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Economical high-end preamp

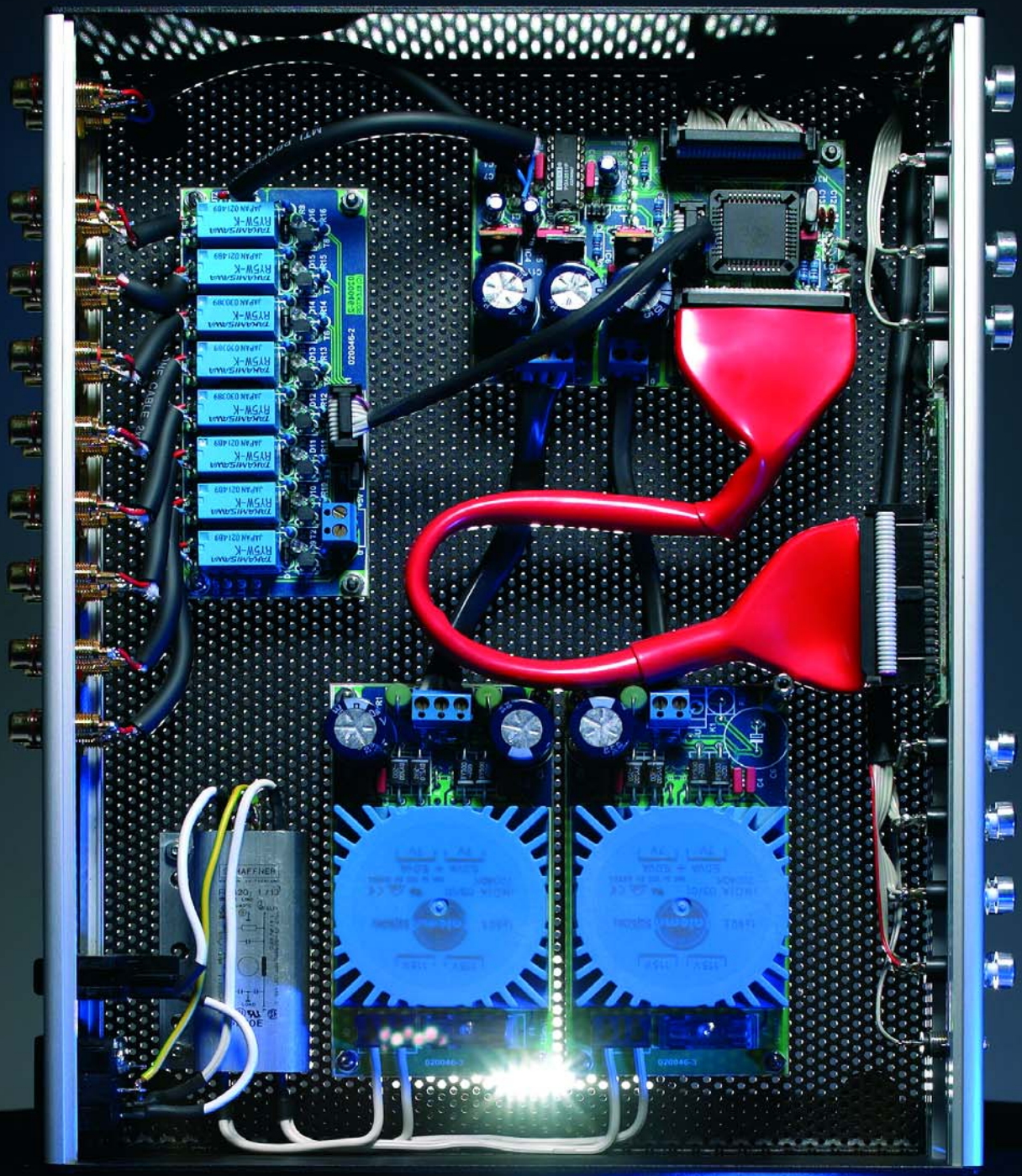
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High-End-Preamp

Benjamin Hinrichs



The modern design of this preamplifier yields audiophile specs, convenient operation and an attractive price. This is made possible by using a top-end digitally controlled attenuator/amplifier IC.

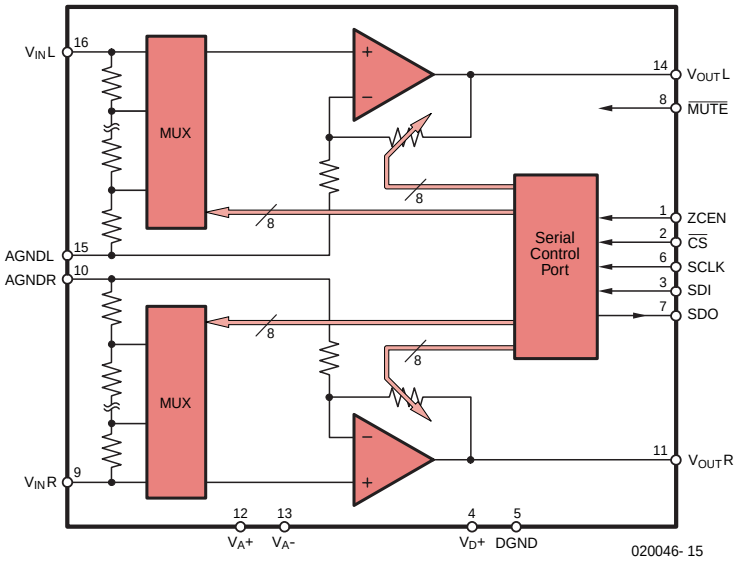


Figure 1. Functional block diagram of the PGA2311. The attenuation and gain are configured using a 3-wire serial bus.

Analogue audio electronics appears to have entered a dormant stage. In our present age of CDs, DVDs and MP3, traditional preamplifiers have been relegated to the role of signal distribution and, primarily, adjusting the volume level.

Our expectations for a modern high-quality preamplifier are that in addition to being easy to use, it should perform these signal distribution and volume adjustment functions with the greatest possible accuracy and the least possible distortion. And that is exactly where things start to get difficult.

Volume controls are commonly implemented using potentiometers, which are available in a wide variety of price ranges and types. Since we are normally dealing with a stereo signal, we need two mechanically coupled potentiometers. The decisive factor is the tracking of the two potentiometers, since this determines how closely the volumes of the right and left channels will match each other. Poor tracking is especially noticeable (and disturbing) at low volume settings. If we also want to have a balance control, we need an additional set of coupled potentiometers, and the tracking errors will add together.

The maximum permissible tracking error for 'audiophile' sensibility is 3 dB, but ideally it should be less than 1 dB. As can easily be seen from **Table 1**, these values are clearly exceeded by normal carbon-film potentiometers, and even high-quality carbon-film potentiometers have difficulty maintaining adequate tracking accuracy with increasing age.

An audiophile alternative to potentiometers is to use high-quality, multi-position rotary switches with close-tolerance resistors. However, the right-hand column of the table shows that suitable special rotary switches having extremely low crosstalk and contact resistance cost around £80–90, which is rather expensive.

The disadvantage of this solution involves more than just the price (we will need two or three such switches for the volume and balance controls and the input selector switch, if present), since rotary switches cannot be remotely controlled.

If we wish to have the convenience of remote control as well as excellent tracking, there's no getting around a design using conductive-plastic potentiometers with a motor and the associated control electronics and mechanical parts. This also involves considerable effort and expense, and just about everything must be duplicated for a balance control.

The PGA2311 stereo audio volume control IC

Admittedly, the idea of using a 'digital' IC for volume adjustment, and furthermore controlling it using a microcontroller, may evoke a sceptical frown from many an audiophile. Ten years or more ago, this scepticism would certainly have been justified, but the semiconductor industry has made enormous progress in this area. All of the major functions can now be integrated into a single chip, with results that can easily hold their own against the best mechanical solutions.

