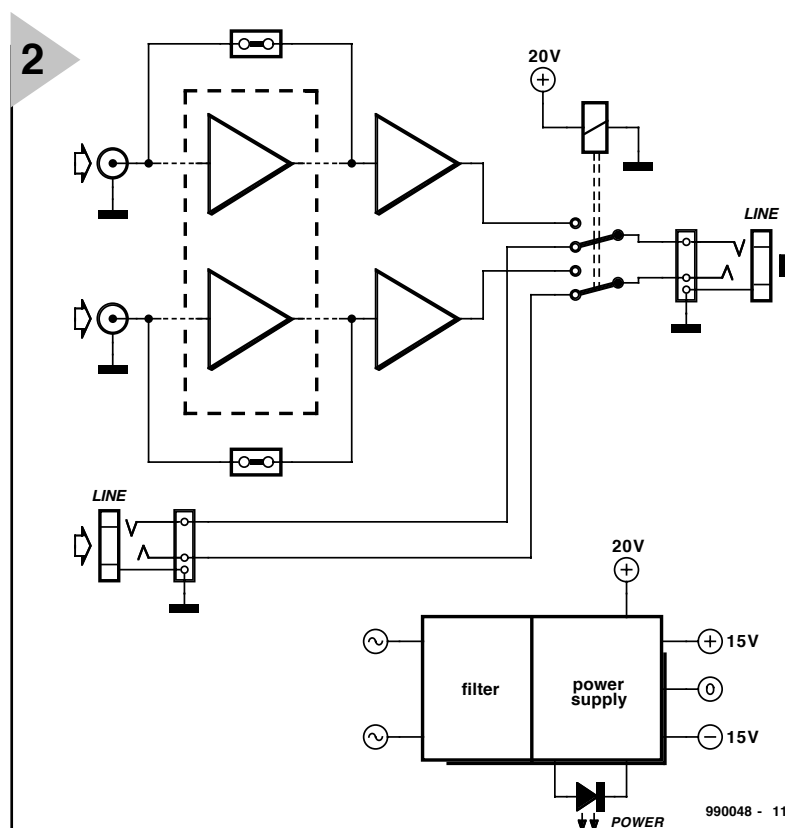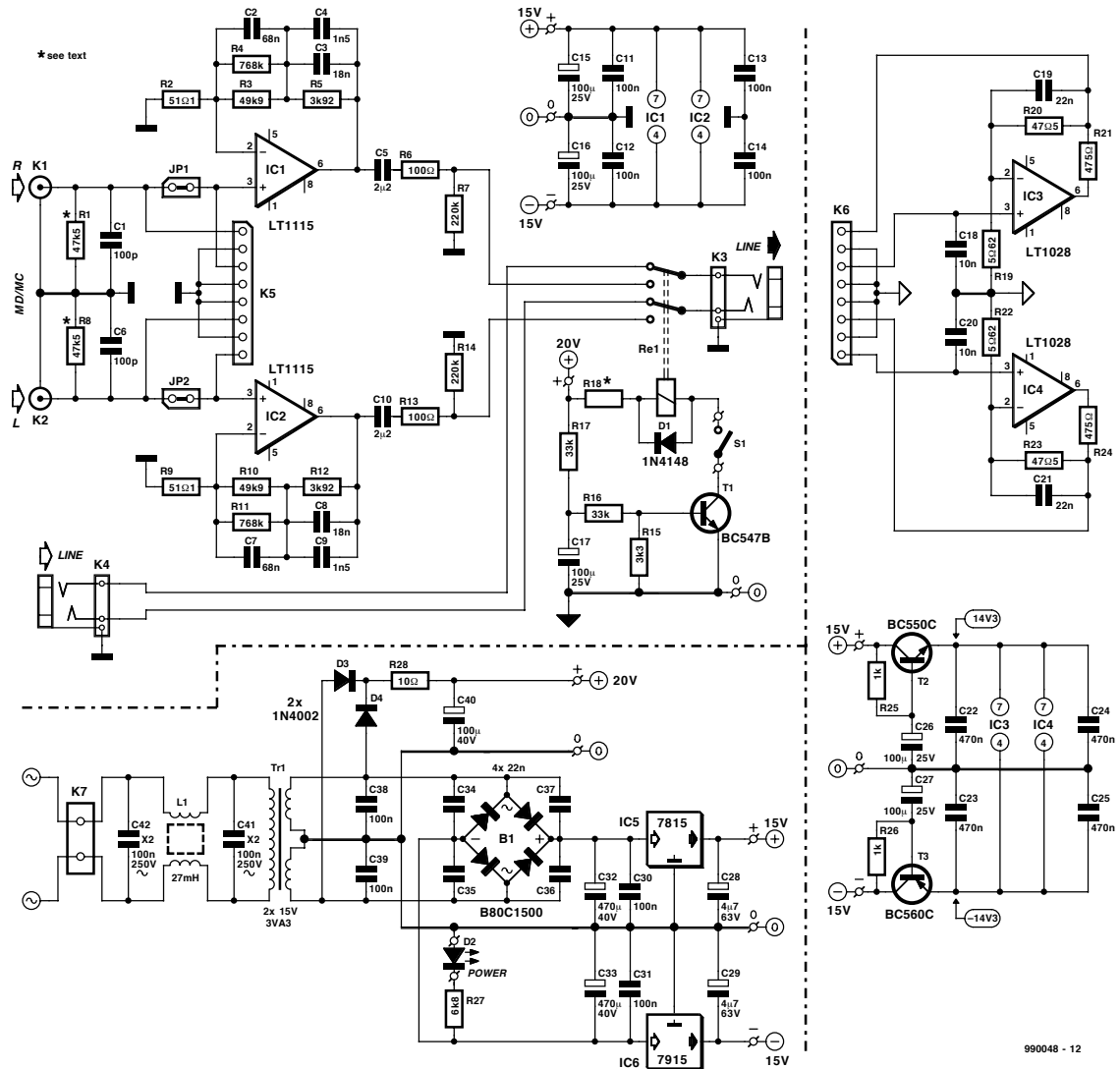
Figure 3. The circuit diagram may conveniently be split into three sections.

CIRCUIT DESCRIPTION

In **Figure 3**, the preamplifier for moving-coil elements is formed by IC_3 and IC_4, while the main amplifier is based on IC_1 and IC_2.

When the output of a dynamic pickup is linked to terminals K_1 and K_2, the input impedance has the standard value of 47 kΩ – determined almost exclusively by R_1 and R_8. Capacitors C_1 and C_6 determine the frequency response between 10 kHz and 20 kHz, which means that their value depends to some extent on the type of dynamic element used.

Operational amplifiers IC_1 and IC_2 are typified by a very low noise figure, a reasonably low bias current, and low input offset. When the output is 200 mV and the input is short-circuited, the amplifiers have a signal-to-noise ratio of 88 dB. In practical use, the noise of the amplifier is produced primarily by the pickup element. Note that the resistance and inductance of an average dynamic

element are about 750 Ω and 450 mH respectively.

The gain of IC_1 and IC_2 is 40 dB at 1 kHz. The RIAA correction network is included in the negative-feedback loop between pins 2 and 6. Capacitors C_5 and C_{10} decouple any offset, while resistors R_6 and R_{13} protect the operational amplifiers against capacitive loads. Resistors R_7 and R_{14} ensure that C_5 and C_{10} are charged in the absence of a load, which helps to prevent switch-on phenomena.

When the power is switched on, relay Re_1 is energized, whereupon the output of the amplifiers is linked to terminal K_3. When the supply is switched off, the relay is disabled, whereupon the additional line input at terminal K_4 is linked to K_3.

To avoid switch-on clicks and plops, the relay is energized with some delay provided by capacitor C_{17} via transistor T_1. Resistor R_{15} ensures that the relay is deenergized rapidly to guarantee that the supply to the amplifiers is switched off instantly.

Switch S_1 serves to enable manual switching of the relay between amplifiers and line input terminal; K_4 without the need of switching off the supply.

When a dynamic pickup element is used, jumper terminals JP_1 and JP_2 are closed. The sections based on IC_3 and IC_4 are then not used and need not be built.

When a moving-coil pickup element is to be used, JP_1 and JP_2 must remain open and resistors R_1 and R_8 must be replaced by 100 Ω types. Amplifiers IC_3 and IC_4 are included in the signal path via terminals K_5 and K_6. These amplifiers provide an amplification of about $\times 10$.

To ensure a low noise figure, the values of R_{19} and R_{22} are very low. To prevent this forming too large a load for the op amps, an additional resistor is used in the negative-feedback loop (R_{21} and R_{24} respectively). The resulting narrowing of the bandwidth is negated to a large extent by the use of very fast operational amplifiers.

4

Parts list

Resistors:
R_1, R_8 = 47.5 kΩ or 100 Ω (see text)
R_2, R_9 = 51.1 Ω
R_3, R_{10} = 49.9 kΩ
R_4, R_{11} = 768 kΩ
R_5, R_{12} = 3.92 kΩ
R_6, R_{13} = 100 Ω
R_7, R_{14} = 220 kΩ
R_{15} = 3.3 kΩ
R_{16}, R_{17} = 33 kΩ
R_{18} = see text
R_{19}, R_{22} = 5.62 Ω
R_{20}, R_{23} = 47.5 Ω
R_{21}, R_{24} = 475 Ω
R_{25}, R_{26} = 1 kΩ
R_{27} = 6.8 kΩ
R_{28} = 10 Ω

Capacitors:
C_1, C_6 = 100 pF, 63 V, 1%
C_2, C_7 = 0.068 µF, 63 V, 1%
C_3, C_8 = 0.018 µF, 63 V, 1%
C_4, C_9 = 0.0015 µF, 63 V, 1%
C_5, C_{10} = 2.2 µF, metallized poly-ester, pitch 5 mm or 7.5 mm
C_{11}–C_{14}, C_{30}, C_{31}, C_{36}, C_{38} = 0.1 µF
C_{15}–C_{17}, C_{26}, C_{27} = 100 µF, 25 V, radial
C_{18}, C_{20} = 0.01 µF
C_{19}, C_{21} = 0.022 µF
C_{22}–C_{25} = 0.47 µF
C_{28}, C_{29} = 4.7 µF, 63 V, radial
C_{32}, C_{35} = 470 µF, 40 V, radial
C_{34}–C_{37} = 0.022 µF, ceramic
C_{40} = 100 µF, 40 V, radial
C_{41}, C_{42} = 0.1 µF, 250 VAC, Class X_2

Inductors:
L_1 = 2×27 mH, 400 mA, 250 VAC

Semiconductors:
D_1 = 1N4148
D_2 = LED, green, high efficiency
D_3, D_4 = 1N4002
B_1 = B80C1500 (straight)
T_1 = BC547B
T_2 = BC550C
T_3 = BC560C

Integrated circuits:
IC_1, IC_2 = LT1115CN6 (Linear Tech-nology)
IC_3, IC_4 = LT1028CN8 (Linear Tech-nology)
IC_5 = 7815
IC_6 = 7915

Miscellaneous:
JP_1, JP_2 = 2-way pin header and jumper
K_1, K_2 = audio socket for board mounting
K_3, K_4 = 3.5 mm PCB mounting audio socket
K_5, K_6 = 8-way SIL (see text)
K_7 = 2-way terminal block for PCB mounting, pitch 7.5 mm
S_1 = single-pole, single-throw switch
Re_1 = 24 V relay, 2.2 kΩ
Tr_1 = mains transformer, 2×15 V secondary, 3.3 VA

Figure 4. The printed-circuit board for the preampli-fier is intended to be cut into two or three sub-boards.

Figure 5. The completed proto-type preamplifier for use with dynamic and/or moving-coil pickup elements.

than a 12 V type, which means that the (adverse) effect on the preamplifier of the ripple superimposed on this current is smaller.

The relay needs an energizing voltage of not less than 18 V, so that the 20 V provided in the present design is more than adequate. If a relay other than that specified is used, it may be possible to lower the current drawn by it by altering the value of R_{18}. Note that this resistor is not needed when the specified relay is used.

<div style="border:1px solid black; padding:8px;">

Table 1.
To lower the gain to 30 dB, alter the values of the following components as indicated.

R_2, R_9 = 162 Ω
R_3, R_{10} = 49.9 kΩ
R_4, R_{11} = 845 kΩ
R_5, R_{12} = 3.83 kΩ
C_3, C_8 = 0.02 μF
C_4, C_9 = 0.0012 μF

</div>

Table 1 shows the values of which components need to be altered if the line input of the sound card in the computer needs a lower level.

The board for the moving-coil type pickup element is linked to the main amplifier board via an 8-way single-in-line (SIL) connector, K_5, which is in essence a half IC socket. This, as well as the corresponding connector K_6, may also consist of an 8-way terminal strip. The two connectors or strips are linked by eight 15 mm lengths of 0.8 mm dia. insulated circuit wire. See the photograph in **Figure 5**.

To avoid any interference between the signal lines and the supply lines of the moving-coil board, the latter do not enter via K_6, but via three additional solder pins at the back of the board.

As mentioned earlier, the interior of a personal computer, and the space immediately surrounding it, are not exactly free of interference. It is, therefore, highly advisable to house the pre-amplifier in a well-screened metal enclosure.

[990048]

Capacitors C_{18} and C_{20} suppress any r.f. radiation. Since the impedance of moving-coil pickup elements is very low, the values of C_1 and C_6 are too low, which results in too wide a bandwidth. This is, therefore, narrowed by capacitors C_{19} and C_{21}.

Any interference on the supply lines to IC_3 and IC_4 is additionally decoupled by gyrators T_2 and T_3.

Regulators IC_5 and IC_6 provide stabilized ± 15 V lines from a traditional power supply. The 20 V supply for the relay is separately rectified and smoothed. Resistor R_{28} provides some filtering of the line.

Note that, because of the small signal voltages, the supply contains rather more r.f. decoupling than usual. Since the mains voltage in the vicinity of a personal computer often is not too 'clean', mains filter L_1-C_{42} is provided at the primary of mains transformer Tr_1.

Diode D_2 is the obligatory on/off indicator.

CONSTRUCTION

The preamplifier is best built on the printed-circuit board shown in **Figure 4**. The board consists of three sections, which may be cut apart. This is highly advisable as far as the power supply is concerned, since, in view of stray fields around the mains transformers, this is best kept as far away as possible from the amplifier section(s).

Construction should present no problems provided it is done with constant and careful reference to the circuit diagram and the parts list. There are, nevertheless, a few points that need special mention.

The output of the pickup element is linked to the preamplifier via audio terminals K1 and K_2. For best, long-life performance, use gold-plated types.

The line input and output terminals, K_3 and K_4, are standard 3 mm audio sockets.

Note that a 24 V relay has been used, since this draws a smaller current

weather-satellite decoder

a PLL-based intelligent interface

The decoder discussed in this article sits between the output of a weather-satellite receiver and a free RS232 port on your computer. Featuring a phase-locked loop (PLL), extensive filtering and a microcontroller for time-critical functions, it will faithfully translate demodulated audio signals into a serial datastream that can be processed by many of the popular wefax image processing programs like JVFAX.

Design by J. Altenburg

Weather satellite reception is a fascinating hobby on which, fortunately, lots of information is available from specialised groups. In the UK, for example, anyone with an interest in the present design should contact the Remote Imaging Group (RIG). We say this without hesitation because the RIG is for beginners and advanced users alike. Their superbly styled website at *www.rig.org.uk* contains a massive amount of useful information and links to hardware and software sources. As we've discovered ourselves during the preparation of this article, RIG members will be pleased to assist and advise not only on the choice of receiver,

antennas and the like, but also on test files, sound samples and the latest weather image processing software. If you are not on the Internet, then contact the RIG through The Secretary, John Tellick, 34 Ellerton Road, Surbiton, Surrey KT6 7TX.

GET THE PICTURE

As already mentioned, the place of the present decoder is between the receiver output and the RS232 port on your computer. One possible configuration of a weather satellite reception system is sketched in **Figure 1**. Note that there are two input 'channels': 137 MHz for low-orbiting (high-resolution) satellites
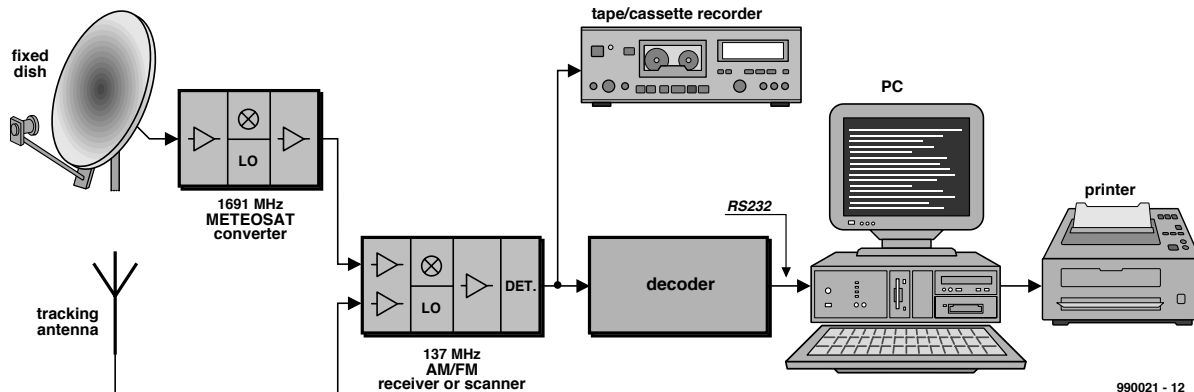
Figure 1. Components that make up a full-blown weather-satellite receiving system. If the system is equipped with a suitable clock control, the cassette recorder may be used for unattended recording of weather charts.

(NOAA, GOES) and 1690 MHz for the geostationary 'Meteosat' cluster. Usually, the Meteosat converter will employ 137 MHz as the intermediate frequency (IF), allowing its output signal to be feed to the input of the 137 MHz converter. The Meteosat converter usually gets its input signal from a parabolic dish, while the 137 MHz converter usually employs a cross-yagi with combined azimuth and elevation control by a tracking system. This subsystem is fed with the so-called Kepler elements you have to calculate for your location on the globe. Suitable programs? Look for Wintrack, PCTrack 3.1

and STSORBIT-Plus – hyperlinks to download sites are on the RIG's Software page!

Note that the equipment as shown in Figure 1 is a 'full-blown' system. Meteosat reception is optional, as is the tracking system for the 137-MHz antenna. If you are satisfied with just the

strongest part of the transmission ('pass') from a low-orbiting satellite like NOAA, then a simple dipole antenna and a wide-band (> 40 kHz) scanner/receiver may be used with acceptable results. Again, RIG members will be happy to advise!