The Texas Instruments PGA2311 volume control IC used in this project is moderately priced and provides out-

Table 1. Tracking errors and price indications of various types of volume control

Type	Tracking error [dB]	Approximate price [£]
Carbon-film potentiometer	>3	3
High-quality carbon-film potentiometer	0.5 - 3	10 - 20
Conductive plastic potentiometer	0.1 - 0,3	25 - 55
Rotary switch	0.1	80 - 90
Motor and accessories	–	12 - 80
PGA2311PA	0.1	2.5 - 12

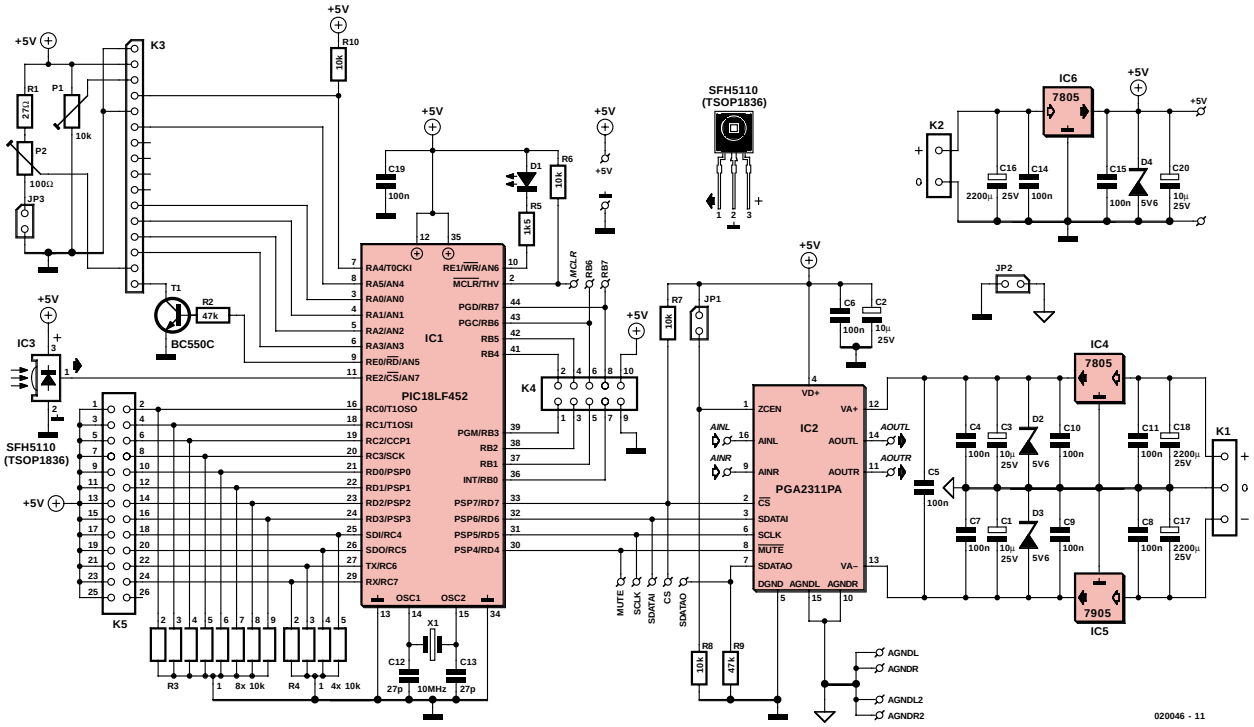


Figure 2. Circuit diagram of the main circuit board, which is divided into an analogue part and a digital part.

standing tracking without any degradation of tracking accuracy when balance adjustment is used, and it also has very good technical specifications. Another major advantage is that it can be digitally configured. This makes user-friendly operation possible (including remote control). An example of a high-end builder who uses this technology is Jeff Rowland.

The PGA2311, whose internal structure is shown in **Figure 1**, is a digitally controlled analogue stereo volume control with certain refinements. The two channels can be independently adjusted over a range of –95.5 dB to + 31.5 dB in steps of 0.5 dB, which yields an adjustment range of 127 dB. The tracking error between the two channels, as well as the absolute setting accuracy of each of the channels, is ±0.05 dB. This naturally means that a balance adjustment can also be implemented without any problems, since the high absolute setting accuracy prevents any offset from occurring. Another noteworthy feature is that the IC can directly drive 600-Ω loads.

The multiplexer (MUX) switches individual resistors to set the attenuation. After the attenuator, the signal passes through an output buffer, which can also provide gain via an adjustable feedback resistance.

The IC is controlled via a serial SPI interface. Clock signal SCLK transfers a single 16-bit word to the IC via the SDI line. The first eight bits set the volume level for the right channel, while the second eight bits set the level for the left channel. The minimum value (0) represents Mute, and the maximum value (255) represents a gain of +31.5 dB.

For fully noise-free switching, zero-crossing detection can be enabled via the ZCEN lead. If it is enabled, the IC analyses the music signal and attempts to perform the switching during a zero crossing. If no zero crossing is detected within 16 milliseconds, the switching is performed without waiting any longer. Due to system design constraints, the ramp function of the software does not work properly if zero crossing detection is enabled.

In principle, four different types of ICs can be used in this circuit. **Table 2** lists the differences among these ICs. The original design was developed for the Crystal (Cirrus Logic) type CS3310. The equivalent competitive product from Texas Instruments is the PGA2310, which is not only pin-compatible, but also has significantly better internal specifications. A particularly attractive feature of the latter IC is that it can handle signals up to 27 V_{pp} if the analogue supply voltage is increased to 30 V. The improved type PGA2311 has even better channel separation, and the selected 'A' version has a better specification for total harmonic distortion plus noise (THD+N). For this reason, we selected the PGA2311A for his project.

We can also mention the PGA4311 in passing. This is a four-channel version of the PGA2311 and is only available in the SOIC package. It can be used with only minor modifications to the circuit board layout and software.

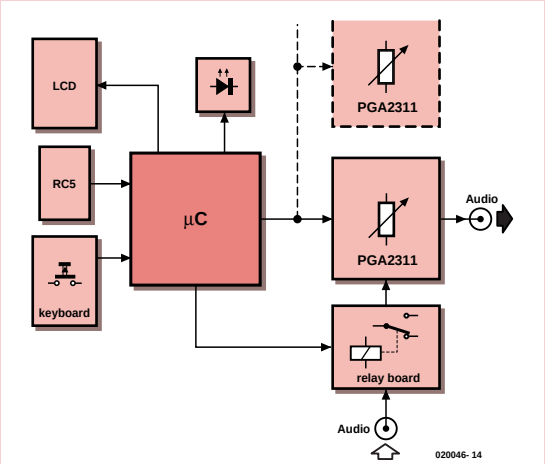
The control centre

The main circuit board, which forms the control centre for the preamplifier, requires surprisingly few components. This is due to the high integration density of the two ICs used here. By far the majority of the components are used to generate clean supply voltages.

The circuit is split into an analogue portion and a digital portion. The digital portion contains a Microchip PIC18LF452 microcontroller clocked at 10 MHz by a crystal oscillator. This microcontroller has 8-bit registers and 16-bit instruction words. Microchip has not shown much flair in assigning part numbers to its PIC microcontrollers. For instance, the PIC16F84, PIC16F628, PIC16F877 and PIC12F675 belong to the PIC14 family, while the

The project

This project has a modular structure, so it can easily be adapted to individual preferences. It consists of a general-purpose power supply board, an input selector board and the main circuit board, which holds the volume control IC and microcontroller. An LC display module, a keypad and an IR remote control unit complete the package. The hardware and software are described in this article, which is the first of two parts. The other two circuit boards are described in Part 2, which will appear in next issue of *Elektor Electronics*. Modifying the control program for the microcontroller is also described in Part 2.



Block diagram of the preamplifier. The option of connecting additional PGA2311 ICs in parallel, as indicated by the dashed outline, is described in Part 2.

If you build all of the circuit boards as described, you will have a complete, remotely controllable preamplifier with input selection and function display. However, you can also dispense with the input selector and use only the volume control capability. If you wish, you can also omit the display, or you can omit the volume control IC on the main circuit board and use the remainder of the circuit as a remotely controllable relay circuit board for various applications. It is also possible to operate several volume controls in parallel, for instance in order to construct a multi-channel amplifier. For this purpose, only the volume control ICs have to be fitted to the main circuit boards for the additional channels. The control software for the microcontroller can be adapted to suit almost any imaginable application without reprogramming the microcontroller.

PIC18LF452 used here belongs to the PIC16 family.
The PIC18LF452 has a Flash program memory with a capacity of 32 KB (which is adequate for the rather extensive software), 1.5 KB of RAM and a 256-byte EEPROM. Its 31 stack levels provide adequate manoeuvring room for calling functions and procedures if the contents of all of the registers are written to the stack to allow the called procedure to use the registers. When control is returned to the calling procedure, the register contents are retrieved from the stack to allow the calling procedure to continue processing from the point where it transferred control. If frequently used subroutines are implemented using functions and procedures, the resulting interleaving of program execution can quickly exceed the capacity of a relatively shallow stack.

Before discussing the software in any more detail, let's have a look at the peripheral resources available to the microcontroller. The volume control (IC2) is connected to the microcontroller via the serial SPI bus. In addition, the microcontroller can select the PGA2311 using the CS line, and it can mute the output by placing a Low level on the MUTE line. These four lines, as well as the data output line (SDO), are externally accessible to allow several volume controls to be connected in parallel (as described in Part 2 of this article).

The remainder of Port D and all of Port C are fitted with pull-down resistors (consisting of the two SIL arrays R3 and R4) and routed to pin header K5, to which the pushbutton switches for controlling the preamplifier are connected.

K5 pins	Function
1&2, ..., 15&16	Channels 1–8
17&18	Volume Down
19&20	Volume Up
21&22	Volume Left
23&24	Volume Right

The functions are essentially self-explanatory, but as you might imagine, additional functions are also implemented using combinations of buttons. All of the functions can also be selected using a remote control unit. IC3 is a 36-kHz infrared receiver, which filters, demodulates and cleans up the received light signal and boosts it to TTL levels, all without a single external component. An RC5 decoder is built into the software, so all types of RC5 remote control units (Philips, Grundig, etc.) can be used to control the preamplifier. The IR receiver is connected to RE2, which is one of the three Port E lines.

Table 2. Volume control ICs suitable for use in this project.

Type	Dynamic range [dB]	THD+N [%]	Channel separation [dB]	U_{out} (max) [V_{pp}]
CS3310-KP	116	0.001	–110	7.5
PGA2310PA	120	0.0004	–126	27
PGA2311P	120	0.0004	–130	7.5
PGA2311PA	120	0.0002	–130	7.5



The configuration options are so manifold that without a clearly organised display presentation you could quickly loose track of where you are, particularly when programming the basic settings. Via Port A, the microcontroller software drives an LC display with two lines of 16 characters and background illumination. In normal operation the display shows the channel names and volume setting, while in Set-up mode it is used to select channel designations and basic volume control settings. Pull-up resistor R10 connected to RA4 is necessary because this port lead has an open-drain output and thus cannot switch to a High level without a pull-up resistor. Trimpot P1 adjusts the display contrast, while trimpot P2 adjusts the brightness of the background illumination. JP3 extends the adjustment range. The microcontroller can switch the background illumination on or off via port line RE0 and transistor T1.

The microcontroller drives the relay board via K4. Each of port lines RB0–RB7 selects one of the eight audio inputs. The behaviour of the Status LED (D1) can be configured using the Set-up menu. This is described in more detail later on, along with the significance of the three lines MCLR, RB6 and RB7 that are led out from the board.

The main circuit board has separate power distribution for the analogue and digital portions. The ground potentials must be connected at a suitable location via wire bridge JP2. The single +5-V digital supply voltage and the symmetrical ±5-V analogue supply voltages are stabilized in the traditional manner using fixed voltage regulators with the customary buffer and decoupling capacitors. For all three voltages, 5.6-V Zener diodes are provided as 'backup' safety devices in case excessively high voltages appear on the outputs of the fixed voltage regulators.

Operation

After switch-on, the software checks whether reasonable value are located in the EEPROM. If this is not the case, such as immediately after the microcontroller has been programmed, default values are loaded. Otherwise the software loads the stored values and configures the volume control accordingly.

When a volume control button is pressed (Up, Down, Left, or Right), the software checks whether the adjustment is possible and whether the lower or upper limit of the adjustment range has been reached. Pressing the Up and Down buttons simultaneously causes the preamp output to be muted. Pressing the Left and Right buttons simultaneously restores the balance to the middle position, with the volume being set to the average of the values for the two stereo channels. If one of the input channel buttons is pressed, the channel is changed, with the output being muted during switching. Alternatively, the preamplifier can be configured via the Set-up menu to use a ramp. In this case, when the channel is switched the volume is first ramped down and then ramped back up again after the channel change.

An offset can be assigned to each channel, which is useful if the signal sources have different volume levels. The offset is applied to the set volume level when the associated channel is selected, and when a different channel is selected it is automatically removed. If an offset would cause one of the volume limits to be violated, it is ignored. The channel name selected using the Set-up menu is shown on the display.

All of the functions of the preamplifier can be controlled using an RC5-compatible remote control unit. Naturally, the preamplifier can be freely configured using the Set-up menu to allow an existing RC5 remote control unit to be used to control all of its functions.

The selected configuration settings are stored in the microcontroller EEPROM and are thus available each time the preamplifier is subsequently used.

Set-up

The software has default values for all configuration settings. All of the functions of the software can be adapted to individual needs via the Set-up menu. To enter the Set-up mode, hold the Channel 1 button pressed while switching on the preamp.

The Set-up configuration can only be modified using the control buttons on the preamplifier; it cannot be adjusted using the remote control. The buttons have the following functions in the Set-up mode:

DOWN	Next entry
UP	Previous entry
LEFT	Exit
RIGHT	Enter

1) RC5 IR Set-up

Reads an RC5 code from a remote control unit, displays the code and assigns it to one of the following buttons: Channel 1–8, Down, Up, Left, Right, or Mute.

Defaults
Configured for a Grundig remote control unit.

Buttons

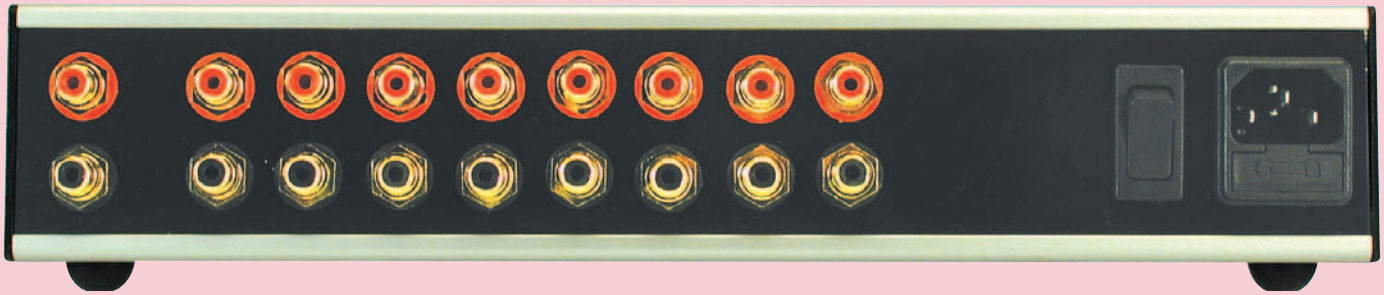
LEFT	Exit
RIGHT	Next

2) RC5 IR Test

Reads and displays an RC5 code from a remote control unit. Intended to be used to check settings made using IR Set-up. Can also be used to test an RC5 remote control unit.

Button

LEFT	Exit



3) Maximum Volume

Sets the maximum allowable volume level in dB.

Default
+31.5 dB (maximum)

Buttons

DOWN	Reduce volume level
UP	Increase volume level
LEFT	Exit

4) Ramp

The ramp function gradually decreases the volume before switching channels and gradually restores it afterwards. The ramp function can be enabled or disabled, and the delay between successive volume steps can be configured. Enabling the zero crossing detection function (JP1) can impair the operation of the ramp function.

Defaults

Use ramp:	Yes
Ramp delay:	15 ms

Buttons

DOWN	Use ramp: Yes/No
	Ramp delay: – 1 ms
UP	Use ramp: Yes/No
	Ramp delay: + 1 ms
LEFT	Exit
RIGHT	Next

5) Relay Test

Energizes all relays for testing.

6) Input Type

This allows the input configuration to be set to either 8 channels (Single) or 2 × 4 channels (Double). This is useful if you want to switch not only the signal leads but also the ground leads. The input type should also be set to Double for balanced signal sources. In the Double mode, the relays are switched in pairs as follows: RE1+RE5, RE2+RE6, RE3+RE7, and RE4+RE8.