THE SATELLITE SIGNAL is an audible signal with the main component (carrier) at 2400 Hz. Weather satellites (even Russian ones) generally use the

Figure 2. Circuit diagram of the decoder for wefax signals.
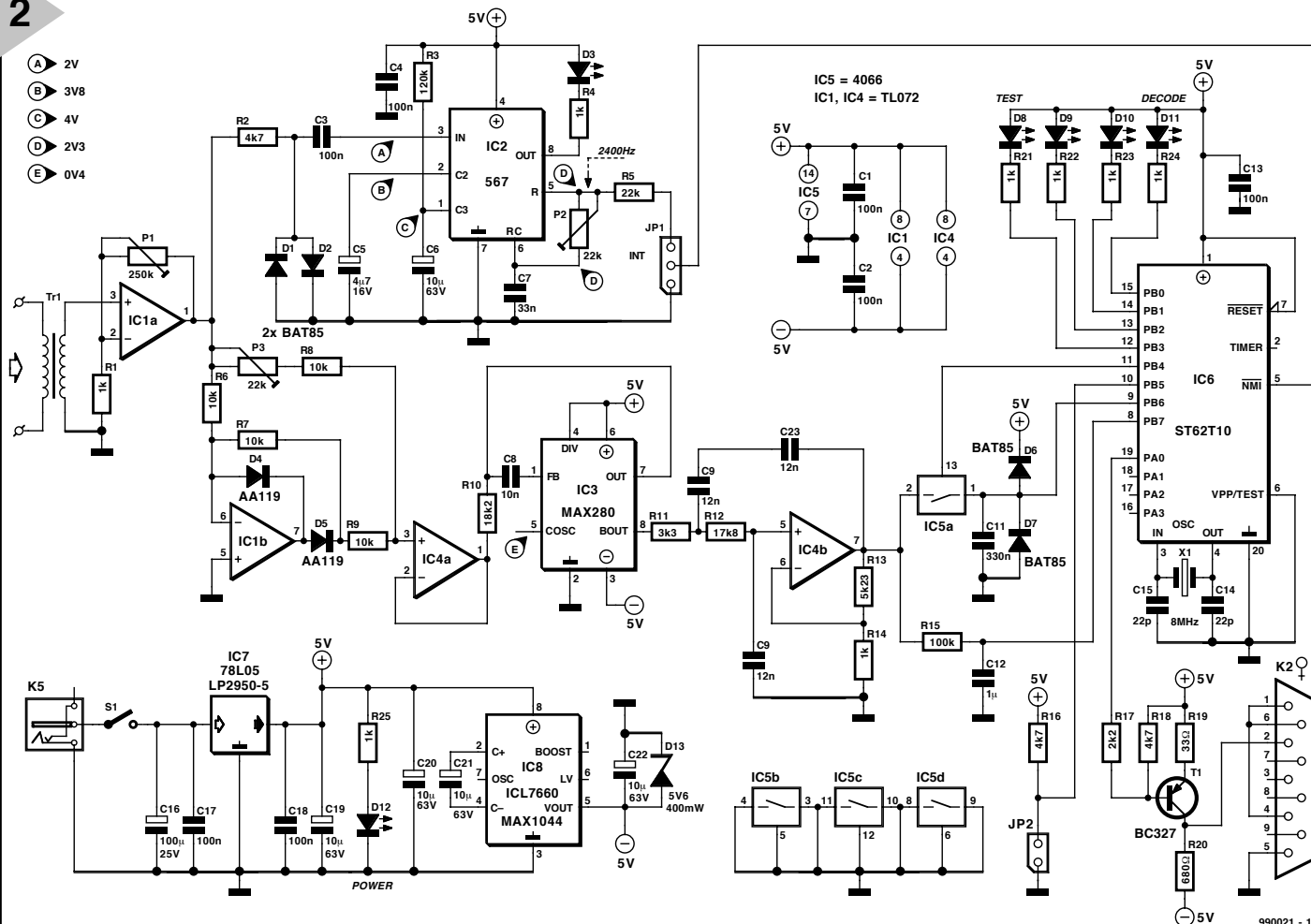
COMPONENTS LIST

Resistors:
R1,R4,R14,R21-R25 = 1 kΩ
R2,R16,R18 = 4kΩ7
R3 = 120 kΩ
R5 = 22 kΩ
R6-R9 = 10 kΩ
R10 = 18kΩ2 1%
R11 = 3kΩ3
R12 = 17kΩ8 1%
R13 = 5kΩ23 1%
R15 = 100 kΩ
R17 = 2kΩ2
R19 = 33 Ω
R20 = 680 Ω
P1 = 250 kΩ preset H
P2,P3 = 22 kΩ preset H

Capacitors:
C1-C4,C13,C17,C18 = 100 nF
C5 = 4µF7 16V radial
C6,C19-C22 = 10 µF 63V radial
C7 = 33 nF
C8 = 10 nF
C9,C10,C23 = 12 nF
C11 = 330 nF
C12 = 1 µF MKT
C14,C15 = 22 pF
C16 = 100 µF 25V radial

Semiconductors:
D1,D2,D6,D7 = BAT85
D3 = LED red, 3mm, low current
D4,D5 = AA119 or BAT85
D8-D11 = LED, yellow, 3mm, low
  current
D12 = LED, green, 3mm, low current
D13 = zener diode, 5V6, 400mW
T1 = BC327
IC1,IC4 = TL072 CP
IC2 = LM567CM
IC3 = MAX280CPA
IC5 = 4066
IC6 = ST62T10 B6-HWD (order code
  996513-1)
IC7 = 78L05 or LP2950 CZ5.0
IC8 = ICL7660 CPA or MAX1044

Miscellaneous:
JP1 = 3-way SIL pinheader with
  jumper
JP2 = 2-way pinheader with jumper
K2 = 9-way sub-D socket (female),
  angled pins, PCB mount
K5 = mains adaptor socket, PCB
  mount
S1 = on/off switch, 1 contact
TR1 = line transformer, Monacor
  (Monarch) type LTR110 or MTR120
X1 = 8MHz quartz crystal
Clip-on lead for 9V PP3 battery
9 volt PP3 battery or 9V 300mA
  mains adaptor
PCB, order code **990021-1** (see
  Readers Services page)
Disk, contains ST6 source code file
  and Satview program, order code
  996019-1 (see Readers Services
  page).

APT format (Automatic Picture Transmission), which is a mix of FM (frequency modulation) and AM (amplitude modulation). The picture information is amplitude-modulated on a carrier, which, in turn, is frequency modulated. Cumbersome and hopelessly outdated as it may seem, this 'AM-in-FM' packaging is actually pretty ingenious because it avoids the complex compensation of Doppler shift at the receiver side. Remember, all low-orbiting satellites travel at *very* high ground speeds, typically covering a horizon-to-horizon arc in just a few minutes.

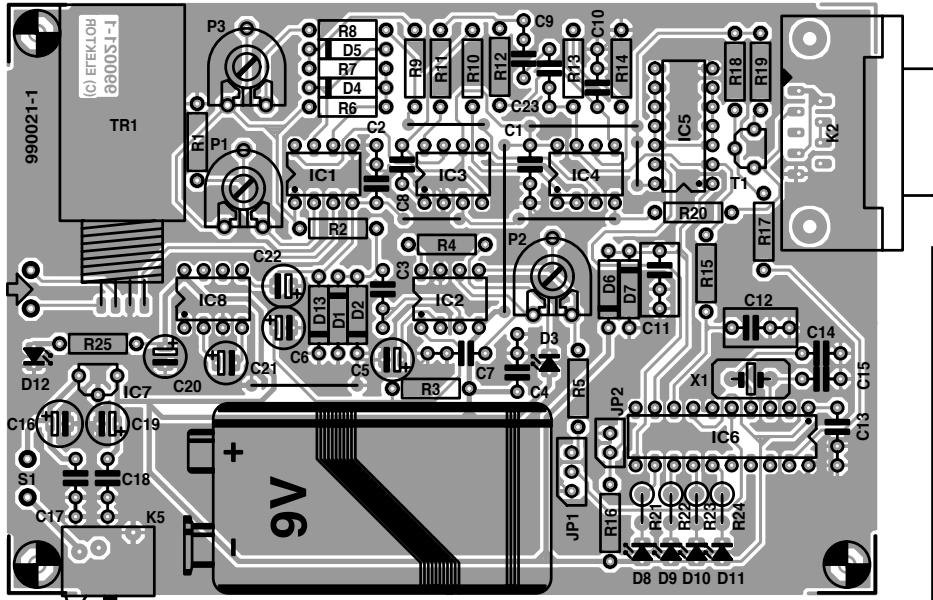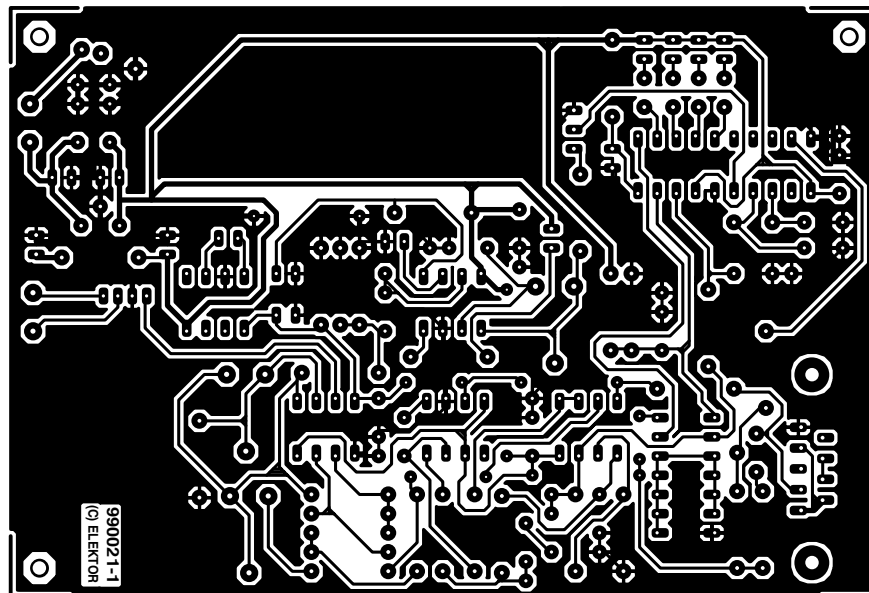The picture transmission rate is usually two lines per second. The signal will typically sound like a hiss interrupted by two 2400-Hz beeps per second. The 2400-Hz tone is also dominant in the picture leader and trailer syncs. The highest frequency of the picture information proper ('pixels') is about 1200 Hz.

Because it is assumed that the weather-satellite signal has been properly demodulated before it is applied to the present decoder, a detailed analysis of its structure is beyond the scope if this article.

HARDWARE DESCRIPTION
The circuit diagram of the converter is shown in **Figure 2**. The circuit is a combination of analogue and digital components.

The audio input signal reaches input amplifier IC1a via a line transformer, Tr1. This is included for electrical isolation and to keep digital noise (8-MHz ST6 clock) away from the receiver. Preset P1 allows an output level of 2.3 V to be set. The opamp output signal is sent to a PLL (phase-locked loop), IC2, and a rectifier/buffer, IC1b-IC4a, which is followed by a filter built around IC3 and IC4b.

Let's first look at what the PLL does. The inexpensive and well-tried NE567

Main decoder routine

The main grey-level decoding routine executed by the ST microcontroller in this project performs the following sequence:
1. Initialise register x with value 00 (aux. value for jump instructions)
2. Enable interrupts
3. Disable timer interrupt
4. Initialise variable 'dcb' (all LEDs off)

Inside 'loop', the following is done:
5. Copy contents of 'dcb' to port B (open sample&hold switch, make LEDs light)
6. Wait for falling edge of NMI

The NMI handling consists of:
7. Close sample&hold switch
8. Read A/D converter
9. Send start bit to serial channel
10. Start A/D conversion using input 'average'
11. Reload watchdog
12. Send bits 0-7 to serial channel
13. Read A/D converter
14. Send stop bit to serial channel
15. Start A/D conversion using input 'sample&hold'
16. If A/D value< 70 then code for 1 LED to 'dcb', go to 'loop'
17. If A/D value< 100 then code for 2 LEDs to 'dcb', go to 'loop'
18. If A/D value< 150 then code for 3 LEDs to 'dcb', go to 'loop'
19. Code for 4 LEDs to 'dcb', go to 'loop'

The timing of these routines is critical. At 2400 Hz, a time slot of 417 μs is available to perform both A/D conversions and transmit the value serially. Fortunately, the A/D conversion may be performed while serial data are being transmitted **and** the microcontroller determines which of the 4 LEDs is to be switched on. The bit rate is 38,400 per second, or 26 μs per bit. In this time slot, the processor executes 16 machine cycles.
A software cycle starts with waiting for the NMI signal supplied by the PLL. During the previous cycle, the A/D converter already measured the input voltage at the pixel input (i.e., the input with sample&hold). The conversion result is held ready in the A/D latch. At step 7, the sample&hold switch is closed, allowing the capacitor to track the input. At step 8, the A/D converter is read. To save time, a start bit is already sent. Next, a new A/D cycle is started, this time the average value at input 8 is measured (PB7), this value is later used to drive the LED bar. One A/D conversion lasts at least 70 μs. No problem, we'd say, because a 10-bit serial word is sent at the same time, lasting 10x26= 260 μs.
Next, the 8 databits are transmitted. The overall timing is critical as discussed further on. While the 8th databit is being transmitted, at step 13 the A/D is already being read (voltage at 'average' input). Meanwhile the time for the last bit has elapsed, and the stop bit is sent (step 14). Next, we start the A/D converter, this time sampling the pixel input. The measured average value is compared checked against a couple of thresholds to allow the right

code to be sent to 'dcb' for the control of the LEDs. The code to open the sample&hold switch is also sent. Next the program counter is returned to 'loop' and the micro waits for NMI again. The A/D conversion takes place while the micro waits for a new NMI pulse edge.
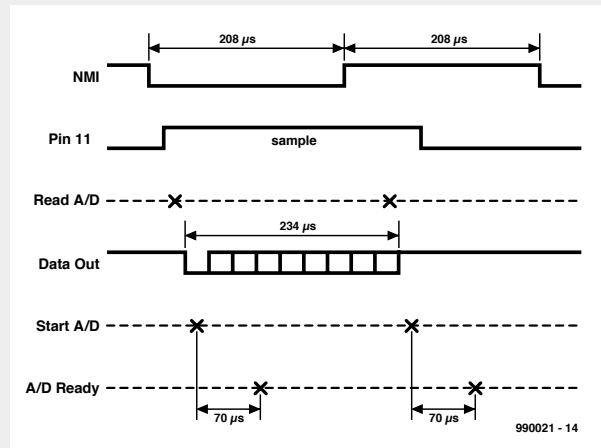
The timing is calculated as follows:
NMI to data-out: 19.5 μs
Serial out: 234 μs
Which LEDs on/off 42.5 μs
Data to port B: 19.5 μs

The total duration of 316 μs corresponds to a frequency in excess of 3 kHz.
The serial signal is generated as follows. The byte in accumulator A has to be transmitted serially, LSB-first. "Piece of cake" you'd say, "just use the usual Rotate or Shift-Left instructions". Surprise, surprise, the ST6 does not have these instructions! Not to be put off, the author devised a solution that does not compromise the control of the LEDs. As shown below, it involves conditional jumps and accurately 'tweaking and padding' the number of machine cycles to make sure the routine always takes the same amount of time to execute whether or not the jump condition is satisfied.

Label	instruction	comment	no. of machine cycles
	Jrs 2a, hp15	Test bit 2, jump as 1	5
	Nop		2
	Res 0, DRA	write 0	4
	Jrr 0, x, hp16	X = 0, so always jump	5
Hp15			
	Nop		2
	Set 0, DRA	write 1	4
	Jrr 0, xhp16	X = 0, so always jump	5
Hp16			

In this way, 16 machine cycles are <u>always</u> executed, corresponding to 26 μs and a data rate of 38,400 bits/second.

is used here in a standard configuration. Diodes D1 and D2 limit its input signal to about 0.4 V. Preset P2 sets the centre frequency to 2400 Hz, enabling the carrier reference clock to be recovered from the composite APT signal.

The signal rectifier (IC1b) is an active 2-phase type whose half-phase output levels can be made equal by adjusting preset P3.

Although it employs just one IC and an opamp, the filter is a seven-pole type! Extensive filtering is required because of the relatively small difference between the carrier frequency

(2400 Hz) and the highest data frequency (approx. 1200 Hz). Remember, in traditional transmission technology, a rule of thumb is that carrier frequencies are at least 10 times higher than the highest modulation frequency. This is done to facilitate extracting the modulation signal. The filter consists of three elements: (1) low-pass R10-C8, (2) IC2 and (3) Butterworth section IC4b. The first is dimensioned for cut-off frequency of about 1400 Hz. The second element, says Maxim Inc., is a '5th order all-pole instrumentation lowpass filter with no dc error'. Its main technical data may

be found on the *Datasheets* on page 61-62 of our March 1999 issue. The filter around IC4b is a Butterworth section which further adds to the selectivity for the pixel component in the APT signal.

The filtered picture data represents grey levels. It is used to charge a capacitor, C11, via electronic switch IC5a. This capacitor acts as a memory device when the microcontroller is busy performing an A-D conversion. The switch is controlled by microcontroller port line PB4.

The average value of the pixel grey levels is read by the ST6 via port line
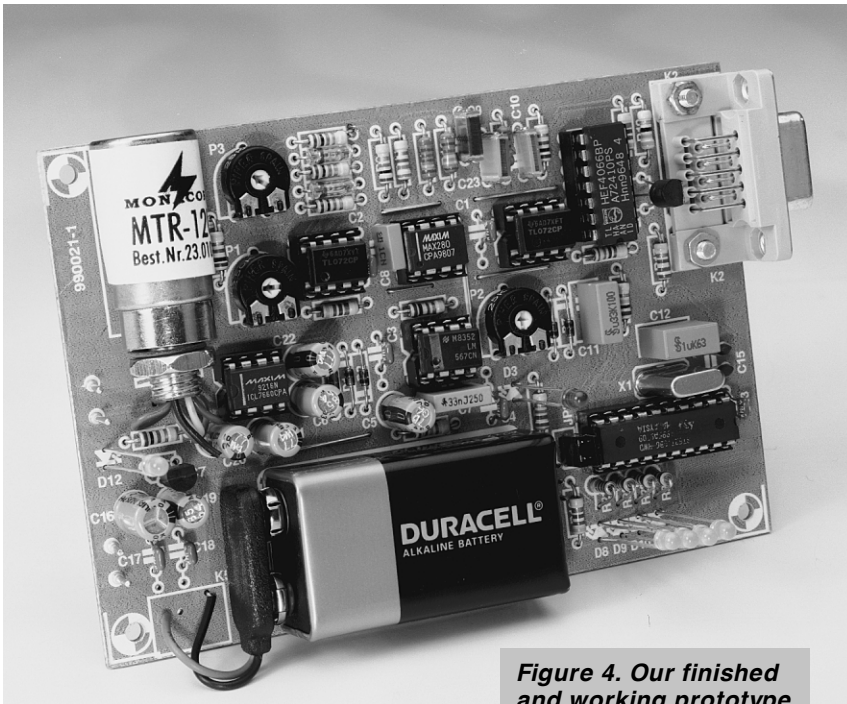
Figure 4. Our finished and working prototype of the weather satellite decoder.

PB7. It is obtained by low-passing the filter output in network R15-C12.

The ST6 microcontroller (IC1) runs a program written to perform quite a few functions quasi-simultaneously. These functions include A-D conversion of the pixel grey levels, supplying a serial datastream to the PC (via T1 and K2) and driving a LED array (D6, D9, D10, D11). The **inset** provides essential information on the operation of the software.

The power supply is fairly conventional. Either a 9-V PP3 battery or a mains adaptor may be used to supply an unregulated input voltage of about 9 V dc. This is stepped down to a regulated 5-V rail by a 78L05 (IC7). The negative (–5 V) auxiliary voltage for the TL072 opamps is generated by an ICL7660 (IC8) in a standard application circuit. A zener diode (D13) is added for safety.

CONSTRUCTION

The artwork layout shown in **Figure 3** is that of a compact single-sided PCB accommodating *all* parts including the battery. This board is available ready-made through our Readers Services and kit suppliers advertising in this magazine. Construction is straightforward if you work carefully and systematically. In particular, make sure you get the following right: 1% resistors (measure value before fitting) and all polarised components (diodes

Figure 5. Example of a weather satellite image received with the decoder in combination with the 'Satview' program.

including LEDs, ICs, electrolytic caps and transistor T1).

ADJUSTMENT

For the following adjustment procedure it is assumed that you have available a recording of a weather-satellite signal (these may be found on the web as 'wave' files).

1. Apply the input signal and adjust P1 for a level of 2.3 V_{pp} on pin 1 of IC1.
2. Adjust P2 until the PLL locks onto the 2400 Hz carrier, when D3 will light.
3. Connect an oscilloscope to pin 1 of IC4 and adjust P3 for best symmetry of the two rectified half-waves.
4. Now first set up the serial link to the PC. Run a terminal emulation or communication program (say,

HyperTerminal), and set up a data rate of **38,400, 8 bits, no parity, 1 stop bit**, **no handshaking**. Next, install JP2 on the converter board. JP1 should be set to the 'ground' position. The converter should write a continuous stream of gobbledegook characters among which can be read the words START!! and 'Bildtest'. If this works, the serial link is okay. If not, debug it first. Have you fitted a *socket* in position K2? Is your serial cable a *straight-through* type (no crossed wires)? Is the *RS232 port* on your PC properly set up?
5. Remove JP2 and install JP1 so that it is at the side of R5.

Quit your communications program. Launch the wefax or image processing program — if correctly set up on the PC, the weather image can be seen to build up line by line.

IMAGE PROCESSING SOFTWARE

Several excellent programs are available for weather image reception and processing on a PC. The best known of these is the formidable JVFAX package written by Eberhard Backeshoff, DK8JV. Version 7.1 of this program may be found on various web sites and ftp sites (*ftp.funet.fi*). Note, however, that JVFAX is neither shareware nor public domain — see the author's Copyright notes on page 50 of the (hefty) manual. JVFAX is not an easy program to master but it excels in the number of weather-satellite modes and standards supported. If you are a newbie, don't expect to have JVFAX up and running in less than an evening.

Other suitable programs include WXSAT 2.4 and SatView. The latter, a Windows program, was written specially for the present decoder and may be found on the project diskette (order code **996019-1**) which also contains the source code file for the ST6 microcontroller (METEOSAT.ASM).   (990021-1)
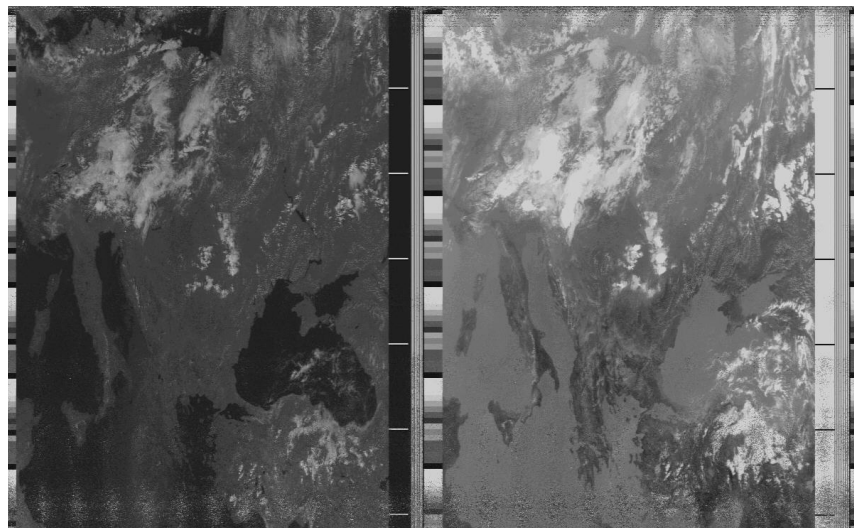
conductive plastics

Of luminescent plastics and plastic transistors

Plastics are polymers, that is, chains of many identical molecules (monomers) that are intercoupled. The reason that most plastics are isolators is that their electrons are localized. Each electron is firmly fixed, as it were, to its own atomic nucleus. This means that the electrons, carriers of the electric current, cannot move freely in the material.