Default
Single

Buttons

DOWN	Single / Double
UP	Single / Double
LEFT	Exit

7) Offsets

An offset can be defined for each channel. It is applied when the channel is selected and removed when a different channel is selected. If applying an offset would violate one of the volume limits (Mute or Maximum Volume), it is not used. The value is shown in dB.

Defaults

Channel 0–8:	0 dB

Buttons

DOWN	– 0.5 dB
UP	+ 0.5 dB
LEFT	Exit
RIGHT	Next

8) Channel Names

Each channel can be assigned a name selected from the following list:
Aux, Aux2, CD, CD2, DAC, DAC2, DVD, DVD2, DVD-Audio, DVD-Audio2, Line, Line2, Phono, Phono2, SACD, SACD2, Tape, Tape2, Tuner, Tuner2, TV, TV2, VCR, VCR2, Video, Video2, Sat, Sat2, DCC, DCC2, MD, MD2, DAT, DAT2, PC, PC2.

Defaults

Channel 1:	CD
Channel 2:	Phono
Channel 3:	DVD
Channel 4:	SACD
Channel 5:	DVD-Audio
Channel 6:	DAC
Channel 7:	Tape
Channel 8:	Line

Buttons

DOWN	Next list item
UP	Previous list item
LEFT	Exit
RIGHT	Next

9) Hardware Set-up

Configures the type of hardware used. This only affects what is shown on the display. The options are Normal (preamp), Input Only (channel selection only), and Volume Only (volume adjustment only).

Default
Normal

Buttons

DOWN/UP	Normal / Input Only / Volume Only
LEFT	Exit

10) LED Set-up

Sets the LED behaviour. Options: Delay Off, Always Off, Always On.

Default
Delay Off

Buttons

DOWN/UP	Delay Off / Always Off / Always On
LEFT	Exit

11) LED Set-up

Sets the behaviour of the LCD background illumination. Options: Delay Off, Always Off, Always On.

Default
Delay Off

Buttons

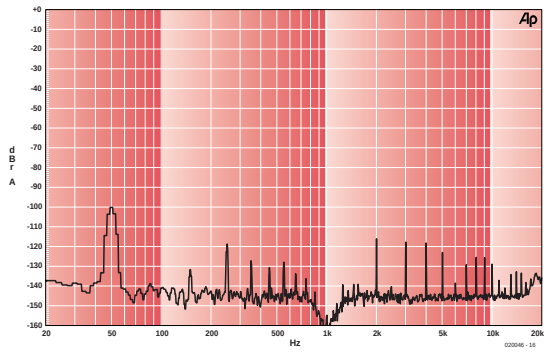
DOWN/UP	Delay Off / Always Off / Always On
LEFT	Exit

12) Power-up Volume

Sets the switch-on behaviour. This menu can be used to configure two settings. The first setting controls the switch-on behaviour and has the following options: Last (set the volume to

Measured performance

Confidence (in the data sheets) is good, but measurement (by the *Elektor Electronics* lab) is better! The results of the FFT analysis of a 1-kHz signal with an amplitude of 1 Veff indicate two things. First, the overall harmonic distortion figure of 0.0012 % is dominated by the induced 50-Hz mains noise (at –100 dB) if the measurement is made over a bandwidth of 20 Hz to 20 kHz. Second, the first three harmonics of the test frequency are located in the range of –116 dB to –118 dB. If the bandwidth range for the measurement is shifted to 100 Hz – 200 kHz, the THD+N value drops to only 0.0005 %. This is a fantastically low value.



The harmonics of the 1-kHz signal lie below –116 dB.

In order to further reduce the effect of mains interference, it is recommended to separate the mains input and power supply board as far as possible. Fully enclosing the main circuit board and relay board (inside a tinned sheet-metal box located inside the main enclosure, for example) could also have a beneficial effect.

Test results at unity gain (0 dB)

Nominal input sensitivity	**200 mV**
Nominal output voltage	**200 mV**
Maximum output voltage	**2.4 Vrms (THD+N = 0.01 %)**
Input impedance	**10 kΩ (input selected)**
	∞ (input not selected)
Output impedance	**< 0.6 Ω**
Bandwidth *	**0–3 MHz**
	0–150 kHz (gain 31.5 dB)
Harmonic distortion (THD+N) *	**0.0005 % (1 kHz, B = 100 Hz – 22 kHz)**
	0.0012 % (1 kHz, B = 80 kHz)
	0.002 % (20 Hz – 20 kHz, B = 80 kHz)
Signal to noise ratio (S/N) *	**100 dB (B = 22 kHz)**
	113 dBA
Channel separation **	**> 88 dB (1 kHz)**
	> 62 dB (20 kHz)
Crosstalk **	**< 98 dB (1 kHz)**
	< 88 dB (20 kHz)

*** at Vout = 1 V**
**** with open input terminated in 560 Ω**

the same level as when the preamp was switched off), Mute, Mute ➞ Last (muted on switch-on, with the previous volume setting being restored after a button is pressed), and Preset (always use a configurable preset value).
The second setting is the preset value. The current volume level can be stored as the preset value by pressing Up or Down.

Default
Last
Preset values: Mute, Mute

Buttons

DOWN/UP	Last / Mute / Mute ➞ Last / Preset (store current volume as preset)
LEFT	Exit
RIGHT	Next

13) Restore Defaults

Restores the predefined default configuration settings. This also occurs if a newly programmed microcontroller without reasonable configuration values in the EEPROM is fitted.

Default

Volume:	Mute
Channel:	1
All others:	see above

(020046-I)

References

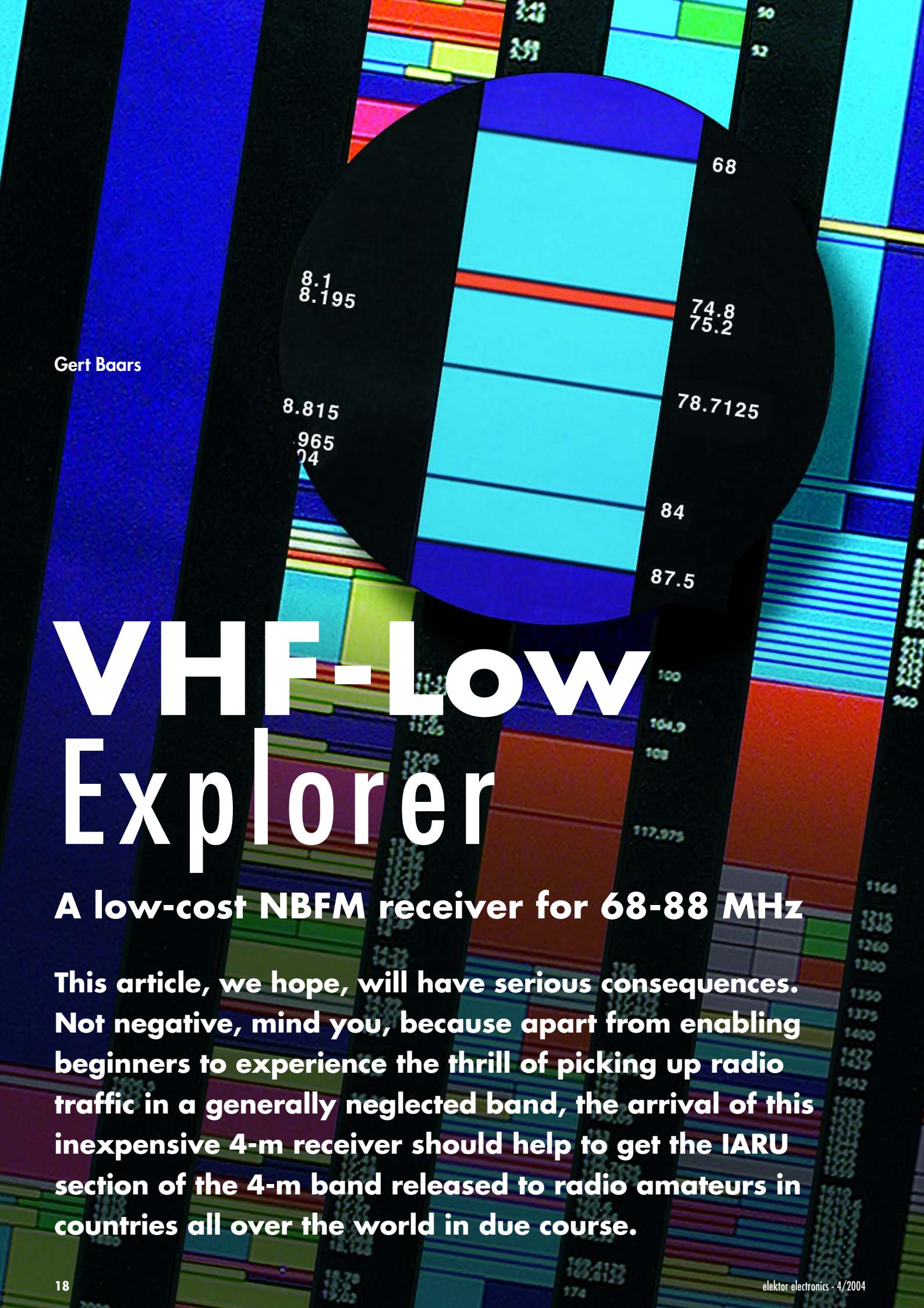
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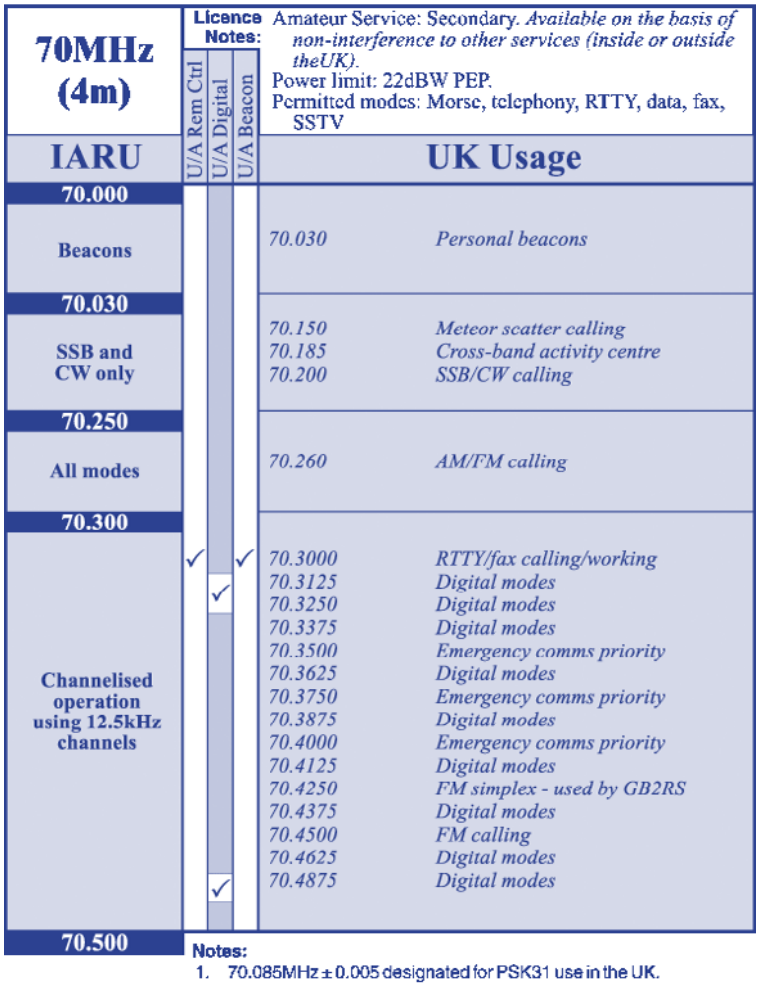


Gert Baars

VHF-Low Explorer

A low-cost NBFM receiver for 68-88 MHz

This article, we hope, will have serious consequences. Not negative, mind you, because apart from enabling beginners to experience the thrill of picking up radio traffic in a generally neglected band, the arrival of this inexpensive 4-m receiver should help to get the IARU section of the 4-m band released to radio amateurs in countries all over the world in due course.



70MHz (4m)	Licence Notes: U/A Rem Ctrl	U/A Digital	U/A Beacon	Amateur Service: Secondary. *Available on the basis of non-interference to other services (inside or outside theUK).* Power limit: 22dBW PEP. Permitted modes: Morse, telephony, RTTY, data, fax, SSTV
IARU				**UK Usage**
70.000				
Beacons				*70.030 Personal beacons*
70.030				
SSB and CW only				*70.150 Meteor scatter calling* *70.185 Cross-band activity centre* *70.200 SSB/CW calling*
70.250				
All modes				*70.260 AM/FM calling*
70.300				
Channelised operation using 12.5kHz channels	✓		✓	*70.3000 RTTY/fax calling/working*
		✓		*70.3125 Digital modes*
				70.3250 Digital modes
				70.3375 Digital modes
				70.3500 Emergency comms priority
				70.3625 Digital modes
				70.3750 Emergency comms priority
				70.3875 Digital modes
				70.4000 Emergency comms priority
				70.4125 Digital modes
				70.4250 FM simplex - used by GB2RS
				70.4375 Digital modes
				70.4500 FM calling
				70.4625 Digital modes
		✓		*70.4875 Digital modes*
70.500				

Notes:
1. 70.085MHz ± 0.005 designated for PSK31 use in the UK.

Figure 1. If and when radio regulatory authorities eventually decide to allocate the 4-m to radio amateurs then the IARU recommendations will be followed, with strong guidance available from the UK example.

For many decades the UK has been home to the 4-m amateur radio band, also known as '70 MHz', and the IARU-approved band plan shown in **Figure 1** is on the wall in many a radio shack up and down the country. So far, only a few other countries including Ireland, Denmark and — quite recently — Croatia have followed suit by releasing the frequency range between 70.000 and 70.5000 MHz for use by licensed radio amateurs. Unfortunately, in many other countries the relevant frequency range is in use by government or military services which need to have a few arms twisted (pun intended) before they move out. Although it is too early to say whether or not the arrival of new communication systems like Tetra, C2000 and Astrid on the European Continent and elsewhere will free up the 70 MHz band to amateurs, there can be no harm in increasing the pressure on various national radio regulatory authorities to do the necessary paperwork. At least in Holland and Poland, the word is out that amateurs are 'interested' in the 70 MHz band. Let's hope the pressure rises as the 4-m band is fantastic for VHF DX-ing. *Elektor Electronics* being an international publication, we will gladly assist in spreading the word in as many countries as possible.

What's in it for me

While the radio amateur fraternity is poised to grab their share of the spectrum around 70 MHz, it should be noted that the 68-88 MHz band has other, equally interesting users and applications including Government, MoD and PMR (private mobile radio) communications (not encrypted in many cases), security services, telemetry and the odd TV station. Unless you live in a really remote place, even a simple antenna in your loft will bring in a surprising number of stations using the 4-m band. Tune and Explore!

Design considerations

From the very start of this project, the design was to remain as simple as possible. This decision has important consequences as well as a background we'd like to share with you. Sure, a receiver for the 68-88 MHz band could be a double-conversion superheterodyne design employing a 10.7-MHz filter, a 10-turn pot for the tuning and a final intermediate frequency (IF) bandwidth of 15 kHz to suit NBFM (narrow-band frequency modulated) signals picked up at a sensitivity of 1 μV or so, not forgetting a squelch to make sure the receiver is quiet when nothing is received. Great shopping list, but such a receiver will be expensive as well as difficult to adjust by beginners. Next, please!