In conductive or semiconductive plastics, discovered by accident in Japan in 1977, this is different. In these, the polymer chains have conjugated connections. This means that the discrete atoms are interconnected alternately by a single and a double chemical bond.
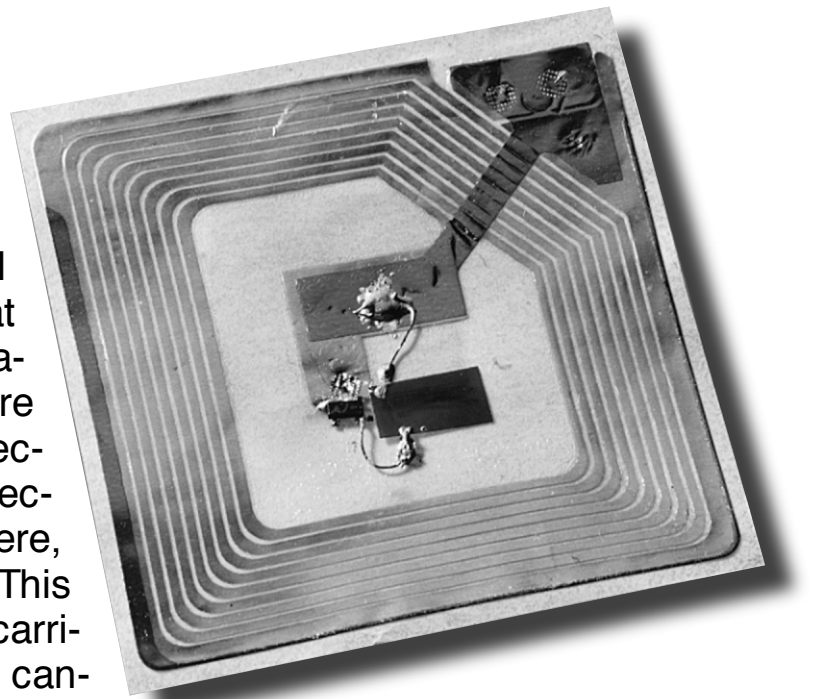
DEVELOPMENTS TO DATE

After conductive polymers had been discovered by accident in Japan in 1977, researchers at the University of Cambridge in England discovered luminescent polymers in the late 1980s. In line with modern practice, the university hived off this discovery for commercial exploitation to Cambridge Display Technologies (CDT). The first luminescent polymer, polyphenyl-vinyl or PPV, is patented by CDT. The company has granted a licence to Philips of the Netherlands for the commercial exploitation of the polymers.

Today, CDT is developing flexible displays based on PolyLEDs and has already demonstrated a prototype of a polymer display, which was developed in cooperation with Seiko-Epson. It concerns a small, monochrome display that as yet does not consist solely of polymers: the electrodes are made of indium tin oxide (InSnO2) and aluminium (Al). Nevertheless, the performance is remarkable: its luminosity is four times as great as that of liquid-crystal displays (LCDs) and does not suffer from the limited viewing angle that is so characteristic of LCDs.

The German firm of Hoechst collaborates with Philips in the manufacture and further development of PolyLEDs. Hoechst has also joined forces with Uniax, an American firm in California which has produced the first seven-segment display consisting of PolyLEDs.

The company that succeeds in producing truly flexible displays will have the commercial world at its feet. The number of possible applications of PolyLEDs runs into the hundreds. However, the technological difficulties to market these applications are still a formidable obstacle.

Figure 1. Chemical structures of some often used conjugated polymers: polyphenyl-vinyl or PPV; polyphenylamine (often called polyaniline) or PANI; and polypyrrole or PPy. The polymers form long chains consisting of thousands of monomers that are coupled in tandem. Characteristic of all semiconductive polymers is their conjugated structure, which means that their bonds are alternately single and double.

RELEVANT PHYSICS

Conductive or semiconductive plastics are polymer chains with conjugated double links. The first PolyLEDs were based on polyphenyl-vinyl (PPV). The principle of their conductivity (or, rather semiconductivity) is best illustrated by the simplest polymer with a conjugated structure: polyacetylene. See **Figure 1**.

The single bond in the conjugated structure is always a σ-bond, whereas the double one consists of a σ-bond and a π-bond, which has a different character. Two variants of polyacetylene that differ only in the locations of the π-bonds are shown in **Figure 2**. It could be said that these variants could be merged freely. The real structure is a mixture of the variants in which each is represented equally.

This has an important consequence: in the case of an σ-bond, the electrons forming the bond are bonded to both nuclei and therefore localized. Normally, this is also the case with electrons forming a π-bond. Because of the conjugated structure,

that is, a mixture as in Figure 2, the electrons are free to move along the entire chain.

This does not mean, of course, that the material itself, which consists of many monomers, becomes conductive. This occurs only when electrons can hop from one chain of polymers to another. It has been found that this becomes possible when the chains are in close proximity of each other. The closer the chains are together, the more mobile the electrons become. This is further enhanced by purification of the

material and doping it, that is, adding charge carriers.

PROCESSING

In their original state, semiconductive polymers cannot be easily processed. This is because the conjugated structure of double bonds results in flat, inelastic chains. The addition of large, flexible side chains—see **Figure 3**— renders the polymers solvent, which simplifies their processing. Varying the chemical composition of the side chains enables properties such as solubility, durability and conductivity to be enhanced.

The great benefit of conductive polymers is that the area of devices made from them is virtually limitless. Conventional semiconductor technology is heavily involved in miniaturization; silicon, the basic material, is eminently suitable in this development. The production of large devices is, however, much more complicated and expensive, because the material used must be crystalline. This means that its lattice must not contain even the tiniest distortion. The production

Figure 2. The normal chemical notation of the structure formula of, for instance, polyacetylene, offers two possibilities. These suggest that the double bonds are localized. In reality it concerns a mixture of the two.
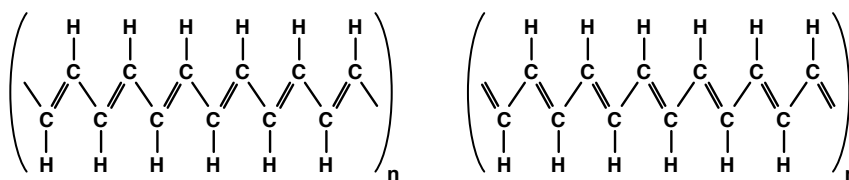
3
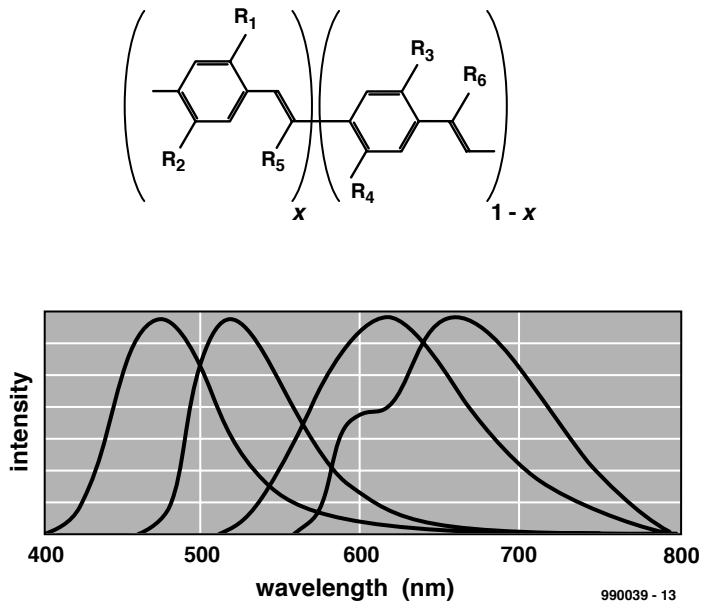
Figure 3. The addition of side chains (R₁–R₆) to polyphenyl-vinyl (PPV) results in the emission of light in all possible colours. Each of these colours corresponds to a different chemical composition of the side chains.

of silicon is tedious and requires accurately defined ambient conditions. The larger the crystal, the more difficult its production becomes. This is not the case with plastics, so that polymers may open the way to large semiconductor components.

THE POLYLED
When an electric potential is applied across semiconductive plastics, they emit light. This property, discovered by researchers at the University of Cambridge in England, forms the basis of the PolyLED..

The PolyLED is essentially a much simpler component than a transistor. Its applications include segment displays such as used in mobile telephones and background lighting in liquid-crystal displays.

PolyLED technology enables the manufacture of thinner displays than is possible in liquid-crystal technology. Moreover, PolyLEDs operate with low (battery) voltage and are therefore eminently suitable for use in modern equipment.

PolyLEDs are also very thrift. Today, the luminous flux is of the order of 4–8 lm W⁻¹ (the picture tube in a current TV receiver emits 1 lm W⁻¹ and an LC display, 1.5 lm W⁻¹. Researchers aim to produce conductive polymers that produce a luminous flux of 25 lm W⁻¹. In comparison, an incandescent bulb emits about 20 lm W⁻¹ and a fluorescent tube about 70 lm W⁻¹.

A further advantage is that since light is emitted omnidirectionally, there is no limiting reading angle.

CONSTRUCTION OF THE POLYLED
A PolyLED consists of a structure of a conductive polymer sandwiched between a cathode and an anode on a substrate of glass or transparent plas-

tic—see **Figure 4**. The electric potential between cathode and anode ensures that a constant stream of electrons enters the polymer from the top. The drainage of electrons via the anode creates vacancies (holes) in the polymer that are mathematically equivalent to positrons (positive electrons). The incoming electrons fill these vacancies so that both hole and electron disappear. In this process, the electron emits a quantum of energy in the form of a photon.

The cathode and anode are fabricated from calcium (Ca) or indium-tin oxide ($InSnO_2$). The latter material is transparent, which is, of course, necessary if light is to be passed. The electrodes may be patterned, so that not only flat luminous structures can be produced, but also company logotypes, segments of displays, and even pixels of a display screen.

POSSIBLE APPLICATIONS
In theory it is possible to produce a light-weight colour television screen from conductive polymers. Although this is futuristic, researchers have already succeeded in producing a matrix display with a vertical and horizontal resolution of 100 pixels on which graphics images could be represented.

All initial research was directed at the removal of technical barriers, for which orange PolyLEDs were used. Today, it is possible to fabricate red, green and blue PolyLEDs. Since these materials behave in a virtually identical manner, with these three primary colours it is possible to make polymers of any desired colour. The requisite potential across the polymer is 3–5 V and the brightness of the emitted light depends on the level of current through the material.

A durability of about 50 000 working hours has already been achieved.

TRANSISTORS ALSO
Apart from diodes, it is also possible to make plastic transistors from conductive polymers. With reference to **Figure 5**, the transistor is deposited on a substrate of polyimide. (1).

With spin-coating*, the first electrode is deposited on to the substrate. This consists of a 200 nm thick layer of polyphenylamine (polyaniline or
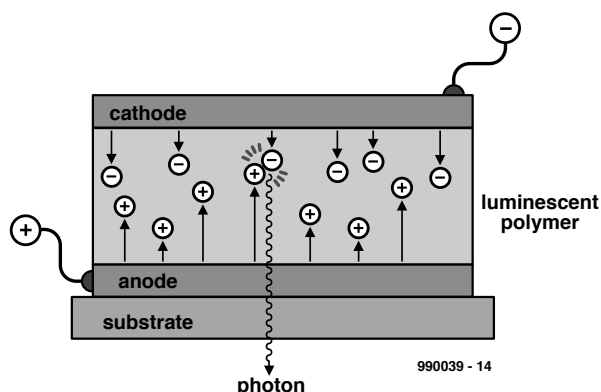
4

Figure 4. A PolyLED consists of a luminous polymer sandwiched between two electrodes with the combination bonded to a substrate of glass or transparent plastic.

PANI). The polyphenylamine is deposited in liquid form to which a photo-initiator has been added. (2).

The electrode is illuminated by ultraviolet light via a stencil. Owing to the photo-initiator, the electrical resistance of the illuminated areas increases by a factor 10^{10}, turning the areas into isolators. The non-illuminated areas remain conductive. (3).

The electrode is heated whereby the residue of the photo-initiator evaporates. (4)

With spin-coating, the next two layers are deposited: a 50 nm thick film of semiconductive PTV and a 250 nm thick film of polyvinyl pyrrolidone (PPV—an isolator). (5)

The second polyphenylamine electrode, to which a photo-initiator has been added, is deposited by spin-coating. (6)

The second electrode is illuminated by ultraviolet light via the same stencil as used earlier. (7).

The final result is a plastic transistor which may be linked to other transistors to form an integrated circuit via a gate, source and drain. The electrical properties of the transistor are determined primarily by the PTV film. (8)

[990039]

5

Figure 5. A plastic transistor can be produced from conductive polymers by a step-by-step lithographic process. The end result is a four-layer wafer bonded to a polyimide substrate.

1. substrate

2. depositing the first electrode — polyphenylamine electrode

3. illuminating — UV

4. heating — isolating / conducting

5. depositing semiconductor and isolator — isolator / semiconductor

6. depositing second electrode — polyphenylamine electrode

7. illuminating — UV

8. transistor — gate / isolator / semiconductor / substrate / source / drain / electrodes

990039 - 15

Figure 6. Spin-coating is a technique that enables a thin film of one material (in liquified form) to be deposited on to another (solid) material.

* Spin-coating is a technique that enables a virtually homogeneous film of liquified material 100–200 nm thick to be deposited on to another material, The solid material, such as the substrate of a diode or transistor, is made circular and the resulting disc is made to rotate at very high speed. A drop of the liquified material is dropped on to the disc and spreads out into a thin, homogeneous film. When the solvent (used to liquify the material to be deposited) has vaporized, the film solidifies. There are virtually no limitations to the size of substrate that can be used in spin-coating. See Figure 6.

Dip-coating is a similar process in which use is made of the bond that can be formed between substrate and film material. In this process, however, the substrate is briefly immersed into the soluble polymer.

development system for 68HC11F1

accepts eight different memory devices

Motorola's 68HC11 microcontroller is a highly valued device when it comes to developing applications based on microcontrollers. This article present a low-cost way for you to start creating such applications based on the 68HC11. Unusually, the present development system is marked by a relatively large amount of system memory.

Design by J. Gonzalez and A. Amaranthe
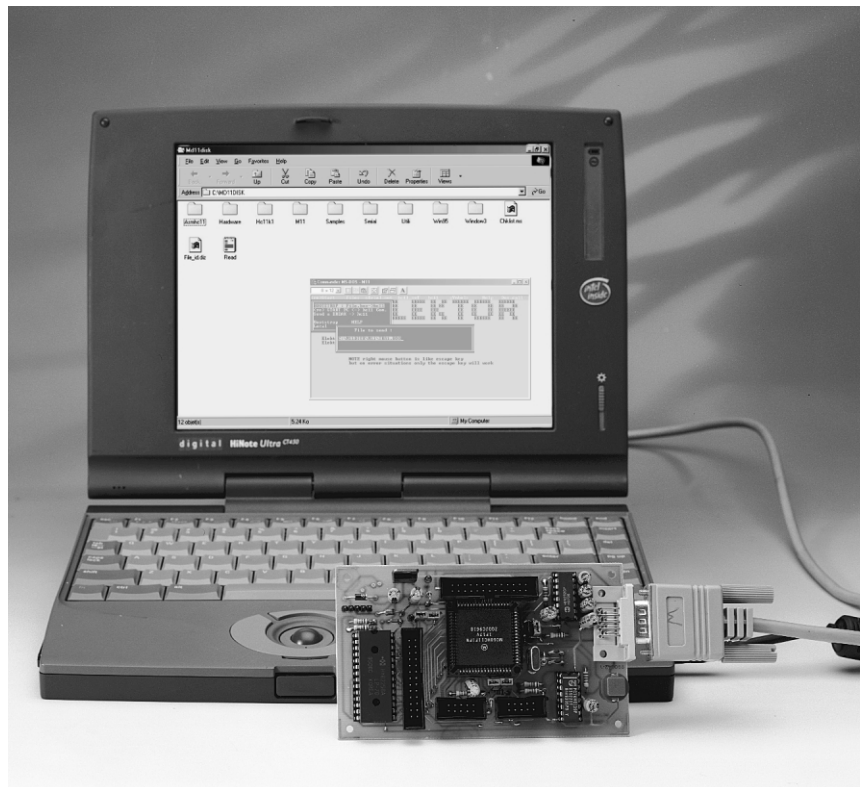
Newcomers may wonder about the purpose of a development system. Many of our readers, we know, attempt to make microcontroller circuits perform certain functions. For this you need certain 'tools', which, depending on the case, equate to hardware and/or software. The two are 'married' in what is commonly referred to as a *development system*.

Over the pas few years, it has transpired from publications in the electronics press that the 'F1' version of Motorola's 68HC11 family is the most popular. Despite the fact that it comes in a 68-pin case, the HC11F1 is actually one of the least expensive devices in

the HC11 family. Moreover, it is easily obtainable.

From its design and features, it is obvious that the 68HC11 is squarely aimed at 'heavy' applications. The device integrates an address decoder and a complete bus (that is, one that is not multiplexed). These features allow system developers (like you, we hope) to just add RAM and EPROM and yet have a working controller. It should be noted, however, that 'single chip' functioning is only possible if you keep RAM size down to 1 kBytes, and EEPROM size, to 512 bytes.

The 68HC11 may be crystal clocked at up to 20 MHz. Typically, at 16 MHz it

achieves a memory access speed of 250 ns. Of course, it is possible to use lower clock frequencies if so desired, and access to peripheral components may be 'slowed down' with respect to memory access (see *Clock Stretching* in the 68HC11F1 documentation, Ref. [1]).

THE ELECTRONICS

Although the basic design of the board is relatively simple, it will be adequate for those of you who limit themselves to assembly code programming. The architecture may also be extended if necessary.

As you can see from the circuit diagram in **Figure 2**, the board is built around five integrated circuits: a 68HC11F1, a MAX232, a 74HC139, a memory device and a voltage regulator.

In addition to what has been said about the **68HC11F1** in the previous paragraphs, we now get down to business by showing the internal structure and device pinout in **Figure 1**. Of all 17 members of the 68HC11 family, the HC11F1 version is without doubt the best known and best selling device in amateur circles.

Looking at the internal structure you will agree with us that the 68HC11 is a fairly complex device. At about 10 years of age the HC11F1 is one of the younger members of the family. It is also one of the most powerful because multiplexing is not used, plus a 1-kByte RAM and a 512-byte EEPROM are available on the chip. The chip is supplied in a 68-pin case. The HC11F1 has no internal ROM or EPROM. Instead, it can take many different external memory devices of impressive capacities. Finally, the HC11F1 micro contains no fewer than four ports, a number of which can be used in bidirectional mode.

The second essential part in the circuit diagram is the **MAX232**. This integrated circuit allows the board to be connected to a serial PC port (RS232). As will be discussed further on, the 'M11' software allows you to download object code to the HC11, place breakpoints, use single-step program execution, and display the contents of program variables, registers, and lots more.

The third circuit is an address decoder type **74HC139**, of which only one half is used. It supplies the $\overline{RD}$ and $\overline{WR}$ signals that make the bus compatible with 'Intel' components. Independent *ReaD* and *WRite* signals are required for EEPROM type 2864.

In the circuit diagram, the memory socket is shown to hold a 32-kByte RAM type 62256. However, this is just an example as the socket can accommodate one of eight different memory
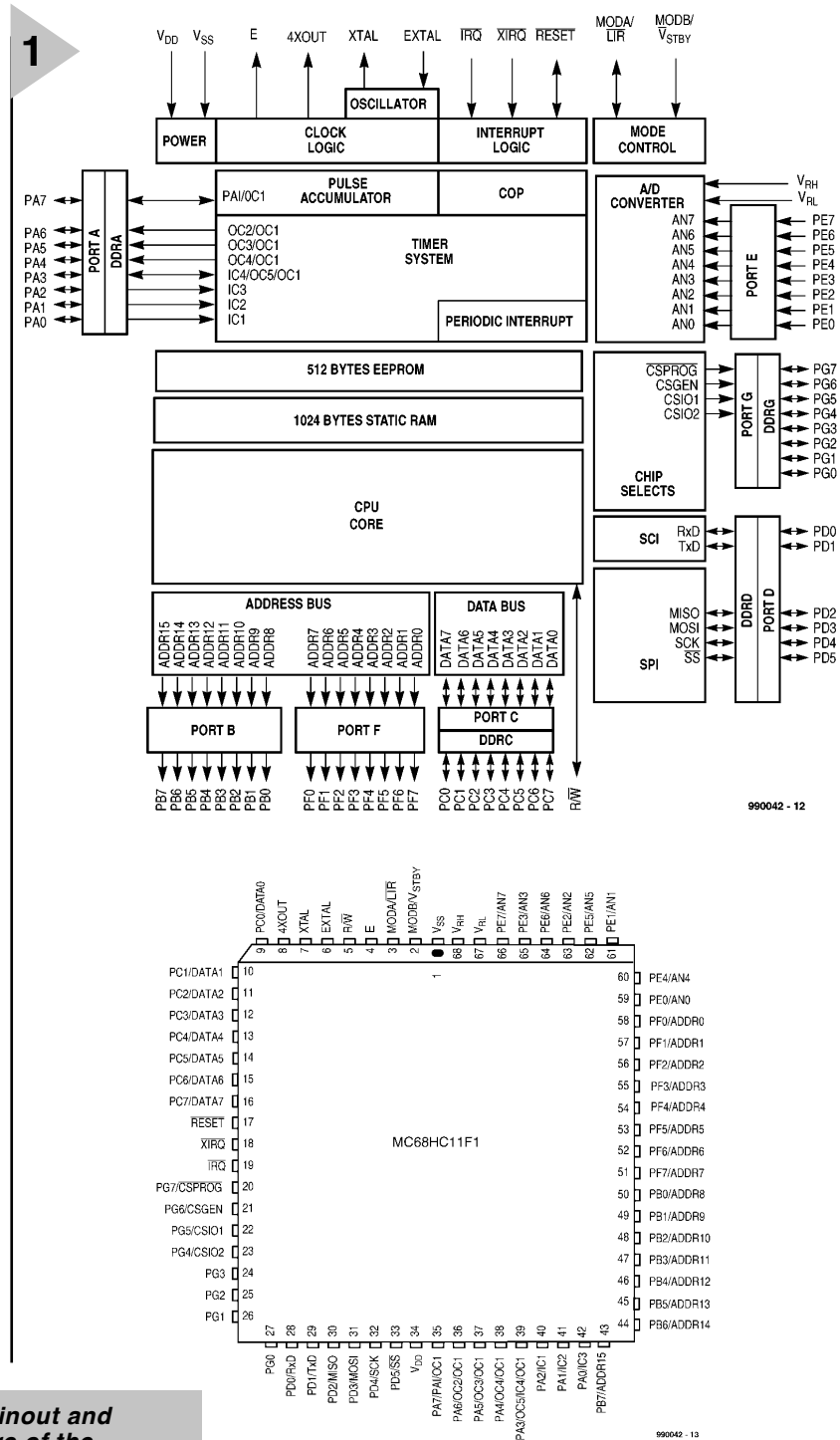
Figure 1. Pinout and architecture of the 68HC11F1 microcontroller (courtesy Motorola).

devices. Arguably, this feature is one of the most attractive of the development system described in this article.

In fact, the memory device can be any RAM, EEPROM or EPROM, as long as the relevant chip comes in a 28-pin DIL case. This includes EPROMs with a capacity of 8, 16, 32 and 64 kBytes, RAMs with a capacity of 8 or 32 kBytes, as well as 8 or 32-kByte EEPROMs. The selection of the memory device inserted in the socket reserved for IC2 is by means of two jumpers on connector K7 (see **Table 1** further on).

The software called 'M11' may be familiar to those of you who have built the circuit described in Ref. [2]. For the present development system, it is has been modified to allow external EEPROMs to be programmed. This is useful for those of you lacking the means to program EPROMs. In that case, a 28C64 EEPROM is a perfect substitute.

That brings us to a new and unexpected application of this project: the board may be used to program EEPROMs intended for use in other circuits. Once the external EEPROM has been programmed with code, it may be protected against (accidental) writing by pulling the $\overline{WR}$ jumper from connector K7. Another, smaller, EEPROM is available inside the HC11 for semi-permanent storage of configuration data.

2

Figure 2. The core of the circuit diagram is obviously the 68-pin microcontroller type 68HC11F1.
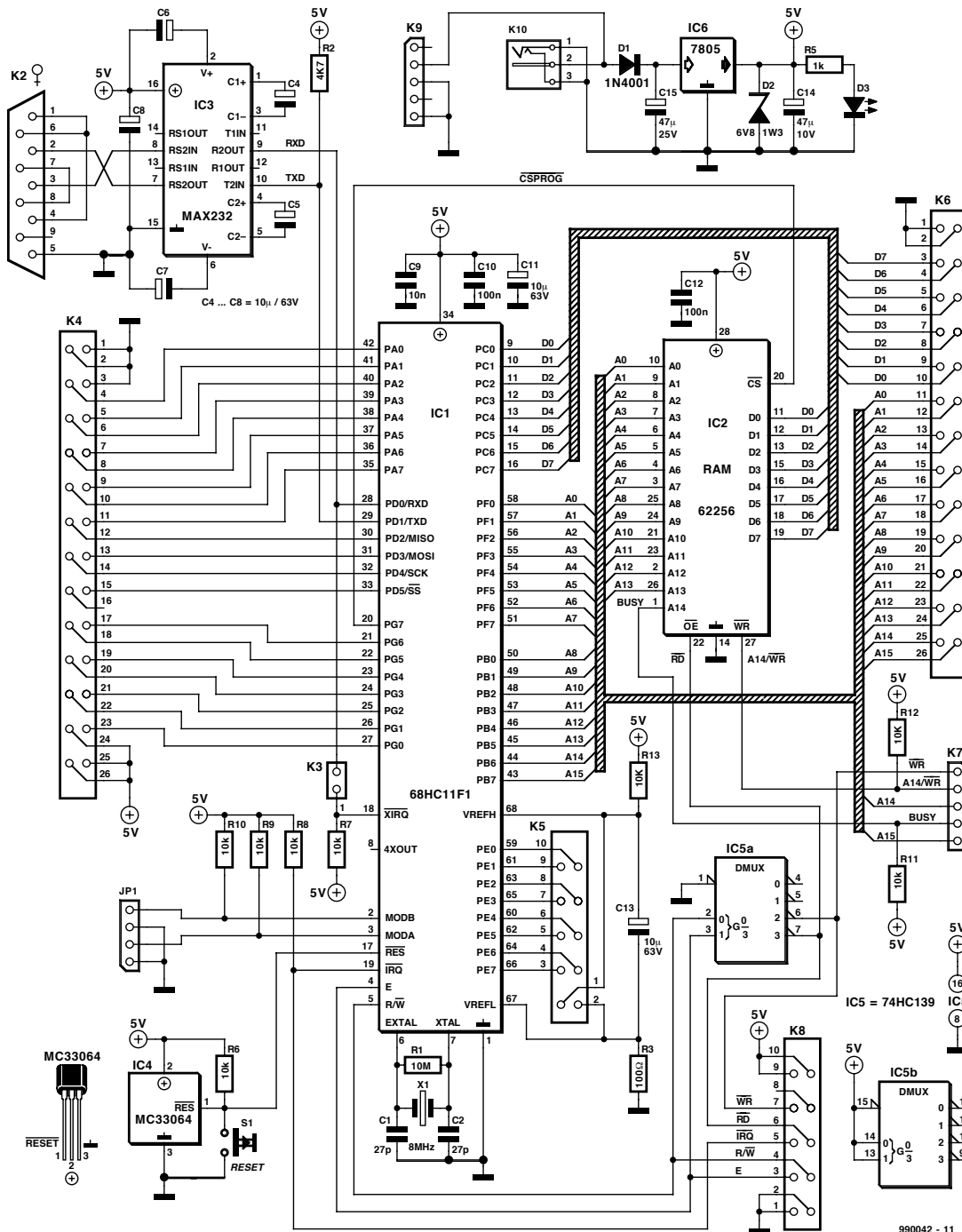
Note: to be able to program an external EPROM you will need an EPROM programmer. The M11 utility is only capable of programming the EPROM area available in certain HC11 microcontrollers (not including the HC11F1).

Given the fact that HC11F1 has an internal RAM of 1 kBytes, most turnkey, standalone or otherwise 'closed' applications will take an EPROM or an EEPROM in position IC2. By contrast, when software is under development, it will often be required to load many different versions of the program, set breakpoints, etc. This obviously requires a RAM or EEPROM installed in position IC2.

While debugging software, the RAM allows you to work much faster when the amount of code starts to exceed a few kilobytes. Finally, although the EEPROM is the most expensive memory option for a given memory capacity, it is the most versatile. For most, if not all, amateur use, an 8-kByte EEPROM will have enough capacity.

The memory device is decoded by the $\overline{CSPROG}$ signal supplied by the PG7 pin on the controller. After a reset, $\overline{CSPROG}$ covers a default area of 64 kBytes. This does not conflict with other memory elements because these, being internal to the 68HC11, are given

priority. It is, therefore, perfectly possible to keep working without reconfiguring $\overline{CSPROG}$, provided you limit yourself to a memory area with the same size as the memory device used. By contrast, $\overline{CSPROG}$ has to be modified if you install a second memory device on the extension connector.

If you are working with the M11 software, you don't normally have to worry about the memory technology used. M11 automatically provides the type of write action required for the memory map declared in the menu *hc11_set*. There may be 'border' conflicts, however, if code extends over two memory elements of different technology and occupying adjacent

Figure 3. Component mounting plan and track layouts of the double-sided through-plated board supplied through the Readers Services.

address areas (rare though). If that happens, a small area of about 40 bytes should be left free before the border. In this way, you force the assembler to separate the relevant memory blocks in the S19 output file. This may look like a shortcoming, but remember that there are few utilities capable of simul-taneously programming several mem-ory devices of different technology.

Note: the M11 software assumes that RAM is available in areas not declared in the 'hc11_set' menu.

THE CONNECTORS

The card has four extension connec-tors: K4, K5, K6 and K8. In most cases, only K4 and K5 will be used.

K4 is for the free binary input out-put lines of ports A, D and G, while K5 covers port E. The inputs that may be used to convey analogue signals are separately routed (K5 is very close to port E). The references used by the
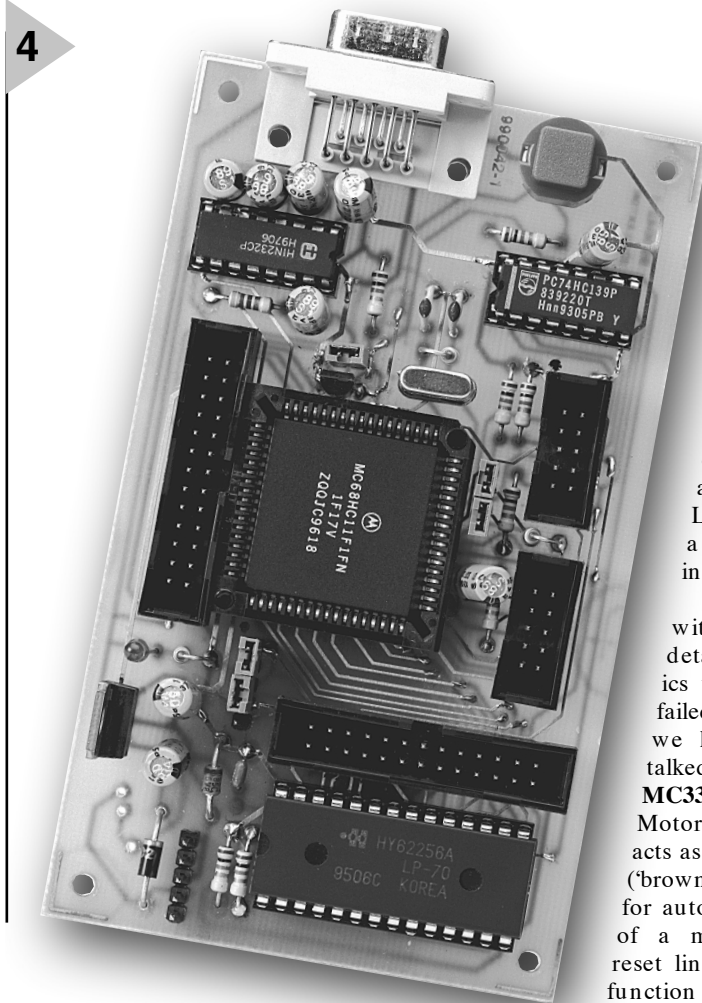
Figure 4. Our finished prototype of the board. Look carefully at the positioning of IC1 and notice the bevelled edge on the IC socket.

analogue-to-digital converter (ADC) are also routed to K5. Short links should be used to keep noise levels as low as possible.

Each port of the HC11 may be employed in binary input/output mode and/or it may be coupled to a special interface: timer for port A, serial links for port D, and ADC for port E.

For certain applications, you may want to have an external memory or an interface component designed to acts as a link to a microcontroller bus. In that case, it is possible to use the bus signals gathered on K6, and the bus control signals gathered on K8.

Next, there are two further options: use 'traditional' address decoding, or pick from port G those pins that could be employed as 'chip selects'. The latter solution is to be preferred but only if the relevant port G pins are not in use already. The advantages: less wiring (but a bit more software) and the possibility to 'slow down' certain pins to access slow components or units (for example, a bus-connected LCD module).

The last component to be discussed is voltage regulator IC6 in the power supply section. The raw input voltage is supplied by a mains adaptor with an output voltage between 9 and 12 V dc. Connector K9 on the board (close to the mains adaptor socket) allows the board to work on a separate regulated supply voltage of about 8 V 'borrowed' from the host circuit (if available, of course). LED D3 acts as a power on/off indicator.

Those of you with an eye for detail in schematics will not have failed to notice that we have not yet talked about IC4, an **MC33064** from Motorola. This chip acts as a low-voltage ('brown-out') detector for automatic control of a microprocessor reset line. Its obvious function is that of a watchdog, monitoring the supply voltage level and supplying a reset pulse when the level drops below 4.6 V.

CONSTRUCTION

The artwork (copper track layouts and component mounting plan) of the PCB designed for the 68HC11 development board is shown in **Figure 3**. This circuit board is double-sided, through-plated and available ready-made through our Readers Services.

The board is populated in the usual manner. In this case, however, it is recommended to start by fitting the 68-way square socket for the microcontroller (position IC1). The orientation of the socket on the board is indicated by **a bevelled edge**. Make absolutely sure you get this right because any error at this point has dramatic consequences — the MC68HC11 will be fitted the wrong way around; it may be damaged and the circuit will not work at all. An even bigger problem, however, is to remove the PLCC socket from the board. Doing so may cause damage to the board and eventually force you to have wreck the whole socket with pliers and then extract the pins one by one. All without damaging the board, of course!

Capacitors C9 and C10 are preferably fitted at the solder side of the board. Next, fit the low-profile parts.

Next up are the sockets for IC2, IC3 and IC5. If you envisage frequently changing the memory device in position IC2 you may consider fitting a ZIF socket (zero insertion force) or, if that is too expensive, a socket with turned pins.

Construction is finished by fitting the boxheaders or pinheaders and the RS232 connector.

Soldering

With proper soldering, a board like this one should last more than 20 years. If you are not an experienced electronics constructor then you are likely to apply too much solder.

The solder iron used should have a fine tip and a power rating of about 30 watts. First heat the solder pad and the wire in it, then apply a little solder, withdraw the solder wire, and keep heating the joint for 2 to 3 seconds. Then remove the tip and let the solder solidify.

THE SERIAL CABLE

The cable that links the development system to the PC is of the 'straight-through' type, that is, it has no crossed wires. Although such cables are available ready-made at very low prices, you may still want to make one yourself from a length of flatcable terminated with a 9-way sub-D socket (female connector) at one side, and a 9-way sub-D plug (male connector) at the other. Care should be taken in the fitting of the connectors to the cable — look very attentively at the pin numbers printed on the connector parts before clamping them together in a vise. The whole exercise of making your own cable may not be worthwhile, however, as the two connectors may cost more than a ready-made shielded cable bought from a computer store.

This brings us to *the moment supreme*, the long-awaited moment when the home constructor can put a feather in his cap, or the dreaded moment when nothing seems to work at all.

FIRST TEST

The steps below describe how the board is taken into service.
✓ Power the board *without* the expensive parts fitted (HC11, memory, but *with* the MAX232 installed in position IC3. Verify the presence of + 5 V at all the relevant IC socket pins, and the presence of the RS232 voltages (approx. + 10 V and –10 V at pins 2 and 6 respectively of the MAX232). Obviously, the power indicator LED

should light.

✓ Switch off the supply voltage, then install the HC11 and (if desired) the memory device in their sockets. Fit two jumpers on pinheader JP1. This causes the MODA and MODB lines to be pulled to ground and selects *bootstrap* mode (in *test* mode, the Talker performs this mode selection by means of software).

✓ Fit a jumper on K3, allowing a Talker to employ $\overline{XIRQ}$ (of extra interest for interrupt-based programming).

✓ Install the cable between the RS232 port on the PC and the development board (9-way serial link). Power up the board.

✓ Launch the M11 software on the PC.

✓ Look at the two 'set' menus:

✓ In the *serial_set* menu, most default values will be okay. You may have to make some adjustments to account for the quartz crystal frequency used on the board.

✓ In the *hc11_set* menu, select FE00 to FFFF for the HC11 EEPROM, and FF0 to FF0 (yes, a dummy address) for the HC11 EPROM because there is no such device! Finally, enter 8000 to 9FFF for the 28Cxx if IC2 is a 28C64. Alternatively, stick to the default range 8000 to 8000 if IC2 is a RAM.

✓ From the Menu select (re)start=> file.BOO -> HC11.

✓ Type c:\m11disk\m11\TKF1XTST.BOO and then press Return. This selects a Talker for the F1 version employing $\overline{XIRQ}$ and auto-switching in *test* mode. When the software prompts you to do so, reset the system by actuating push-button S1, then press Return. If M11 displays 'Waits for echo' or 'bad echo', read the relevant help notes (in this menu and in *serial_set*). Everything is all right if M11 displays a single line with 'OK'. In fact, M11 echoes the reply received from the PC. The Talker is then safely stored in RAM.

✓ Launch the communication with the Talker (9,600 baud if the HC11 runs at 8 MHz). If everything works as it should, five memory areas will be displayed (upper left) plus the HC11 registers (upper right). The HC11 functions normally. If external memory is not available, switch off the system, insert the memory device in its socket and restart. M11 stores the system configuration in M11.TXT.

✓ Now verify access to the external memory. Display $8000 in a memory address area (upper left), click in the corresponding data area, and write 40 41 42 43 44 45 46 47. If this works, the equivalent ASCII characters @ A B C D E F G will be displayed to the right. Click on '*' next to 8000 and verify that this character string is not repeated anywhere else. By using a second area, also make sure this

sequence does not appear at addresses 8100, 8200, 8400, 8800, 9000, A000 or C000. At each of these addresses, perform a write test using different values every time, and at least once the value $55 followed by $AA.

Testing large-scale memory access to external memory

Load the program Seconde.S19 (path: C:\m11disk\samples\F1) and its symbols file Seconde.LST. Display the program variables by writing 100 101 103 105 107 into the memory address areas. Once you have loaded the symbols file, the names of the variables should appear. Click on PC, enter 8000, click on '*' at the PC side (disassembling). Next, verify that the program is present in the external memory (quit by ESC).

Start a stopwatch or look at your watch and do 'Go'. From time to time, press the 'R' key to refresh the display without halting the program. In this way, you can verify that the SEC and SEC_BCD variables 'tick' like seconds.

Testing write access to Port A and Port G.

Load the program LedF1.S19 (path: C:\m11disk\samples\F1) and its symbol file LedF1.LST. Display the values 1000 and 1002 in the address area. The *porta* and *portb* symbols will be displayed. Next, proceed with the same sequence as discussed with the previous program. If you have an oscilloscope or a frequency meter available, then the level changes on the port outputs are easily monitored. If you do not have such equipment, insert a breakpoint into the loop and measure the port line levels with a voltmeter.

THE M11
CONTROL SOFTWARE

The M11 software utility and the program examples discussed above are supplied on a diskette with order code **996005-1**. This diskette may be ordered from our Readers Services. Alternatively, it may be included in a kit if you buy the project from a kit supplier advertising in this magazine (Stippler, Viewcom or C-I Electronics). The basic operation and features of the M11 package are discussed below.

Although it runs under DOS, M11 does provide a graphic user interface based on (simple) windows and the mouse. If you are conversant with the Windows Explorer, it is possible to link the assembler and editor application programs to source code files, the editor to list files, and M11 itself to a working directory. With these links in operation, you can work much faster: program assembly/editing by right-clicking on the mouse.

Follow this procedure to configure

it all in Windows Explorer.

After double clicking on an LST (list) file, Windows Explorer will ask you which application it should open. Select Notepad from the list. Windows Explorer will create a link which is memorised and used for all files with the extension .LST. Do the same with one file having the extension .ASC, and one .S19 file if you want to be able to read these, too. Next, for ASC files you will want to see the assembler pop up besides Notepad (the choice between editor and assembler is as follows: **right-click the mouse > menu local**).

This may be a bit complex. Menu Display -> option -> file types (locate ASC and place checkmark) -> modify -> new -> write 'assembler' in the title line (above) and

```
c:\m11disk\asmhc11.EXE %1; x
```

in the command line.

Finally, provide a link between M11 and its working directory, for example, *samples*. Windows Explorer offers a very simple method to create such a link: click on M11.EXE and drag it into the folder called *samples*. Next, right-click on M11.EXE to access its Properties. Select and set the following: close on exit, full screen, low idle sensitivity, no screen saver.

Note: if you set up a link between M11 and its working directory, you have to leave the 'working directory' line in the Program Properties sheet empty. In this way, the default working directory is the one you are in when running Windows Explorer.

Once set up, all this will also function from the desktop, so you can choose any utility you want (see the directory \M11DISK\Win95 for more information).

The files with the extension .LST help you locate assembly errors (if there are any, of course). An LST file also provides a table of all symbols defined for the assembly process.

The assembler will only work if two files, *offset.ASC* and *codes.ASC*, are present in the current directory (the one you are working in).