The good news is that an attractive alternative is available in the form of the TDA7000 chip from Philips that's been around for more than 10 years now, which is quite remarkable for a consumer-market chip. This 'evergreen', then, contains a complete radio receiver with a very low IF of just 70 kHz. Okay, so image frequencies occur just 2 x 70 kHz = 140 kHz away from the desired signals, but that need not be a problem because on the positive side we do not have to worry too much about the input filtering. Also, the IF filter responsible for

COMPONENTS LIST

Resistors:
R1=100kΩ
R2=150kΩ
R3=100Ω
R4=22kΩ
R5=330kΩ
P1=50kΩ logarithmic potentiometer
P2=50kΩ linear potentiometer

Capacitors:
C1=39pF
C2=27pF
C3,C6,C14=10nF
C4,C11,C13,C19,C23,C24,C25,C26,C29=100nF
C5,C12=1nF, lead pitch 5mm
C7=100nF, lead pitch 5mm
C8=220pF
C9,C18=330pF
C10=10pF
C15,C17=3nF3, lead pitch 5mm
C16=180pF
C20=22pF PTFE trimmer
C21=150pF

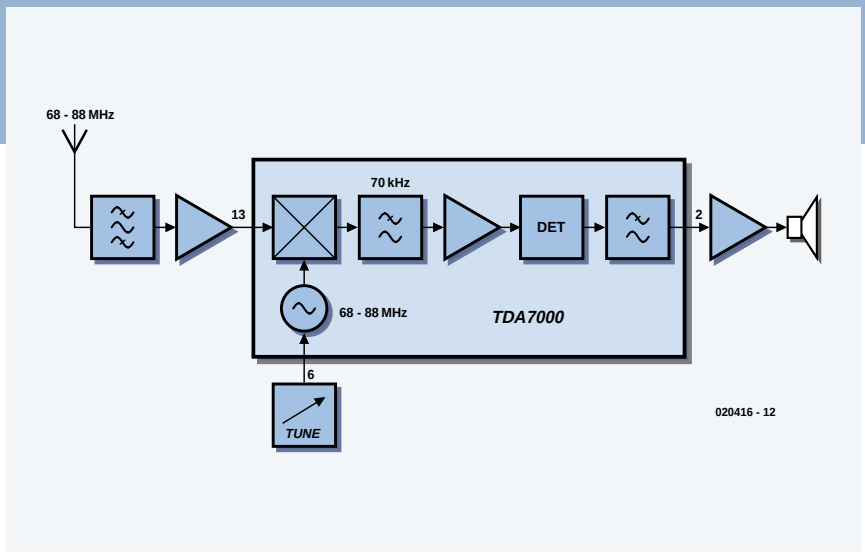


Figure 2. Block diagram of the single-conversion receiver. Note the low intermediate frequency of just 70 kHz which in our case has a number of advantages!

the selectivity may be realised as a simple R-C network, obviating the need for expensive and esoteric quartz or ceramic filters.

Block diagram

Even if you are not a radio boffin, the block diagram of the proposed receiver in **Figure 2** should be largely self-explanatory. The TDA7000 contains a muting circuit which is activated at a level of about 6 μV. As we will want to use a whip antenna as the bare minimum, an RF preamplifier will have to be inserted between the antenna and the input of the TDA7000. Everything from the output of the RF preamp right up to the input of the audio amplifier is contained in the TDA7000. If you want to know everything about the chip, get a copy of the datasheet (see Web pointers).

Inside the receiver

Figure 3 pictures the circuit diagram of our little receiver. MOSFET T1 at the antenna input provides a gain of about 18 dB across the band, driving the TDA7000 RF input via coupling capacitor C5. The receiver's input impedance is 50 Ω to match most types of coax cable available these days. A number of capacitors strewn around the TDA7000 ensure an IF bandwidth of about 70 kHz. The VFO (variable frequency oscillator) inside the chip is tuned by a varicap (D1) which gets its bias voltage from tuning pot P2. IC2, a 78L05, supplies the regulated 5 volts for the receiver chip, the preamp and, importantly, the tuning pot.

The circuit configuration around the TDA7000 follows information from Philips on making the chip better compatible with NBFM signals. After all, the TDA7000 was originally designed for reception of VHF FM broadcast stations, which at 100+ kHz deviation are much wider than the 'thin' PMR signals (3 kHz) we're interested in. None the less, as the IC will produce a rather low nett output signal, some extra amplification is furnished in the audio section by adding an electrolytic capacitor between pins 1 and 8 of the LM386 AF power amp (another evergreen).
Hang on, where are the adjustments and the dreaded home-made coils in this receiver? Well there's only trimmer C20 to adjust the tuning to 68-88 MHz. The receiver employs off the shelf miniature chokes only, so there are 0 (say, zero) coils to wind.

Build it!

At this point, you should have enough confidence and 'inside knowledge' about the receiver to start building it, if necessary with the help of a friend with RF experience. If you do not have the means to make your own board, you can easily order a ready-made one through our Readers Services. The board, pictured in **Figure 4** together with its external elements, is single-sided with a large copper plane at the solder side to assist in RF stability, screening and decoupling. There are many small ceramic capacitors on the board which need to be positively identified before they are soldered in place. The same goes for the three miniature chokes, the coloured bands on them indicating the value in micro-henries. The TDA7000 should be soldered directly on to the board.
The 4-legged MOSFET T1 is mounted at the solder side of the board, perched directly onto four solder pads. The close-up photograph in **Figure 5** should help to get our message across.

We would suggest using a small diecast case from Hammond to house the receiver and the battery. The case is then drilled to secure the volume pot, tuning pot and the loudspeaker. Battery powering is not a must however, and you may decide to power the receiver from an existing DC source like a cheap mains adapter. This will require one additional hole to be drilled and filed for the mains adaptor socket.

Caveats and limitations

Due to the simplicity of the design, some inherent limitations should be

C22=100pF
C27=2nF2
C28=10µF 16V radial
C30,C31=100µF 16V radial

Inductors:
L1=100nH (brown, black, silver)
L2=330nH (orange, orange, silver)
L3=180nH (brown, grey, silver)

Semiconductors:
D1=BB911
T1=BF981
IC1=TDA7000
IC2=78L05
IC3=LM386 N4

Miscellaneous:
8 ohm, 1 watt miniature loudspeaker
Diecast case: e.g., Hammond type 1590B

9V battery (PP3 / 6F22) with clip-on leads
PCB, order code 020416-1 (see Readers Services page or www.elektor-electronics.co.uk)

taken into account. First, the receiver will be found rather susceptible to cross modulation, breakthrough and general interference from nearby FM broadcast transmitters. This not at all surprising in view of the nearby frequencies (89-107 MHz) and power levels in the kilowatts range. Good shielding, coax cable and a tuned antenna for 4 m (see 'Antenna' inset) should remove most of the interference. Second, a small problem with spurious oscillation was discovered when the receiver's RF input is not terminated

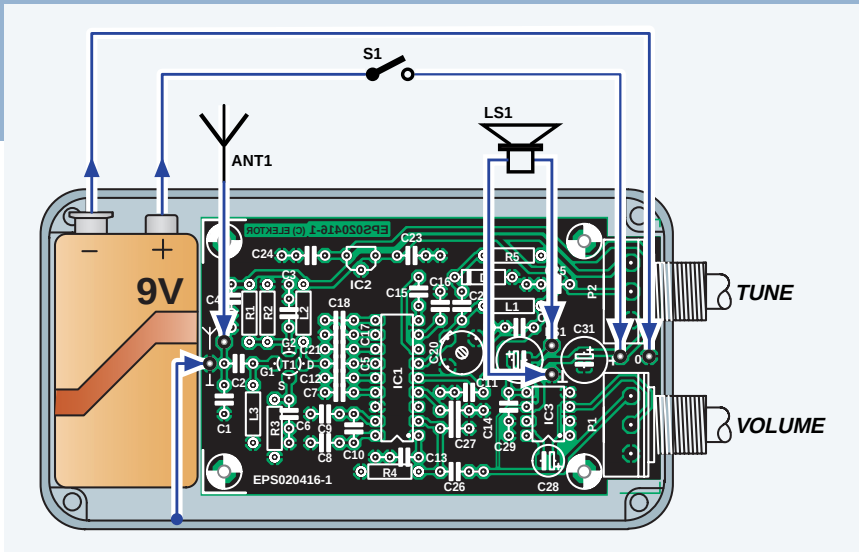


Figure 4. A drawing to show how the board is connected to its external elements. The copper track layout may be found elsewhere in this issue.