M11 does not work on a networked computer and has to be used with the 'full screen' option enabled in Windows 95.

Note: the full-screen mode still allows you to return to Windows — simply press Alt-TAB.

On slow PCs, or when using high data speeds on the RS232 port (say, 38k4), the serial link may get overloaded or it may not function at all, despite the 'low idle sensitivity' you set earlier on in the program Properties. If this happens, restart the PC and select MS-DOS mode. Similarly, because the special operation of the watchdog results in a rather high workload on the PC, it does not work very well under Windows 95.

M11 downloads a small utility called *Talker* into the HC11 when the micro runs in *bootstrap* mode. The serial link is employed for this function. Next, the Talker provides access to the memories and registers, which, in turn, allows M11 to create 'debug' functions and program all of the HC11's internal memories (RAM, EEPROM or EPROM).

In principle, M11 can adapt itself to all versions of the HC11 microcontroller (except the D versions which do not have sufficient internal memory), in all functional modes and with very modest cabling requirements. For additional information on the M11 software, please consult the article '68HC11 Emulator' (Ref. [2]).

Updating M11 from an older version

Those of you who have the previous version of M11 (version found on project disk supplied for '68HC11 Emulator') should note that new Talkers are included (versions for K1/K4 versions of the 68HC11, these are 84/80-pin devices with 1-Mbyte memory addressing capability and on-chip PWM).

PROGRAMMING EXTERNAL EPROMs

Some help pages have been modified from the February 1997 article (Ref. [2]).

A few simple suggestions are given for experiments with the watchdog.

Once updated from an older version, M11 will pop up an error message 'size problem with file M11.TXT'. Actually, this configuration file now contains the external EEPROM addresses, so it will be too long. M11 will propose to rewrite a default version of this file. You have to accept this and review your system configuration in the two *set* menus.

Note: each project in its own folder will have its very own file M11.TXT. Each of these may need to be reviewed.

The board described here employs a Talker called TKF1XTST.BOO (jumper fitted between $\overline{\text{XIRQ}}$ and $\overline{\text{RXD}}$). Although this Talker switches to *test* mode to provide access to external memories, it does not change the vectors set up by the *bootstrap* mode.

The communication speed used with this Talker is 19,200 bits/s when using a 16 MHz quartz crystal (or 9,600 at 8 MHz, 14,400 at 12 MHz).

Given its architecture, the HC11F1 will run with most Talkers for the A1 and E1 versions of the micro. So, if you require a fairly special Talker not specifically geared to the F1, it is usually possible to pick one for the E1 or A1 and always have the instruction LDS #$3FF at the start of your program.

MISCELLANEOUS

Quartz crystal frequency for the HC11

At a clock of 16 MHz, the memory access frequency is 4 MHz or 250 ns. As a rule of thumb, the access time specified for the memory device has to be about half the memory access time. So, a specification of 125 ns is required when a 16-MHz crystal is used, and 250 ns (maximum) when you clock the micro at 8 MHz. This last value allows many currently available memory devices to be used.

Note: often, memory devices continue to work just fine at access times well below their specification. Their current consumption may increase beyond specified limits, however, raising the device temperature and causing erratic behaviour of the memory.

Using the assembler to program EEPROMs for other applications

It is possible to generate an S19 file containing *any* code and employ the assembler's FCB, FDB or FCC directives at the end of an ORG statement which is offset with respect to the base address of the memory device on the HC11 board.

(990042-1)

References:

[1] Motorola: *Microcontrollers 68HC11, 68HC12, 68HC16 & MPC500 Families* (CD-ROM)

[2] 68HC11 Emulator, *Elektor Electronics* February 1997, pages 22-27.

Table 1. Jumper functions (K7)

The table below shows the jumper positions on 5-way pinheader K7 as a function of the memory device type fitted in position IC2. Pin 1 of the pinheader is marked by a bevelled edge.

	1–2	2–3	3–4	4–5
EPROM 8 K (2764)	–	–	–	–
EPROM 16 K (27128)	–	–	–	–
EPROM 32 K (27256)	–	X	–	–
EPROM 64 K (27512)	–	X	–	X
EEPROM 8 K (28C64)	X	–	–	–
EEPROM 32 K (28256)	X	–	X	–
RAM 8 K (6264)	X	–	X	–
RAM 32 K (62256)	X	–	X	–

X = jumper fitted
– = no jumper fitted

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Evaluation System for 80C166 (parts 1 and 2)
March & April 1999, 990028

Errors in Components List

IC9 and IC10 should be type 74HC573, not HCT573.

C1 and C2 should be 10µF 16 tantalum bead.

R20 should be an 8-way SIL array, value 4kΩ7.

The crystal frequency mentioned with C18, C19 should be 40MHz, not 100MHz.

IC3, MAX690 should be listed as MAX690 (BATT), not MAX690 (I/O)

Miscellaneous

Pull-up resistors R1-R4 need not be mounted with the serial channels.

When a 40-MHz crystal is used for X1, it may oscillate at the fundamental frequency (13.333 MHz) instead of the third overtone. If this happens the system baudrate will not be correct and the serial communication will fail to work as described. Sug-gested solutions are (1) to use a 40-MHz crystal oscillator module, (2) use a different 40-MHz crystal or (3) adapt the value of C18 and C19 until oscillation occurs at the third overtone.

The battery may be a 3.6-V NiCd type, but note that this is very slowly charged by the MAX690. A better solution is to use a Lithium battery.

The circuit diagram does not make it clear how the serial connection is made via D9. Pins 2 and 3 of this connector should not be connected TxD1 and RxD1 (i.e. the TTL side of the MAX232), but to the other (RS232) side of the MAX232. Pin 2 of D9 is therefore connected to pin 2BC via

990028-11

bus K1. Likewise pin 3 of D9 is connected to pin 3BC via bus K1. The correct connections are shown in the illustration.

Development System for 68HC11F1
June 1999, 990042

JP1 is a simple jumper. In the text and parts list, it is erroneously referred to as K3.

Part K3 is a 4-way SIL pinheader. In the text and parts list, it is erroneously referred to as JP1

R4 is erroneously listed as a SIL array with a value of 10kΩ. The PCB however only accommodates eight discrete resistors. These are numbered R4 and R6-R12. Resistor R13 in the circuit diagram equals R4 on the PCB.

PC-Controlled Model Railway: EEDTS Pro
June 1999, 990082-2

On page 60, the text references t S3, D1 and S4 should read S2, D2 and S1 respectively.

DEVELOPMENT SYSTEM FOR 68HC11F1
June 1999, 990042

JP1 is a simple jumper. In the text and parts list, it is erroneously referred to as K3. Part K3 is a 4-way SIL pin-header. In the text and parts list, it is erroneously referred to as JP1.

R4 is erroneously listed as a SIL array with a value of 10kΩ. The PCB however only accommodates eight discrete resistors. These are numbered R4 and R6-R12. Resistor R13 in the circuit diagram equals R4 on the PCB.

Titan 2000

Part 5: half-bridging two single amplifiers

In the introduction to Part 1 it was stated that the Titan 2000 could deliver up to 2000 watts of 'music power', a term for which there is no standard definition but which is still used in emerging markets. Moreover, without elaboration, this statement is rather misleading, since the reader will by now have realized that the single amplifier cannot possibly provide this power. That can be attained only when two single Titan amplifiers are linked in a half-bridge circuit. The true power, that is, the product of the r.m.s. voltage across the loudspeaker and the r.m.s current flowing into the loudspeaker, is then 1.6 kilowatts into a 4-ohm loudspeaker.

Design by T. Giesberts
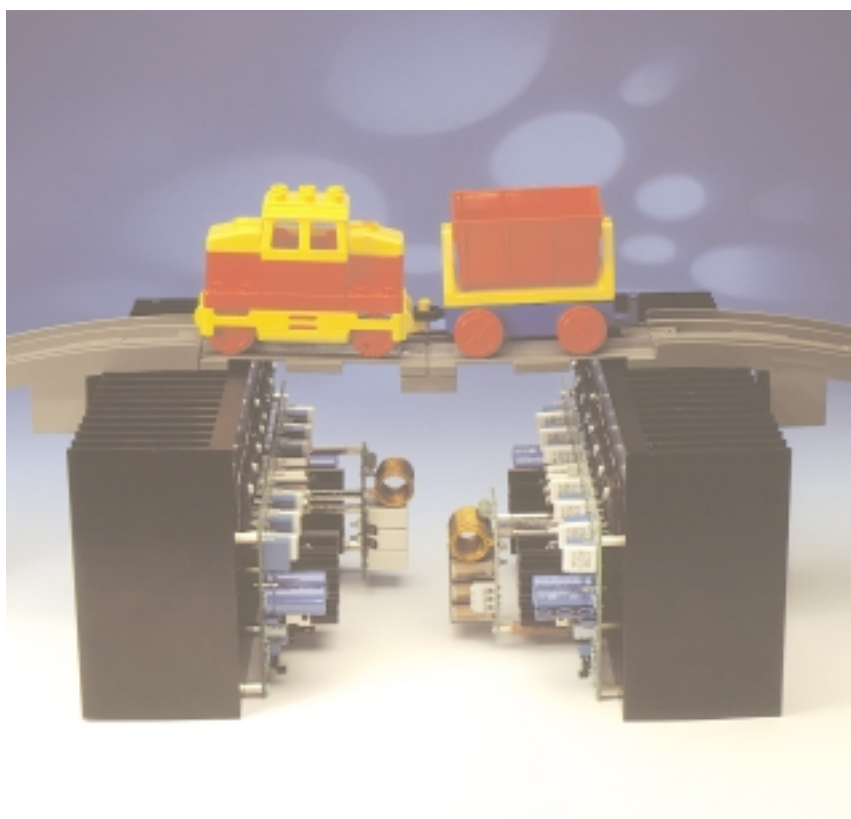
BRIDGING: PROS AND CONS

Bridging, a technique that became fashionable in the 1950s, is a way of connecting two single output amplifiers (valve, transistor, BJT, MOSFET, push-pull, complementary) so that they together control the passage of an alternating current through the loudspeaker. This article describes what is strictly a half-bridge configuration, a term not often used in audio electronics. When audio engineers speak of bridge mode, they mean the full-bridge mode in which four amplifiers are used.

In early transistor audio power amplifiers, bridging was a means of achieving what in the 1960s were called public-address power levels as high as,
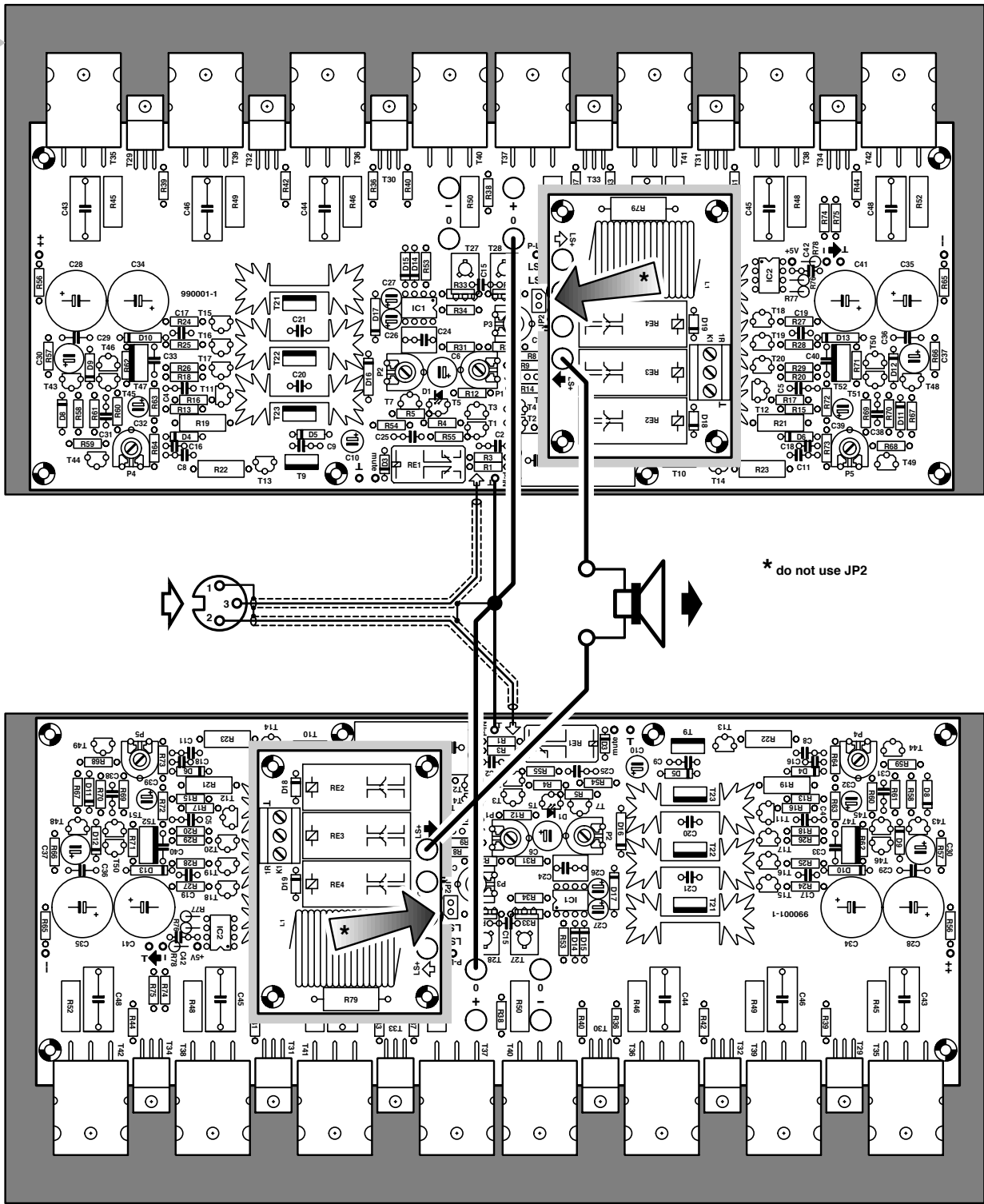
1

* do not use JP2

990001 - 4 - 11

Figure 17. The interlinking required to form a half-bridge amplifier from two single Titan 2000 units. Note that the resulting balanced input may be reconverted to an unbalanced one with the Brangé design (Balanced/unbalanced converters for audio signals) published in the March 1998 issue of this magazine. The PCB for that design (Order no. 980026) is still stocked.

say 50–80 W into 8 Ω. Such power levels were then way beyond of what the voltage ratings of output transistors would permit.

Bridging is considered by many to be a good thing, since it automatically provides a balanced input (drive). However, opponents will quickly point out that it halves output damping, doubles the circuitry and virtually cancels even-order harmonics created in the amplifier.

Opponents also claim that bridging amplifiers is tedious and requires too much space. It is, however, not simple either to design a single amplifier with the same power output and the requisite power supply. A single 2 kW amplifier requires a symmetrical supply voltage of ± 130 V, that is, a total of

260 V. The power supply for this would be quite a design. And where would a designer find the drivers and output transistors for this? Advocates point out that bridging amplifiers have the advantage of requiring a relatively low supply voltage for fairly high output powers.

Bridging just about doubles the rated output power of the single amplifier. Again, opponents point out that loudness does not only depend on
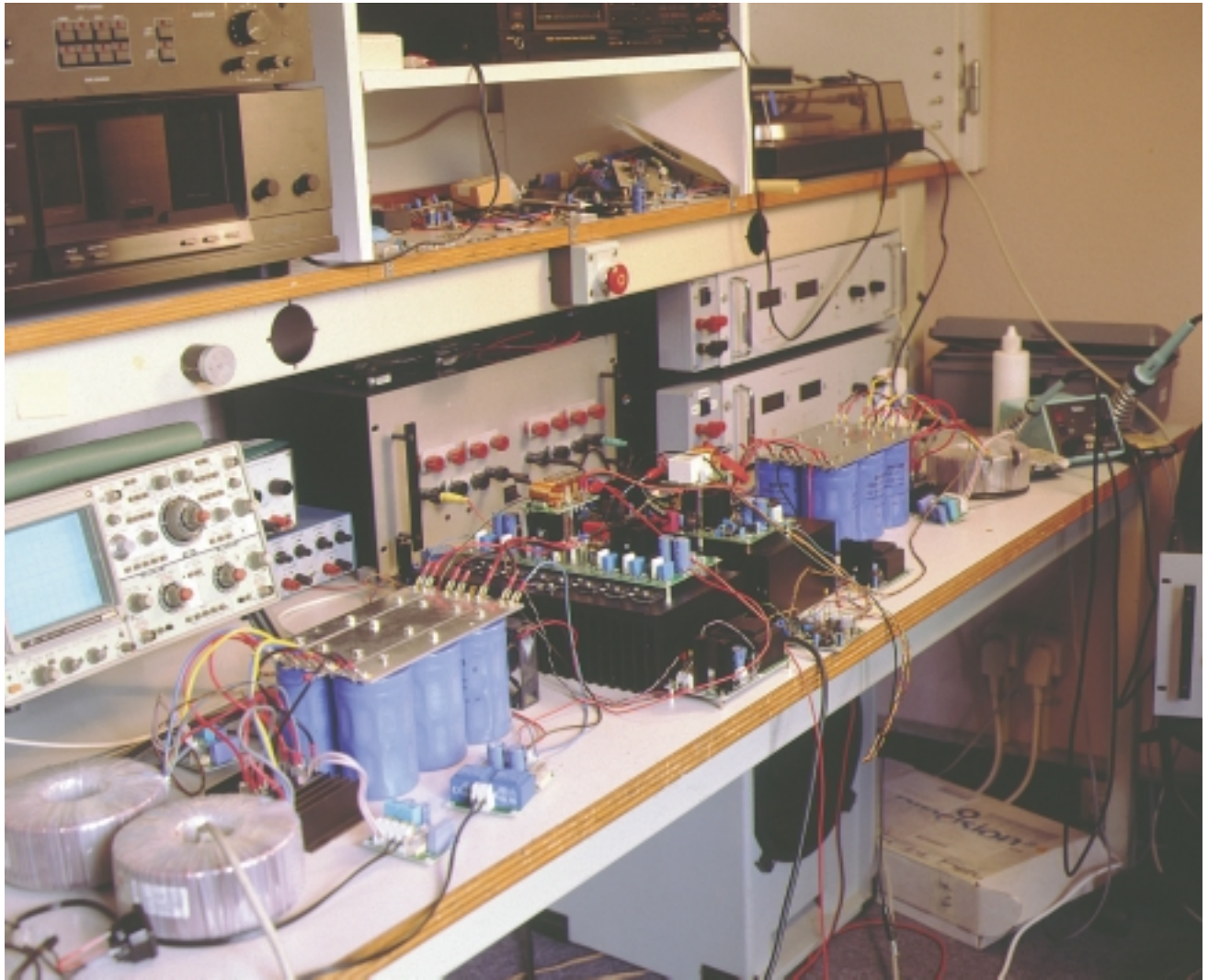
the amplifier, but also on the loud-speaker. Bear in mind, they say, that just changing a loudspeaker with a sensitivity of, say, 90 dB$_{SPL}$ per watt per metre to one with a sensitivity of 93 dB$_{SPL}$ per watt per metre is equal to doubling the amplifier power rating.

Clearly, bridging two amplifiers is a mixture of good and bad audio engineering and sonics.

INTERCONNECTING

It is, of course, necessary that two completed single Titan 2000 amplifiers are available, each with its own power supply. It should then be possible to simply interlink the earths of the two units, use the inputs as a common balanced input, and connect the loudspeaker between terminals LS+ on the two amplifier. However, a few matters must be seen to first.

Owing to the requisite stability, it is imperative that the two amplifiers are juxtaposed with the space between them not exceeding 5 cm (2 in). They should, of course, be housed in a com-

mon enclosure.

The inter-wiring is shown in **Fig-ure 17**. Make sure that the power supplies are switched off and that the smoothing capacitors have been discharged before any work is carried out.

Start by interlinking the negative supply lines (terminals 0) with insulated 40/02 mm wire. Remove the insulation at the centre of the length of wire since this will become the central earthing point for the new (balanced) input. Link the ⊥ terminals on both boards to the new central earth with 24/02 mm insulated wire.

Connect the loudspeaker terminals to the LS+ terminals on the two boards with 40/02 mm insulated wire.

Link pins 2 and 3 of the XLR connector to the input terminals on the boards with two-core screened cable. Solder the screening braid to pin 1 of the XLR connector and to the new central earthing point.

Finally, on both boards remove jumper JP$_2$ from the relevant pin strip.

FINALLY

When all interconnections between the boards as outlined have been made, the single amplifiers form a half-bridge amplifier. If all work has been carried out as described, there should be no problems.

In the design stages, network R$_9$-P$_1$, inserted into the circuit with pin jumper JP$_1$ (see Part 1), was considered necessary for common-mode suppression. However, during the testing of the prototype, the network was found to be superfluous. It may be retained if the half-bridge amplifier is to be used with a second half-bridge amplifier for stereo purposes, when it may be used to equalize the amplifications of the two half-bridge amplifiers.

[990001]

Parameters

With a supply voltage of ± 70 V (quiescent ± 72 V) and a quiescent current of 200–400 mA

Input sensitivity	*2.1 V r.m.s.*
Input impedance	*87 kΩ*
True power output for 0.1% THD	*950 W into 8 Ω; 1.5 kW into 4 Ω*
True power output for 1% THD	*1 kW into 8 Ω; 1.6 kW into 4Ω*
Power bandwidth	*1.5 Hz – 220 kHz*
Slew limiting	*170 V µs⁻¹*
Signal+ noise-to-noise ratio (at 1 W into 8 Ω)	*97 dB (A-weighted*
	93 dB (B= 22 kHz)
Total harmonic distortion (B= 80 kHz)	
at 1 kHz	*0.0033% (1 W into 8 Ω)*
	0.002% (700 W into 8 Ω)
	0.0047% (1 W into 4 Ω)
	0.006% (700 W into 4 Ω)
at 20 kHz	*0.015% (700 W into 8 Ω)*
	0.038% (1200 W into 4 Ω)
Intermodulation distortion	
(50 Hz:7 kHz = 4:1)	*0.0025% (1 W into 8 Ω)*
	0.0095% (500 W into 8 Ω)
	0.004% (1 W into 4 Ω)
	0.017% (500 W into 4 Ω)
Dynamic intermodulation distortion	
(square wave of 3.15 kHz and	*0.0038% (1 W into 8 Ω)*
sine wave of 15 kHz)	*0.0043% (700 W into 8 Ω)*
	0.005% (1 W into 4 Ω)
	0.0076% (1200 W into 4 Ω)
Damping (with 8 Ω load)	*≥ 350 (at 1 kHz)*
	≥ 150 (at 20 kHz)
Open-loop amplification	*×8600*
Open-loop bandwidth	*53 kHz*
Open-loop output impedance	*3.2 Ω*
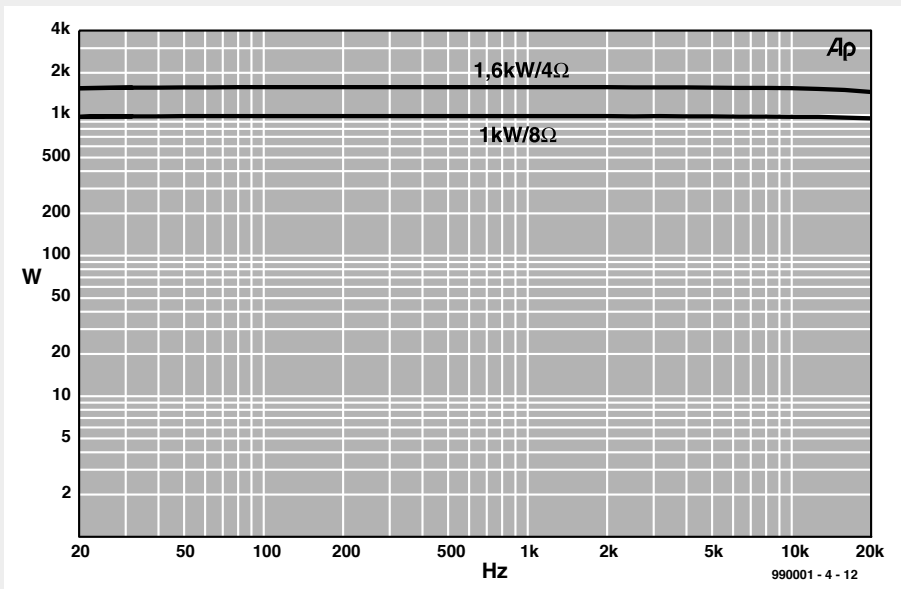
A comparison of these parameters with the specifications given in Part 4 ((May 1999 issue) show that they are gener-
ally in line. In fact, the intermodulation distortion figures are slightly better. Because of this, no new curves are given
here other than power output (1 kW into 8 Ω and 1.6 kW into 4 Ω) vs frequency characteristics for 1 per cent total har-
monic distortion.

During listening tests, it was not possible to judge the half-bridge amplifier at full volume, simply because there were
no loudspeakers available that can handle this power output. However, up to 200 W true power output, the half-bridge
amplifier sounds exactly the same as the single amplifier. Instrument test figures show no reason to think that the per-
formance at higher output powers will be degraded.

Bridge-tied-load (BTL) amplifier Type TDA8552

high voltage on chip

The TDA8552 from Philips Semiconductors is a two-channel audio power amplifier that provides an output power of 2×1.4 W into an 8 Ω load with a 5 V power supply. The circuit contains two BTL power amplifiers, two digital volume controls and standby/mute logic. Volume and balance of the amplifiers are controlled via two digital input pins which can be driven by simple push-buttons or by a microcontroller. It is contained in a 20-pin small outline package (SOP).

A Philips Semiconductors Application

INTRODUCTION
The TDA8552T is a Bridge-Tied Load (BTL) audio power amplifier capable of delivering 2×1.4 W into an 8 Ω load at a Total Harmonic Distortion (THD) of 10% operating from a 5 V power supply.

The gain of the amplifier can be set by the digital volume control. The gain in the maximum volume setting is 20 dB (low gain) or 30 dB (high gain). The maximum gain is selected via the

PINNING

SYMBOL	PIN	DESCRIPTION
GND1	1	ground 1, substrate/leadframe
OUT2+	2	positive loudspeaker terminal output channel 2
V_{DD1}	3	supply voltage 1
HPS	4	digital input for headphone sensing
MODE	5	digital trinary input for mode selection (standby, mute and operating)
UP/DOWN1	6	digital trinary input for volume control channel 1
UP/DOWN2	7	digital trinary input for volume control channel 2
V_{DD2}	8	supply voltage 2
OUT2-	9	negative loudspeaker terminal output channel 2
GND2	10	ground 2, substrate/leadframe
GND3	11	ground 3, substrate/leadframe
OUT1+	12	positive loudspeaker terminal output channel 1
V_{DD3}	13	supply voltage 3
GAINSEL	14	digital input for gain selection
IN2	15	audio input channel 2
SVR	16	half supply voltage, decoupling ripple rejection
IN1	17	audio input channel 1
V_{DD4}	18	supply voltage 4
OUT1-	19	negative loudspeaker terminal output channel 1
GND4	20	ground 4, substrate/leadframe

1

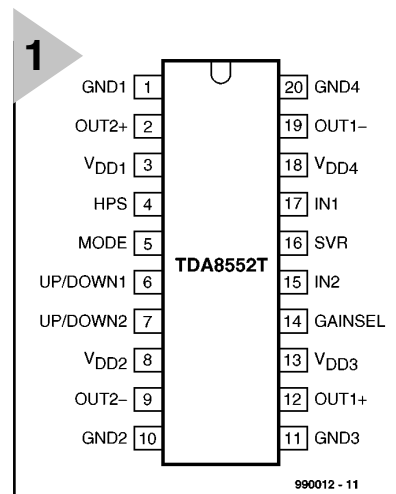|		
GND1	1	20	GND4
OUT2+	2	19	OUT1–
V_{DD1}	3	18	V_{DD4}
HPS	4	17	IN1
MODE	5	16	SVR
UP/DOWN1	6	15	IN2
UP/DOWN2	7	14	GAINSEL
V_{DD2}	8	13	V_{DD3}
OUT2–	9	12	OUT1+
GND2	10	11	GND3

TDA8552T

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Figure 1. Pin configuration of the TDA8552.

gain selection pin.

The headphone sense input (HPS) can be used to detect if a headphone is plugged into the jack (socket) connector. If a headphone is plugged into the socket, the amplifier switches from the BTL to the Single Ended (SE) mode and the BTL loudspeakers are switched off. This also results in a reduction of quiescent current drain.

The device can be switched to the standby condition, mute condition, or normal operating condition via the MODE pin.

The device is protected by an internal thermal shutdown protection mechanism.

POWER AMPLIFIER

The power amplifier is a bridge-tied load amplifier with a complementary CMOS output stage. The total voltage loss for both output power MOS transistors is within 1 V and with a 5 V supply and an 8 Z loudspeaker an output power of 1.4 W can be delivered. The total gain of the power amplifier can be set at 20 dB or 30 dB via the gain selection pin.

GAIN SELECTION

The gain selection can be used for a fixed gain setting, depending on the application. The gain selection pin

must be hard-wired to ground (20 dB) or to V_{DD} (30 dB). Gain selection during operation is not advised since switching is not guaranteed plop-free.

INPUT ATTENUATOR

The volume control operates as a digitally controlled input attenuator between the audio input pin and the power amplifier. In the maximum volume control setting, the attenuation is 0 dB and in the minimum volume control setting, the typical attenuation is 80 dB. The attenuation can be set in 64 steps via the UP/DOWN pin.

Both attenuators for channels 1 and 2 are separated from each other and are controlled via their own UP/DOWN pin.

Balance control can be arranged by applying UP/DOWN pulses to pins 6 and 7.

VOLUME CONTROL

Each attenuator is controlled via its UP/DOWN pin (trinary input):

• Floating UP/DOWN pin: volume remains unchanged
• Negative pulses: decreasing volume
• Positive pulses: increasing volume

Each pulse on the UP/DOWN pin results

in a change in gain of 80/64 = 1.25 dB (typical value).

In the basic application, the UP/DOWN pin is switched to ground or V_{DD} by a double push-button. When the supply voltage is connected initially, after a complete removal of the supply, the initial state of the volume control is an attenuation of 40 dB (low volume), so the gain of the total amplifier is –20 dB in the low gain setting or –10 dB in the high gain setting. After power-up, some positive pulses have to be applied to the UP/DOWN pin for turning up to listening volume.

AUTO REPEAT

If the UP/DOWN pin is low or high for the wait time (t_{wait} in seconds) (one of the keys is pressed), the device starts making up or down pulses by itself with a frequency given by $1/t_{rep}$ repeat function). The wait time and the repeat frequency are set with an internal *RC* oscillator with an accuracy of ± 10%.

VOLUME SETTINGS IN STANDBY MODE

When the device is switched via the MODE select pin to the mute or standby condition, the volume control attenuation setting keeps its value, provided that the voltage on the V_{DD} pin

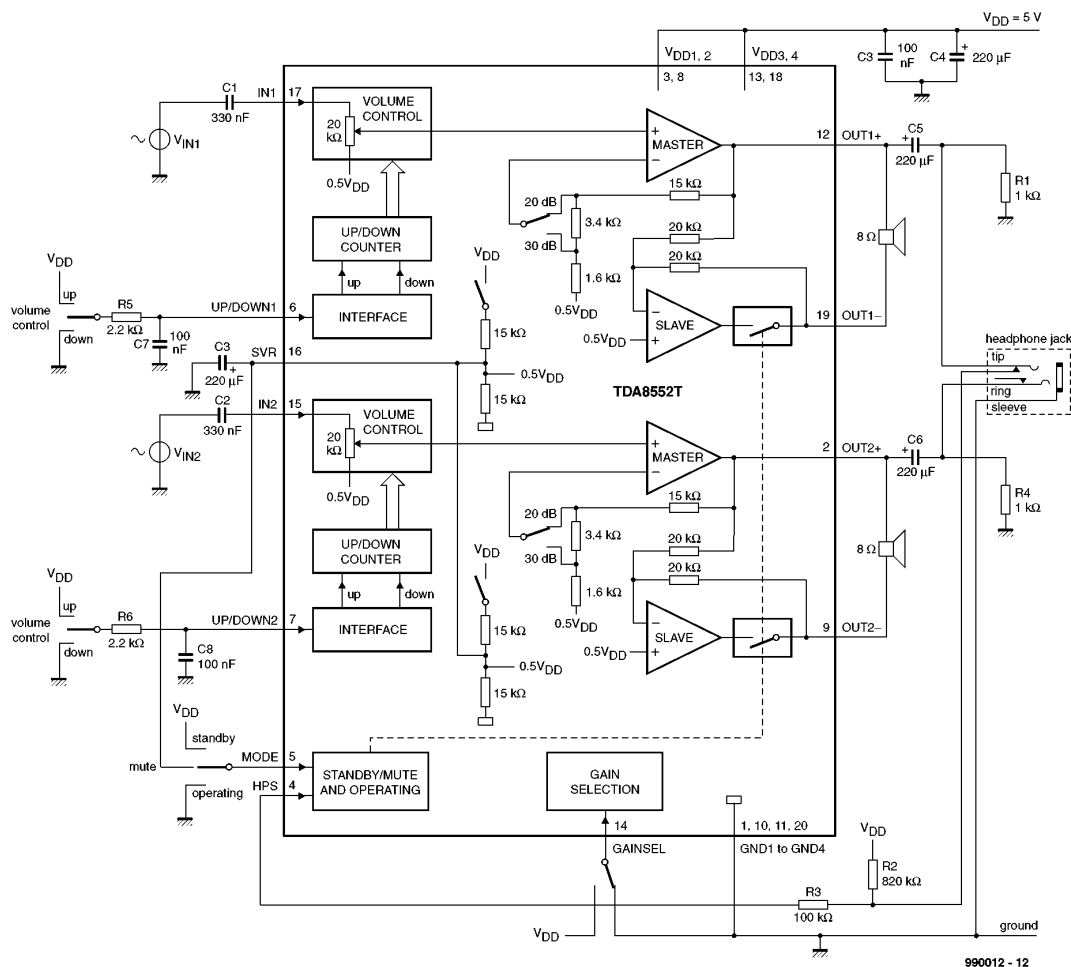**Figure 2. Test and application diagram of the TDA8552.**

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Table 1. DC characteristics.

SYMBOL	PARAMETER	CONDITIONS	MIN.	MAX.	UNIT
V_{DD}	supply voltage	operating	-0.3	+ 5.5	V
V_i	input voltage		-0.3	V_{DD} + 0.3	V
I_{ORM}	repetitive peak output current		-	1	A
T_{stg}	storage temperature		-55	+ 150	°C
T_{amb}	operating ambient temperature		-40	+ 85	°C
V_{sc}	AC and DC short-circuit safe voltage		-	5.5	V
P_{tot}	maximum power dissipation	SO20	-	2.2	W
		SSOP20	-	1.1	W

does not fall below the minimum supply voltage.

After the device has been switched back to the operating mode, the previous volume setting is maintained. In the standby mode, the volume setting is maintained as long as the minimum supply voltage is available. The current drain is very low: about 1 μA (typical).

In battery-operated applications, the volume setting can be maintained during battery exchange if there is a supply capacitor available.

MODE SELECTION

The device is in the standby mode (with a very low current drain) if the voltage at the MODE pin is between V_{DD} and V_{DD}–0.5 V. At a mode select voltage of ≤ 0.5 V, the amplifier is fully operational.

In the range between 1 V and V_{DD}–1 V, the amplifier is in the mute condition. The mute condition is useful for using it as a 'fast mute'; in this mode, the output signal is suppressed, while the volume setting retains its value.

It is advisable to keep the device in the mute condition while the input capacitor is being charged. This can be achieved by holding the MODE pin at a level of 0.5 V_{DD}, or by waiting about 100 ms before applying the first volume-UP pulses.

HEADPHONE SENSING

A headphone can be connected to the amplifier by using a coupling capacitor for each channel. The common ground pin of the headphone is connected to the ground of the amplifier—see **Figure 2**. When the HPS pin is used as illustrated in this diagram, the TDA8552T detects if a headphone plug is inserted into the socket.

When no headphone is plugged in, the voltage level at the HPS pin will remain low. A voltage ≤ V_{DD}–1 V at the HPS pin will keep the device in the BTL mode: the loudspeakers can then be operational. If the HPS pin is not connected, the device will remain in the BTL mode.

When a headphone is plugged into the socket, the voltage at the HPS pin will be set to V_{DD}. The device then switches to the SE mode, which means that the slave power amplifiers at outputs OUT1 and OUT2 are switched to the standby mode. This results in floating outputs: the loudspeaker signal is then attenuated by about 80 dB and only the headphone can operate.

One of the benefit of this system is that the loudspeaker current does not flow thorough the socket switch, which could cause some power loss. Te other benefit is that the quiescent current is reduced when the headphone plug is inserted into the socket.

[990012]

Table 2. AC characteristics.

| SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
P_o	output power	THD = 10%; R_L = 8 Ω	1.0	1.4	-	W
		THD = 10%; R_L = 16 Ω	-	0,8	-	W
		THD = 0,5%; R_L = 8 Ω	0.6	1.0	-	W
		THD = 0,5%; R_L = 16 Ω	-	0.6	-	W
THD	total harmonic distortion	P_o = 0.1 W; note 1	-	0.15	0.4	%
		P_o = 0.5 W; note 1	-	0.1	0.3	%
$V_{o(n)}$	noise output voltage	GAINSEL. = 0 V; note 2	-	60	100	μV
		GAINSEL. = V_{DD}; note 2	-	100	-	μV
SVRR	supply voltage ripple rejection	note 3	50	55	-	dB
Vi(max)	a maximum input voltage	THD = 1%;	-	-	1.75	V
		G_v = –50 to 0 dB				
a_{sup}	channel suppression	V_{HPS} = V_{DD}; note 4	70	80	-	dB
a_{cs}	channel separation		50	-	-	dB

Notes

1. Volume setting at maximum.

2. The noise output voltage is measured at the output in a frequency band from 20 Hz to 20 kHz (unweighted), R_{source} = 0 Ω.

3. Supply voltage ripple rejection is measured at the output, with a source impedance of R_{source} = 0 Ω at the input. The ripple voltage is a sine wave with a frequency of 1 kHz and an amplitude of 100 mV (RMS) is applied to the positive supply rail, gain select pin is LOW (0 V).

4. Channel suppression is measured at the output with a source impedance of R_{source} = 0 Ω at the input and a frequency of 1 kHz. The output level in the operating single-ended channel (OUT+) is set at 1 V (RMS).

PC-controlled model railway: EEDTS Pro

Part 1: the hardware

The Elektor Electronics Digital Train System published in this magazine in 1989/90 was warmly received by the regrettably declining core of model train enthusiasts. Now, almost ten years later, much of the electronics technology has changed and it is, therefore, felt necessary to update a number of key elements of that system. One of the more important changes is the prominent role given to the personal computer (PC). This development also opens the way to a number of new applications.
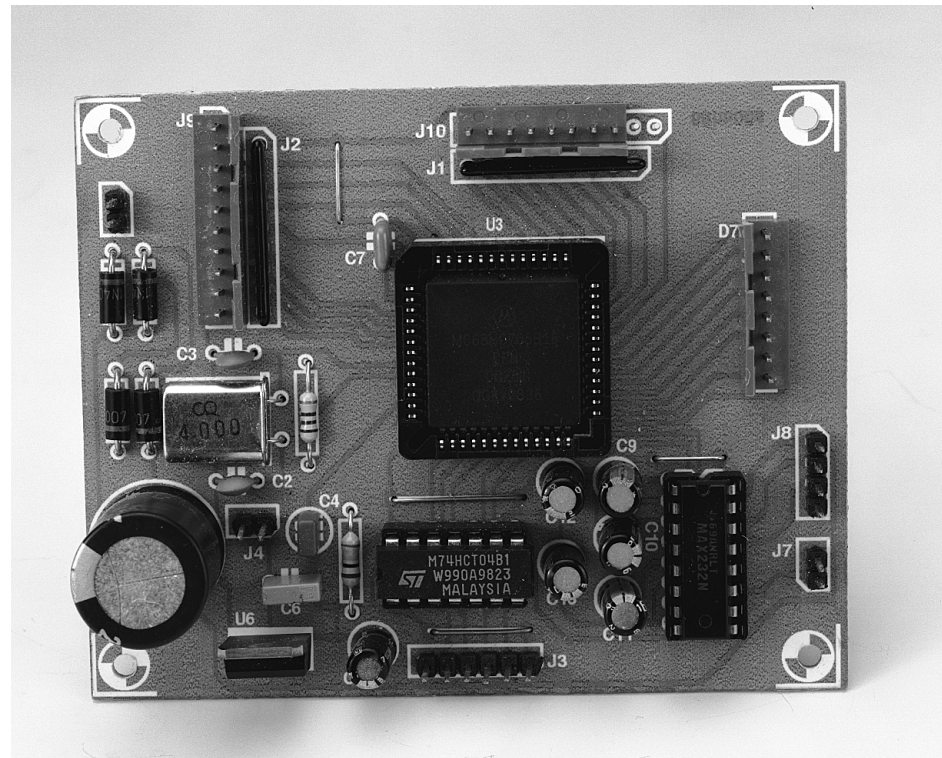
INTRODUCTION

Modern electronics and model train systems get along well together. This is primarily because most electronic components have become so small that they can easily be fitted into a locomotive, turnouts (points; USA: switches), or signals to give these some intelligence. Developments in this field were the subject of an article in the May 1999 issue of this magazine.

The snag with many proprietary model train systems is that they are fairly expensive. The EEDTS of yesteryear showed that it is possible to build a much less expensive alternative. As a bonus, the system is transparent so that it can be incorporated into proprietary systems: locomotives fitted with a Märklin decoder may be readily integrated into the EEDTS. This means that the user has all the freedom he/she wants in building up a model train system.

EEDTS Pro continues along these lines by replacing those components of the earlier system that have become a little long in the tooth, technologically speaking.