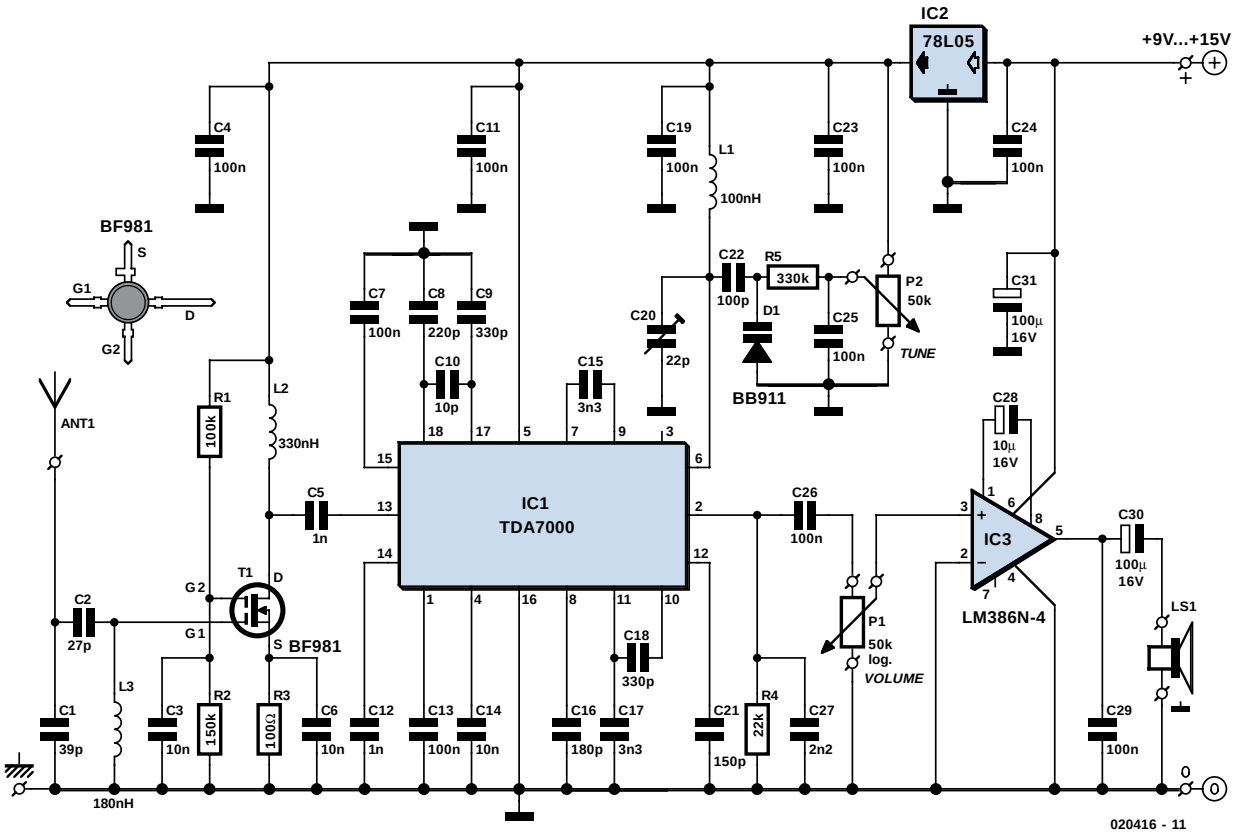


Figure 3. The circuit diagram of the VHF-Low Explorer has few surprises and proves the simplicity of the design.



About the author

Gert Baars (42) has been active in electronics from a young age. In 1988 he graduated in electronics at the Polytechnic of Alkmaar in the Netherlands. Gert's main interest is RF electronics, but also software and digital hardware. He has had over 20 projects published in this magazine since 1997, including the successful Airband Receiver, the 20-m Band Receiver and the DDS RF Signal Generator. In the future, Gert hopes to write about Atmel micros in control of RF equipment, and possibly the design of a UHF sweep generator. He invites comments and suggestions by email, g.baars13@chello.nl.

with 50 Ω. For the rest, nothing to stop you from exploring the 4-metres band.

(020416-1)

Web pointers

TDA7000 datasheet:
www.semiconductors.philips.com/pip/TDA7000.html#datasheet

70 MHz info page and news reflector:
www.70mhz.org

International Amateur Radio Union (IARU):
www.iaru.org

Radio Society of Great Britain (RSGB):
www.rsgb.org

4-m band in Ireland:
www.qsl.net/ei7gl/vhfpage.htm#70mhz

70 MHz Yagi antennas (DK7ZB):
www.qsl.net/dk7zb/start1.htm

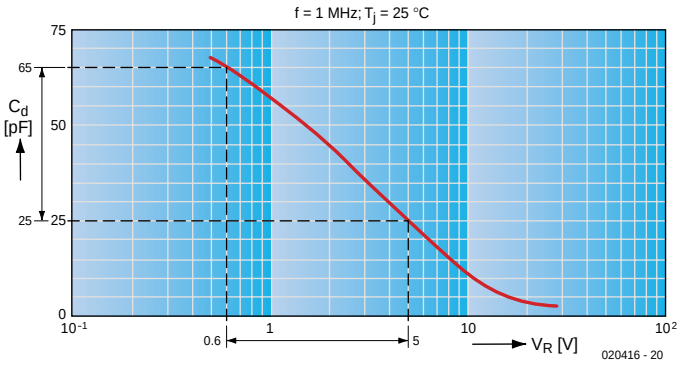
7-element Yagi for 70 MHz (M1CCZ):
www.qsl.net/zr6dxb/PROJECTS/4mBeam/4MBeam.htm

Figure 5. This is as close as we could get a lens to the MOSFET at the solder side of the board. Use the MOSFET pinout drawing in the circuit diagram to get the device positioned the right way around.



Varicap selection or the delta-C / delta-V issue

A



For some strange reason varicaps (or variable capacitance diodes) have always been rather elusive components. Try this: design, engineer and publish a design at instant T and you'll find that the varicap you've specified after hours of careful researching has disappeared from the market at [T+1 day]. Here at Elektor we're optimists by nature but because we anticipate supply problems with the BB911 varicap specified for this present receiver, we thought we'd give you a few clues to help you find equivalent types.

The component values in the circuit diagram guarantee a tuning range of 68-88 MHz, with trimmer C20 defining the edges of the tuning range and the capacitance ratio of the BB911 defining the width of about 20 MHz. In other words, trimmer C20 shifts the tuning range and D1 determines the width of the tuning range. The two parameters have some interaction, of course.

If you are only interested in, say, the 4-m amateur band (70.0-70.5 MHz) then a narrow tuning range is sufficient and you will have no trouble getting just about any old VHF varicap to work in the receiver, simply by adjusting C20 to a 'known-good' signal in the band (ask for assistance from a local licensed radio amateur).

If, on the other hand, you want everything from low-band TV (68 MHz) to police, MoD and government PMR (in some cases just below 87 MHz) then some thought should be given to the selection of the varicap.

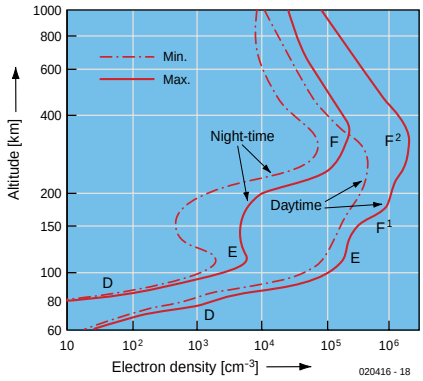
*The Philips BB911 was chosen because of its relatively large capacitance range of 25 pF to 65 pF for a corresponding tuning voltage of 0.6 V to 5 V — see **Figure A** (courtesy Philips Semiconductors). That's right, the capacitance presented by a varicap is inversely proportional to the voltage applied across the device! Mathematically, though, [delta-C / delta-V] is a dimensionless device constant which, in the case of the BB911, works out at about 9 for the linear part of its capacitance range.*

If you can't get the BB911locally there's no reason to abandon the project or send Blue Murder emails to the Editor because there are lots of alternatives. Do not be afraid to experiment. In many cases, unlabelled varicaps picked up at radio rallies or salvaged from an FM radio may be used, provided you know they are for VHF. Connecting a few varicaps in parallel ('stacking') is perfectly legitimate in order to arrive at the desired $\Delta V/\Delta C$ value and hence the receiver's tuning span.