In the design, great importance was attached to retaining or obtaining compatibility with existing systems. The result of this policy is that many items of the EEDTS, such as the booster, and the decoders for turnouts (switches) and signals, can be used with EEDTS
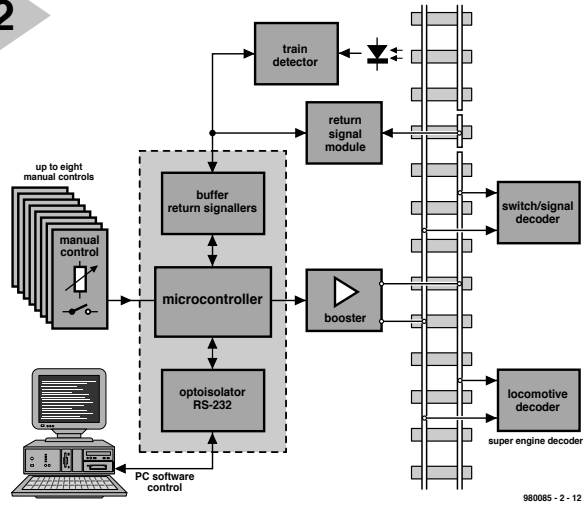
Design by S van de Vries

Figure 1. Block diagram of a simple configuration of a model railway.
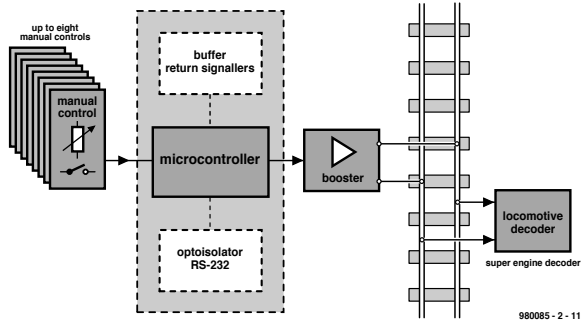
Figure 2. Block diagram of a digital model railway based on the new controller. All requisite functions are readily found in the circuit diagram in Figure 3.

Figure 3. Circuit diagram of the controller which is based on a Motorola microprocessor.

Pro. Proprietary products from, say, Märklin and Lenz may also be used without any problems.

A new aspect of EEDTS Pro is the integration in the control unit of a PC interface to enable monitoring and guiding the trains on the track on the VDU (Visual Display Unit) of the PC. The software necessary for this will be discussed and dissected in next
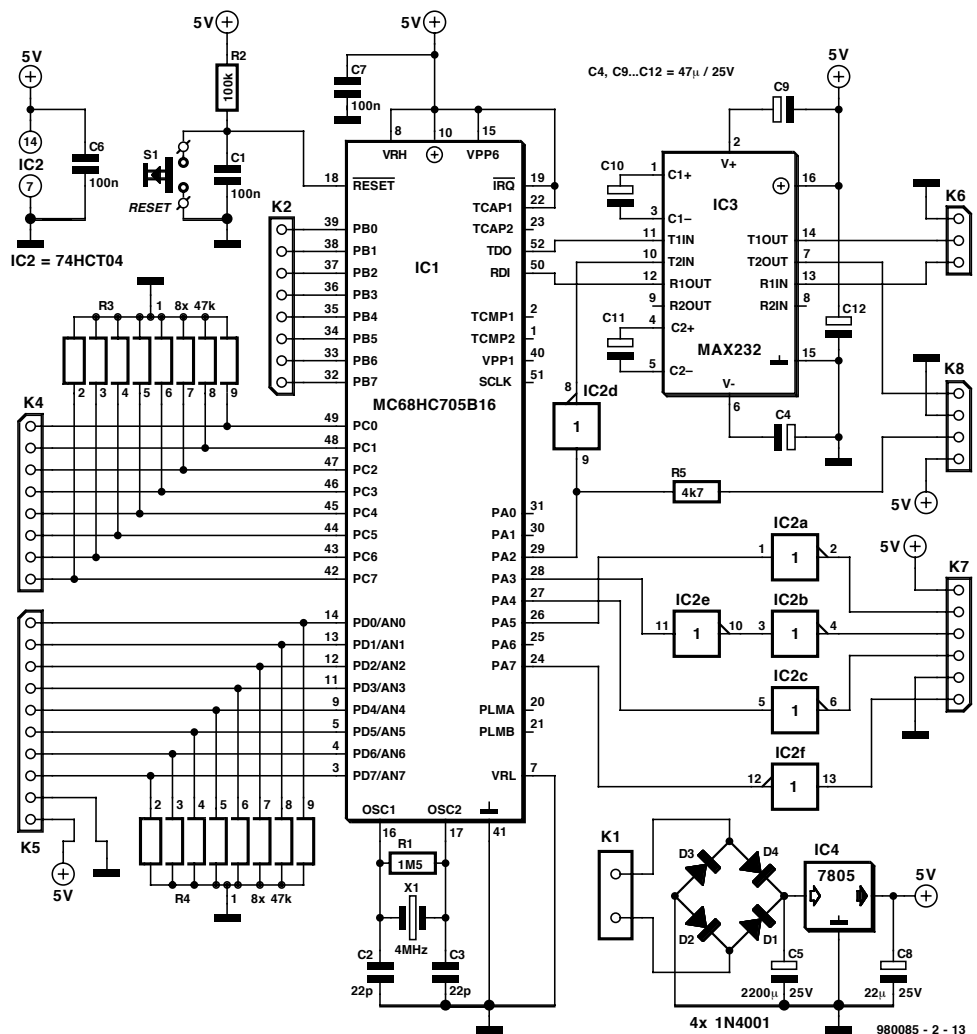
month's instalment.

BLOCK DIAGRAMS

A block diagram of the control system of the model train is shown in **Figure 1**. The solid parallel lines between the booster and the locomotive decoder represent the rails. At the left of these lines is the control section which, in the stand-alone configuration, consists of

4

K8 on printed circuit board

R1 12k

S1

D1 8V2

K1

Re1 12V

D3

D2 1N4148

S2

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Parts lists BOOSTER INTERFACE:

Resistors:
R_1 = 12 kΩ

Semiconductors:
D_1 = zener, 8.2 V, 400 mW
D_2 = LED
D_3 = 1N4148

Miscellaneous:
S_1 = push-button switch with break contact
S_2 = push button switch with make contact
K_1 = 5≠ pole DIN connector for board mounting
Re_1 = relay, 12 V

5

to K7 on printed circuit board
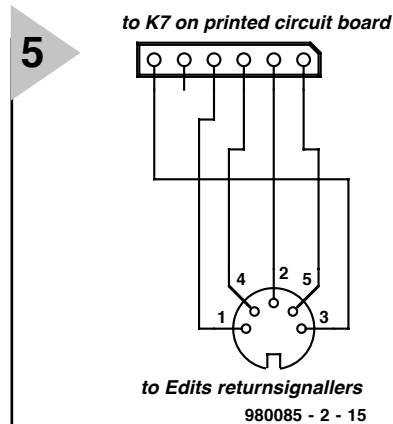

to Edits returnsignallers
980085 - 2 - 15

Figure 5. Simple adaptor to enable the monitor units to be connected to the 6-way header on the controller.

up to eight functional controls, a microcontroller, and booster. The digital control signals are conveyed to the track via the booster. With this simple system up to eight independent locomotives can be used on the track.
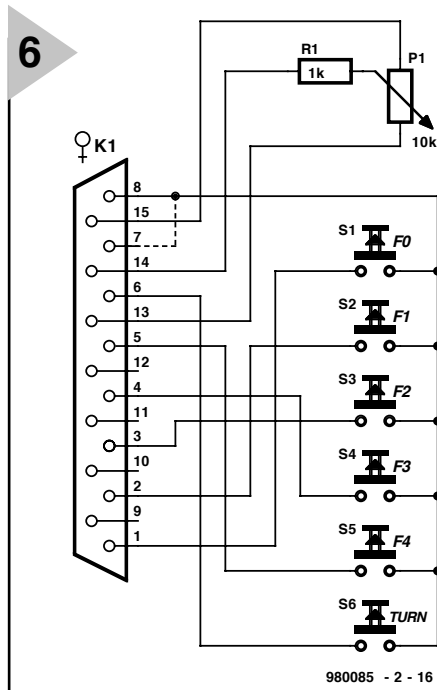
The engine decoder converts the digital signals on the track into analogue control voltages for the engine. This may take two forms: one that is fitted in the locomotive during manufacture, or one that is built at home and then fitted into the locomotive.

Those readers who use Märklin equipment may note that the EEDTS Pro supports the C80, C81, and C90 decoders (old Motorola format) and the 60901, 60902, C95, and C95/2 decoders (new Motorola format). Compatible decoders and Lenz decoders are also supported.

A new super engine decoder is in

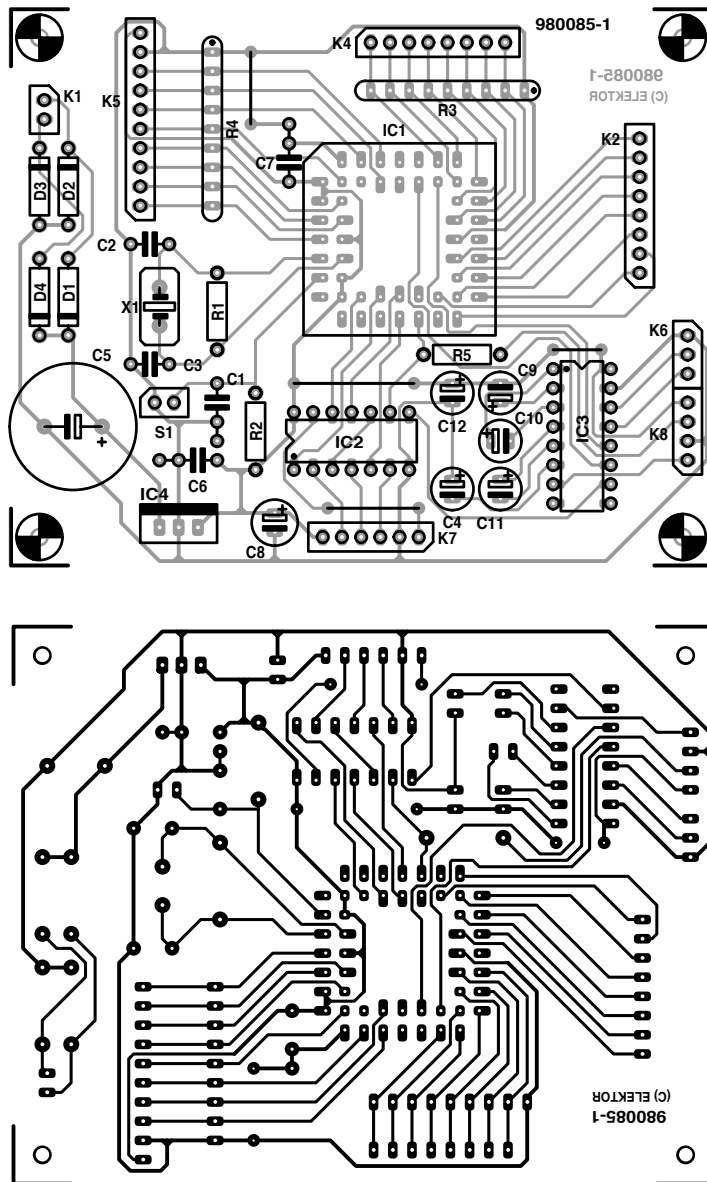the design stages and will be described in a future issue of this magazine.

An engine decoder designed under the old Motorola format enables controlling the speed and direction of a locomotive. There is also a possibility for a third function, such as interior lighting, a steam whistle or decoupling.

EXPANSION

The simple system in Figure 1 may be expanded to that in **Figure 2**. This is based on the units within the dashed lines: buffer monitors, microcontroller, optoisolator, and optional serial link to a computer. A computer may add a new dimension to the model railway.

A computer-controlled track can work only, however, if there is feedback from it to the computer. After

6

R1 1k

P1 10k

K1

S1 *F0*
S2 *F1*
S3 *F2*
S4 *F3*
S5 *F4*
S6 *TURN*

manual control

DB15 connector:

pin 1: to K4 pin 1
pin 2: to K4 pin 2
pin 3: to K4 pin 3
pin 4: to K4 pin 4
pin 5: to K4 pin 5
pin 6: to K4 pin 6
pin 7: to K4 pin 7
pin 8: to K2 pin 1...8 (up to eight manual controls)

pin 13: GND (e.g., K5 pin 9)
pin 14: to K5 pin 1...8 (up to eight manual controls)
pin 15: + 5V (e.g., K5 pin 10)

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Parts lists MANUAL CONTROL

Resistors:
R_1 = 1 kΩ
P_1 = 10 kΩ

Miscellaneous:
S_1–S_6 = push button switch with make contact
K_1 = 15-pole female sub-D connector

all, even a computer is able to control the traffic on the track only if it knows where the trains are. Note that control is purely on the basis of the location of trains. The position of turnouts (switches) or signals is not monitored. They can, however, be controlled by software, as is to be expected from a sophisticated system like EEDTS Pro.

7

980085-1

K1
K5
K4
K3
IC1
R4
C7
K2
D3 D2
C2
D4 D1
X1
R1
K6
C5 C3
R5
C1
C9
K8
S1
R2
IC2
C10
IC3
IC4
C6
C12
C4 C11
C8
K7

Parts lists CONTROLLER

Resistors:
$R_1 = 1.5\ M\Omega$
$R_2 = 100\ k\Omega$
$R_3, R_4 =$ SIL array, $8 \times 47\ k\Omega$
$R_5 = 4.7\ k\Omega$

Capacitors:
$C_1, C_6, C_7 = 0.1\ \mu F$
$C_2, C_3 = 0.022\ \mu F$
$C_4, C_9-C_{12} = 47\ \mu F$, 25 V, radial
$C_5 = 2200\ \mu F$, 25 V, radial
$C_8 = 22\ \mu F$, 25 V, radial

Semiconductors:
$D_1-D_4 = $ 1N4001

Integrated circuits:
$IC_1 = $ MC68HC705B16 (available via
  our Reader Services, Order No
  986518-1
$IC_2 = $ 74HCT04
$IC_3 = $ MAX232
$IC_4 = $ 7805

Miscellaneous:
$X_1 = $ crystal, 4 MHz
$S_1 = $ single-pole switch with make con-
  tact
$K_1 = $ 2-fold SIL header
$K_2, K_4 = $ 8-pole SIL header
$K_5 = $ 10-pole SIL header
$K_6 = $ 3-pole SIL header
$K_7 = $ 6-pole SIL header
$K_8 = $ 4-pole SIL header
Diskette, 3.5", with Windows software,
  available via our Reader Services,
  Order No 986027-1a+ b

Figure 7. Printed-circuit board for the controller.

DECODERS

A digitally controlled railway needs a number of decoders to control the traffic on the tracks. Apart from the moving trains, there are turnouts (switches), signals, decoupling units, and other components that need to be controlled remotely. Then there are functions within the carriages, such as lighting, which it is nice to control remotely.

Furthermore, it is important for the controller to know where the various trains are located. Monitor units signal the passing of a train and send back this information to the controller.

(MICRO)CONTROLLER

The circuit diagram of the new controller is shown in **Figure 3**. It is compact and wholly digital and is based on a Motorola microcontroller, IC_1. This device provides eight analogue inputs, AN_0-AN_7, three 8-bit I/O ports, a serial connector and an integral oscillator.

The oscillator is configured, together with crystal X_1, resistor R_1, and capacitors C_2, C_3 to generate a clock signal of 4 MHz.

The controller communicates with the computer via an RS232 interface, which is why the level of the TTL sig-

nals is raised by IC_3 to enable the signals to be used by the RS232 interface. The computer is linked to header K_6 via one of communication ports COM_1-COM_4.

The single 5 V supply line is converted to the requisite voltage levels by capacitors C_4, C_9, C_{10}, and C_{11}.

There is a second serial output on header K_8, which carries four signals: the booster signal at RS232 and TTL level; the supply voltage; and earth.

The pulse-shaped signals are put on to the track via the booster. The circuit diagram of the requisite booster interface is shown in **Figure 4**. The interface
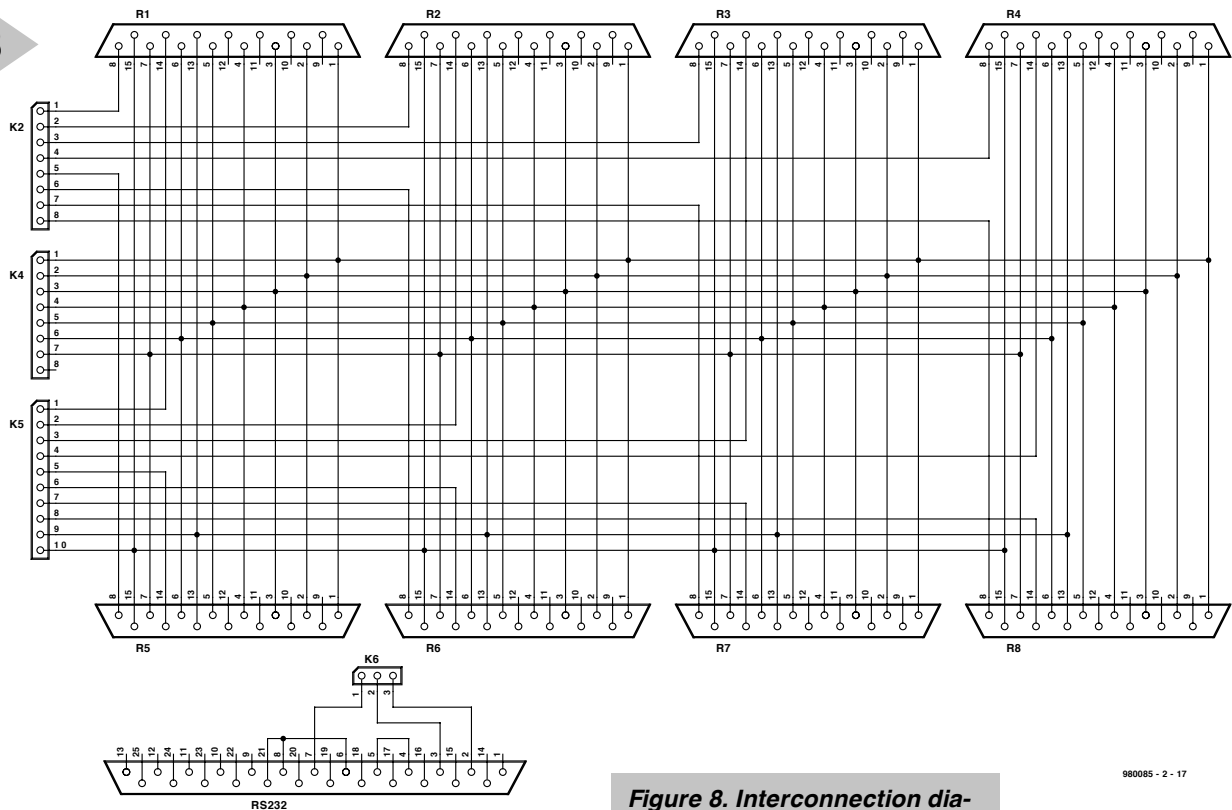
Figure 8. Interconnection diagram of the controller.

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adds a start-stop function to the signals. When the booster is actuated by briefly closing switch S_3, the relay is energized, whereupon diode D_1 lights. The relevant data is then passed by the controller to the booster.

When switch S_4 is pressed briefly, the relay is deactuated, whereupon the booster is decoupled from the controller.

When the short-circuit protection of the booster is actuated, the relay is automatically deenergized.

Connections PA_0–PA (port A) are used for communication with the monitor units. The links between the port and the units are 6-core cables, conforming to the Märklin design. The EEDTS monitor units may also be used, but they must be linked via a special adaptor cable, since they use a 5-core cable. How this cable is converted to a 6-way one is shown in **Figure 5**. If a mixture of EEDTS and Märklin monitors is used, insert the Märklin decoders between the EEDTS monitors and header K_7 on the controller.

MANUAL CONTROLS

Ports B, C, and D, on the controller are used for communication with the eight manual controls. The circuit diagram of such a control is shown in **Figure 6**.

Each manual control consists of a potentiometer, with which the speed of the associated locomotive is varied, and a number of switches.

The V_{CC} terminals of all potentiometers are interlinked, as are the

GND terminals. The wipers are individually connected to a relevant analogue input on the microcontroller.

Each manual control contains six function keys and a wire bridge. Two of the function keys – reverse and F_0 – are used with both the old and the new Motorola format. F_0 is independent of the direction of travel and is mostly used for controlling the front lights of the locomotive.

The other four keys can be used only with engine decoders that support the new Motorola format. This format is selected by a wire link on the header between pins 7 and 8. When this link is in place, the manual controls may be connected as relevant by a 10-core flatcable. The wire link enables engines driven via the old format or the new format to be used on the same track.

The individual terminals of the function keys are linked to the C-port as relevant, while their common terminals are connected to the B-port. This arrangement enables cyclical scanning of the manual controls.

It is clear that if only the old Motorola format is going to be used, function keys F_1–F_4 may be omitted.

BOOSTER

The booster, that is the amplifier that converts the digital signals to an analogue voltage which is switched between + 15 V and –15 V, will not be discussed in this part of the article, but in the next. It is intended that in future

the Märklin booster will also be supported. The TTL signals on header K_8 are intended for this booster.

CONSTRUCTION

The controller is best built on the printed-circuit board shown in **Figure 7.** Careful attention to the drawings and parts list should ensure that no problems will be encountered.

Figure 8 shows how the manual controls are interconnected and linked to the various headers.

RAILWAY MODELLING IN THE UK

There are a number of model railway magazines, but these may not always be available from newsagents without being ordered.

Railway Modeller • Peco Publications & Publicity Ltd • Underleys • Beer • Seaton • Devon EX12 3NA.

Model Railway Journal • Wild Swan Publications Ltd • 1–3 Hagbourne Road • Didcot • Oxon OX11 8DP.

Model Railway Enthusiast • Link House Magazines Ltd • Link House • Dingwall Ave • Croydon • Surrey CR9 2TA.

Further information may be gleaned from:
http://www.geocities.com/CapeCanaveral/7045/railmodel.html

[980085]

CORRECTIONS & updates

Evaluation System for 80C166 (parts 1 and 2)
March & April 1999, 990028

Errors in Components List
IC9 and IC10 should be type 74HC573, not HCT573.

C1 and C2 should be 10µF 16 tantalum bead.

R20 should be an 8-way SIL array, value 4kΩ7.

The crystal frequency mentioned with C18, C19 should be 40MHz, not 100MHz.

IC3, MAX690 should be listed as MAX690 (BATT), not MAX690 (I/O)

Miscellaneous
Pull-up resistors R1-R4 need not be mounted with the serial channels.

When a 40-MHz crystal is used for X1, it may oscillate at the fundamental frequency (13.333 MHz) instead of the third overtone. If this happens the system baudrate will not be correct and the serial communication will fail to work as described. Sug-

gested solutions are (1) to use a 40-MHz crystal oscillator module, (2) use a different 40-MHz crystal or (3) adapt the value of C18 and C19 until oscillation occurs at the third overtone.

The battery may be a 3.6-V NiCd type, but note that this is very slowly charged by the MAX690. A better solution is to use a Lithium battery.

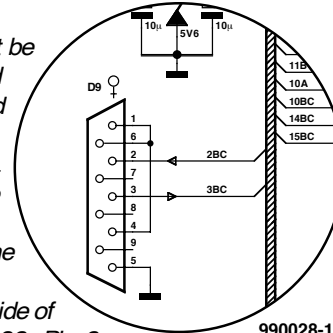The circuit diagram does not make it clear how the serial connection is made via D9. Pins 2 and 3 of this connector should not be connected TxD1 and RxD1 (i.e. the TTL side of the MAX232), but to the other (RS232) side of the MAX232. Pin 2 of D9 is therefore connected to pin 2BC via

990028-11

bus K1. Likewise pin 3 of D9 is connected to pin 3BC via bus K1. The correct connections are shown in the illustration.

Development System for 68HC11F1
June 1999, 990042

JP1 is a simple jumper. In the text and parts list, it is erroneously referred to as K3.

Part K3 is a 4-way SIL pinheader. In the text and parts list, it is erroneously referred to as JP1

R4 is erroneously listed as a SIL array with a value of 10kΩ. The PCB however only accommodates eight discrete resistors. These are numbered R4 and R6-R12. Resistor R13 in the circuit diagram equals R4 on the PCB.
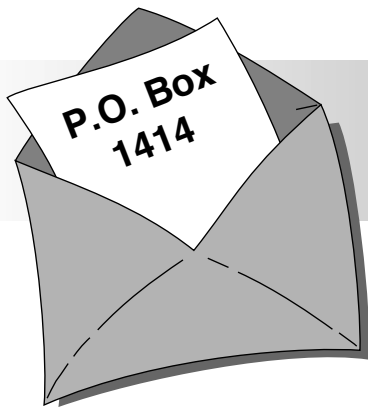
PC-Controlled Model Railway: EEDTS Pro
June 1999, 990082-2

On page 60, the text references t S3, D1 and S4 should read S2, D2 and S1 respectively.

PCB design and mains voltage
(April 1999)

Dear Editor—In the article 'PCB design and mains voltage' you state that the distance between the mains termination on a board to any other conductor must be not less than 6 mm. However, I would not like to be responsible for the designs shown. In Figure 4, the circuit is made unnecessarily dangerous by specifying a minimum distance of 3 mm between the mains terminals and the secondary circuit. It may well be that the intention here is to use a 3-core mains cable with the protective earth securely connected.

Although the concept behind the article is correct and useful, I would suggest that it is better (and safer) to specify a distance between mains-carrying tracks and terminals of at least 10 mm.

G. v. Hamersfeld

We have indicated in the article that the legal regulations must be seen (as intended) as minimum dimensions. That is why the drawings show > 3 mm and > 6 mm respectively. Any constructor or designer is, of course, free to increase these minimum spacings. This is particularly so if there is a likelihood that the minimum spacings may be inadequate (such as use in a damp atmosphere). It is, of course, sensible to make the layout of a printed-circuit board so that the spacings between tracks are not reduced needlessly.

Your suggestion of a minimum of 10 mm is laudable, but cannot be defended by any regulation or standard. But, again, even if there are no legal requirements to do so, it is always advisable to maximize spacings.

Joystick and MIDI interface
(December 1998)

Dear Editor—I have noticed that on quite a few sound cards, such as several from Yamaha, the MIDI-IN does not work if R_{10} at pin 4 of K_4 has a value of 220 Ω as specified. If the value is increased to 470 Ω, all work well.

G, Huizinga

Thanks for this tip, which many readers will no doubt find very useful.

Digital audio formats
(February 1999)

Dear Editor—In the article 'Digital audio formats' MPEG 1 Layer 3 (MP3) is discussed and it is intimated that this format uses lossy compression which is, however, not audible. I cannot agree with this.

Some years ago, I had a DCC recorder which uses PASC compression. This is similar to MP3 in that it uses a masking effect, combined with loss-free reduction, based on the threshold of hearing. I could definitely hear the compression. With complex pieces of music, this manifested itself as short repetitive whistles superimposed as it were on the music.

More recently, I listened to a MiniDisc recorder in a hi-fi retail shop. This system uses ATRAC compression, which is also similar to MP3. I was perhaps even more astonished than the salesman that I could hear the compression straightaway. In this case, it manifested itself in very short interruptions at certain frequencies. For instance, in normal speech, the s- or f- sound is seamless, but in the case of the MiniDisc it sounds as if they are subdivided into many parts which are not connected seamlessly. In the consequent 'seams' the high frequencies that should be there are all but absent.

MP3 sounds identical if the compression takes place with less than 256 kbps (6:1); at that compression, I don't hear it, but at 128 kbps (12:1) it is quite audible (at least to me). I don't know anyone else who can hear the compression that clearly (without an A/B comparison).

I should like to hear from other readers with similar experiences at lweekers@yahoo.com

L. Weekers

Compression always affects the signal. In MP3 and other systems, a method was sought and found in which the inescapable effect is inaudible for virtually everybody. Your experiences show that there are people with such good hearing that they can discern the effect.

electronics on line

DIY audio CDs
manuals, hardware and software descriptions

As a perfect complement to the articles in this month's Supplement, this Electronics On-Line page is devoted to making your own music CDs. What does the Internet have on this subject?

Naturally, every self-respecting manufacturer of hardware or software has a website where extensive product descriptions may be found. For example, if you need information on soundcards of the Soundblaster type, all you have to do is type *www.soundblaster.com* into the addressbar of web browser and what you were looking for is delivered 'free' to your PC screen.

However, besides a staggering number of product-related websites, there are many other addresses where useful information may be found on the subjects 'computer audio' and 'digital audio recording'.

The Internet magazine ('webzine') called *The Tapeless Studio* (*www.tapeless.com*) is a pretty good starting point. As indicated by its name, the webzine covers anything to do with computerised recording of audio material, including descriptions of new hardware and software for this area, and lots of additional information. Beginners, too, should find this webzine useful because it tells you, among other things, which basic equipment is required.

The Internet being such a vast repository of information, it is not surprising to see that it also supplies good descriptions of the process of copying audio tracks from a CD to a CD-R, or what you have to do if you want to record a whole LP. The address *resource.simplenet.com/primer/primer.htm* is good for an extensive *Primer on CD-R*, of which a large section is devoted to computerised audio-recording. If you want to know everything about making a digital (*last-forever*) copy of your favourite LP on a CD-Rewriteable then a more than useful story may be found at

homepages.nildram.co.uk/~abcomp/lp-cdr.htm

For an overview of PC soundcards and their main characteristics we'd advise you to have a look at the **Digital Experience** website. Although mainly aimed at working with DAT recorders, the website also offers lots of information on soundcards (*www.digitalexperience.com/cards.html*) and recording software (*www.digitalexperience.com/recsoft.html*)

(995051-1)
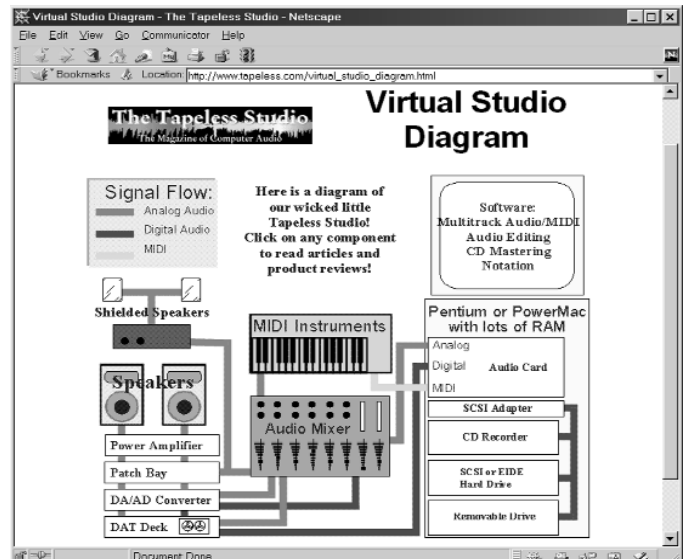
AT90S2313

can sink 20 mA. As inputs, Port D pins that are externally pulled low will source current if the pull-up resistors are activated. Port D also serves the functions of various special features of the AT90S2313 (see full datasheet, page 43).

RESET
Reset input. A low on this pin for two machine cycles while the oscillator is running resets the device.

XTAL1
Input to the inverting oscillator amplifier and input to the internal clock operating circuit.

XTAL2
Output from the inverting oscillator amplifier.

Crystal Oscillator
XTAL1 and XTAL2 are input and output, respectively, of an inverting amplifier which can be configured for use as an on-chip oscillator, as shown in Figure 2. Either a quartz crystal or a ceramic resonator may be used. To drive the device from an external clock source, XTAL2 should be left unconnected while XTAL1 is driven as shown in Figure 3.

The SRAM Data Memory
Figure 4 shows how the AT90S2313 Data Memory is organized.
The 224 Data Memory locations address the Register file, I/O Memory and the data SRAM. The first 96 locations address the Register File + I/O memory, and the next 128 address the data SRAM.
The five different addressing modes for the data memory cover: Direct, Indirect with Displacement, Indirect, Indirect with Pre-Decrement and Indirect with Post-Increment. In the register file, registers R26 to R31 feature the indirect addressing pointer registers.
The Direct addressing reaches the entire data address space.
The Indirect with Displacement mode features 63 address locations reach from the base address given by the Y and Z register.
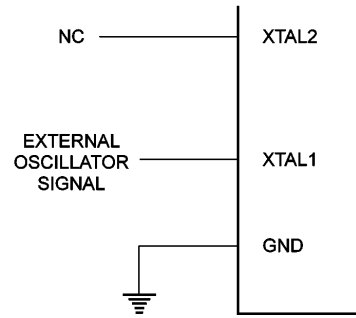When using register indirect addressing modes with

automatic pre-decrement and post-increment, the address registers X, Y and Z are used and decremented and incremented.
The 32 general purpose working registers, 64 I/O registers and the 128 bytes of data SRAM in the AT90S2313 are all directly accessible through all these addressing modes.

Figure 3. External clock drive configuration.

Figure 4. SRAM organization.

AT90S2313

AT90S2313
8-bit AVR® Microcontroller with 2K bytes In-System Programmable Flash

Manufacturer
Atmel.
Website: www.atmel.com

Features
➥ AVR® - High Performance and Low Power RISC Architecture
➥ 118 Powerful Instructions — Most Single Clock Cycle Execution
➥ 2K bytes of In-System Reprogrammable Flash
   – SPI Serial Interface for Program Downloading
   – Endurance: 1,000 Write/Erase Cycles
➥ 128 bytes EEPROM
   – Endurance: 100,000 Write/Erase Cycles
➥ 128 bytes Internal RAM
➥ 32 x 8 General Purpose Working Registers
➥ 15 Programmable I/O Lines
➥ V_{CC} : 2.7 - 6.0V
➥ Fully Static Operation
   – 0 - 10 MHz, 4.0 - 6.0V
   – 0 - 4 MHz, 2.7 - 6.0V
➥ Up to 10 MIPs Throughput at 10 MHz
➥ One 8-Bit Timer/Counter with Separate Prescaler
➥ One 16-Bit Timer/Counter with Separate Prescaler and Compare and Capture Modes
➥ Full Duplex UART
➥ Selectable 8, 9 or 10 bit PWM
➥ External and Internal Interrupt Sources

➥ Programmable Watchdog Timer with On-Chip Oscillator
➥ On-Chip Analog Comparator
➥ Low Power Idle and Power Down Modes
➥ Programming Lock for Software Security
➥ 20-Pin Device

Application Example
Wave File Player, *Elektor Electronics* February 1999
Description
The AT90S2313 is a low-power CMOS 8-bit microcontroller based on the AVR enhanced RISC architecture. By executing powerful instructions in a single clock cycle, the AT90S2313 achieves throughputs approaching 1 MIPS per MHz allowing the system designer to optimize power consumption versus processing speed.
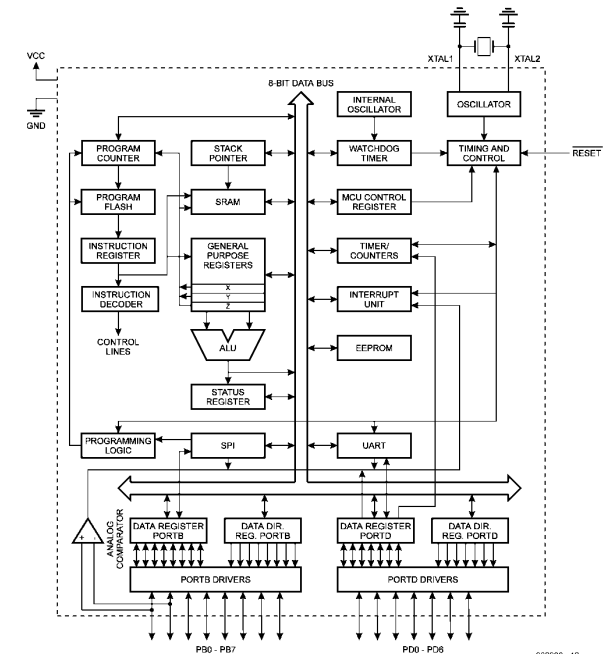The AVR core combines a rich instruction set with 32

Figure 1. AT90S2313 block diagram.

general purpose working registers. All the 32 registers are directly connected to the Arithmetic Logic Unit (ALU), allowing two independent registers to be accessed in one single instruction executed in one clock cycle. The resulting architecture is more code efficient while achieving throughputs up to ten times faster than conventional CISC microcontrollers. The AT90S2313 provides the following features: 2K bytes of In-System Programmable Flash, 128 bytes EEPROM, 128 bytes SRAM, 15 general purpose I/O lines, 32 general purpose working registers, flexible timer/counters with compare modes, internal and external interrupts, a programmable serial UART, programmable Watchdog Timer with internal oscillator, an SPI serial port for Flash Memory downloading and two software selectable power saving modes. The Idle Mode stops the CPU while allowing the SRAM, timer/counters, SPI port and interrupt system to continue functioning. The power down mode saves the register contents but freezes the oscillator, disabling all other chip functions until the next interrupt or hardware reset. The device is manufactured using Atmel's high density non-volatile memory technology. The on-chip In-System Programmable Flash allows the program memory to be reprogrammed in-system through an SPI serial interface or by a conventional nonvolatile memory programmer. By combining an enhanced RISC 8-bit CPU with In-System Programmable Flash on a monolithic chip, the Atmel AT90S2313 is a powerful microcontroller that provides a highly flexible and cost effective solution to many embedded control applications. The AT90S2313 AVR is supported with a full suite of program and system development tools including: C compilers, macro assemblers, program debugger/simulators, in-circuit emulators, and evaluation kits.

Pin Descriptions
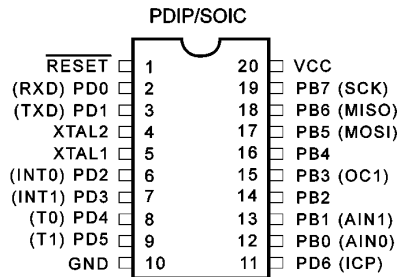
VCC
Supply voltage pin.

GND
Ground pin.

Port B (PB7..PB0)
Port B is an 8-bit bi-directional I/O port. Port pins can

provide internal pull-up resistors (selected for each

PDIP/SOIC


```
         RESET  [ 1      20 ]  VCC
   (RXD) PD0    [ 2      19 ]  PB7 (SCK)
   (TXD) PD1    [ 3      18 ]  PB6 (MISO)
         XTAL2  [ 4      17 ]  PB5 (MOSI)
         XTAL1  [ 5      16 ]  PB4
   (INT0) PD2   [ 6      15 ]  PB3 (OC1)
   (INT1) PD3   [ 7      14 ]  PB2
    (T0) PD4    [ 8      13 ]  PB1 (AIN1)
    (T1) PD5    [ 9      12 ]  PB0 (AIN0)
         GND    [ 10     11 ]  PD6 (ICP)
```
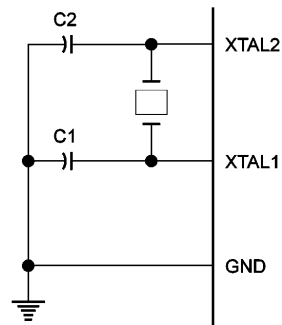
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Pin configuration

bit). PB0 and PB1 also serve as the positive input (AIN0) and the negative input (AIN1), respectively, of the on-chip analog comparator. The Port B output buffers can sink 20mA and can drive LED displays directly. When pins PB0 to PB7 are used as inputs and are externally pulled low, they will source current if the internal pull-up resistors are activated. Port B also serves the functions of various special features of the AT90S2313 (see full datasheet, page 38).

Port D (PD6..PD0)
Port D has seven bi-directional I/O pins with internal pull-up resistors, PD6..PD0. The Port D output buffers

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Figure 2. Oscillator connections

I/O Memory

The I/O space definition of the AT90S2313 is shown in the following table.

Address Hex	Name	Function
$3F ($5F)	SREG	Status Register
$3D ($5D)	SPL	Stack Pointer Low
$3B ($5B)	GIMSK	General Interrupt MaSK register
$3A ($5A)	GIFR	General Interrupt Flag Register
$39 ($59)	TIMSK	Timer/Counter Interrupt MaSK register
$38 ($58)	TIFR	Timer/Counter Interrupt Flag register
$35 ($55)	MCUCR	MCU general Control Register
$33 ($53)	TCCR0	Timer/Counter 0 Control Register
$32 ($52)	TCNT0	Timer/Counter 0 (8-bit)
$2F ($4F)	TCCR1A	Timer/Counter 1 Control Register A
$2E ($4E)	TCCR1B	Timer/Counter 1 Control Register B
$2D ($4D)	TCNT1H	Timer/Counter 1 High Byte
$2C ($4C)	TCNT1L	Timer/Counter 1 Low Byte
$2B ($4B)	OCR1H	Output Compare Register 1 High Byte
$2A ($4A)	OCR1L	Output Compare Register 1 Low Byte
$25 ($45)	ICR1H	T/C 1 Input Capture Register High Byte
$24 ($44)	ICR1L	T/C 1 Input Capture Register Low Byte
$21 ($41)	WDTCR	Watchdog Timer Control Register
$1E ($3E)	EEAR	EEPROM Address Register
$1D ($3D)	EEDR	EEPROM Data Register
$1C ($3C)	EECR	EEPROM Control Register
$18 ($38)	PORTB	Data Register, Port B
$17 ($37)	DDRB	Data Direction Register, Port B
$16 ($36)	PINB	Input Pins, Port B
$12 ($32)	PORTD	Data Register, Port D
$11 ($31)	DDRD	Data Direction Register, Port D
$10 ($30)	PIND	Input Pins, Port D
$0C ($2C)	UDR	UART I/O Data Register
$0B ($2B)	USR	UART Status Register
$0A ($2A)	UCR	UART Control Register
$09 ($29)	UBRR	UART Baud Rate Register
$08 ($28)	ACSR	Analog Comparator Control and Status Register

All the different AT90S2313 I/O and peripherals are placed in the I/O space. The different I/O locations are accessed by the IN and OUT instructions transferring data between the 32 general purpose working registers and the I/O space. I/O registers within the address range $00 - $1F are directly bit-accessible using the SBI and CBI instructions. In these registers, the value of single bits can be checked by using the SBIS and SBIC instructions. Refer to the instruction set chapter for more details. When using the I/O specific commands, IN, OUT, SBIS and SBIC, the I/O addresses $00 - $3F must be used. When addressing I/O registers as SRAM, $20 must be added to this address. All I/O register addresses are shown with the SRAM address in parentheses.
The different I/O and peripherals control registers are explained in the following sections. When using the I/O specific commands, IN, OUT, SBIS and SBIC, the I/O addresses $00 - $3F must be used. When addressing I/O registers as SRAM, $20 must be added to this address. All I/O register addresses are shown with the SRAM address in parentheses.