Features at a glance

- Single conversion receiver
- Frequency range 68-88 MHz (VHF-Low band)
- Free-running VFO
- TDA7000 FM Radio Circuit modified for NBFM
- MOSFET preamplifier
- Single-board construction
- On-board audio amplifier
- 1.7 µV sensitivity for 12 dB SINAD (3 kHz deviation)
- Power supply 9-18 VDC, 20 mA (muted)

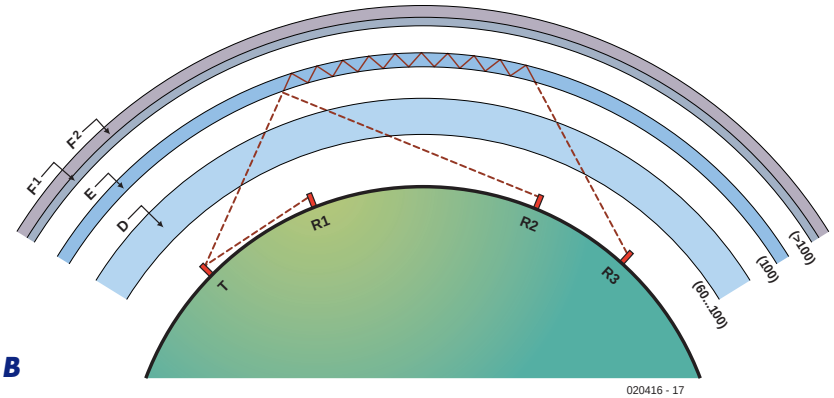
Propagation — the total surprise factor



A

The propagation of radio waves is a fascinating phenomenon because most of it is guesswork and sheer surprise. That is not to say the subject has not been studied extensively by researchers and radio amateurs — far from it, a number of underlying principles have been described in scientific terms as early as the 1920s by Nobel Laureate Sir Edward V. Appleton (1892-1965). Appleton discovered that radio waves, depending in their frequency, were subject to refraction, reflection and (partial) absorption by certain regions of the earth's atmosphere. These regions are marked by different electron densities and occur at heights of 60-400 km above the earth. The basic distribution is shown in ***Figure A****. You will search in vain for the A, B and C regions. This is because Appleton first discovered the region around 100 km height and called it 'electron' region. The D and F (actually F1 & F2) regions were discovered later when the name E region was already established. Today, researchers employ extremely sophisticated radio equipment as well as observations from radio amateurs in an attempt to prove the existence of more 'layers' in the atmosphere.*

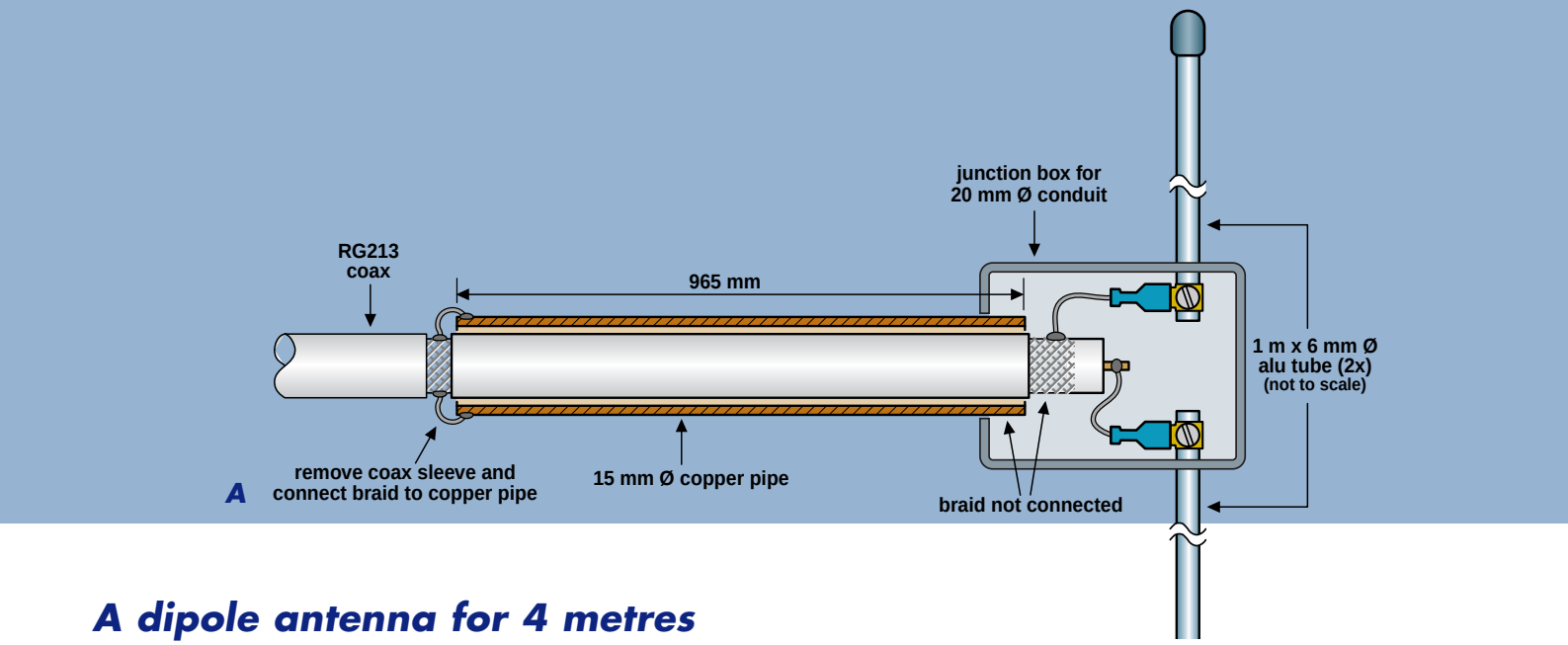
Because it is easily ionised, the E region is favourable for reflection and refraction of signals in the 70 MHz and VHF bands in general. Apart from rather unexpected behaviour, usually during periods of high air pressure, the E region is also predictable in that the electron density drops considerably at sunset due to a lesser degree of ionisation. As an aside, the E region reflects medium-wave band signals at night time when the absorption by the D region largely disappears.



B

Sporadic E (Es) is what we are after for our 70-MHz receiver, and the reason should be obvious if you look at ***Figure B****. Normally, the range of a transmitter T using the VHF-Low band is governed by line of sight, so it will reach receiver R1 as the farthest location. However, with a bit of help from Es the signal may be reflected and reach receiver R2 which, seen from T, is way below the horizon. In extreme case the signal may even 'bounce' within the E layer and reach receiver R3.*

Es is due to the formation of 'clouds' of densely ionised regions in the atmosphere at a height of 100-125 km. Es typically occurs during summer months, but exceptions have been noted. Given a sufficient degree of ionisation (sometimes helped by sunspot outburst), radio contacts via sporadic E have been made over distances of 2000 miles and more. A good way to check for Es activity is to use your receiver to monitor the signal strength of one of the many beacons in the 70-MHz radio amateur band, or TV stations near the low end of the band. Many years ago, thanks to a peak in sunspot activity coupled with massive Es cloud activity across the Atlantic Ocean, police cars from Boston and New York could be heard loud and clear in Europe, some signals even making it across police repeaters on this side of the ocean. Starski & Hutch Ten-four!



A

A dipole antenna for 4 metres

No receiver is complete without a matching antenna. Commercial offerings for the 4-m band being few and far between (or scrapped by PMR fleet owners), we decided to present a design for a low-budget get-u-going dipole. Not sophisticated, want a directional antenna? Then try the links at the end of this article. Too difficult? It doesn't get much simpler than this, so give this antenna design a try and you'll be pleasantly surprised. The antenna is great for initial experiments even when installed on your attic.

Our ingredients and tools are:

- *a length of 50 Ω RG213 or RG8 coax cable (10.3 mm outside diameter)*
- *a piece of copper pipe, 15 mm outside diameter, length 965 mm*
- *two aluminium rods, 6 mm diameter, length 1 m*
- *two cable eyelets*
- *a round T-junction box for electrical conduit, 20 mm openings*
- *some not too thin wire*
- *a powerful soldering iron (> 50 watts)*
- *nylon or plastic bushes, 20 mm diameter*
- *permission from the missus*

*The drawing in **Figure A** is intended as a guide to constructing the antenna. The copper pipe acts as a balun (balanced to unbalanced transformer), not only matching the asymmetrical coax cable to the symmetrical dipole, but also stepping down the dipole impedance of about 72 Ω to the cable impedance of 50 Ω. RF buffs will like to refer to it as a bazooka or sleeve balun. Unless you have electrical connection materials to fit to the rod ends (like a 60-A electrical 'chocolate block' terminal strip), simply flatten and drill the ends of the aluminium rods to allow screws to be used for the connection with the cable eyelets. For extra rigidity, the rods are fed through bushes (drilled to accept them) where they enter the junction box, and the box itself may be filled with potting compound or hard setting silicone sealant. The other ends of the alu rods should be deburred, rounded off and sealed to prevent moisture ingress.*

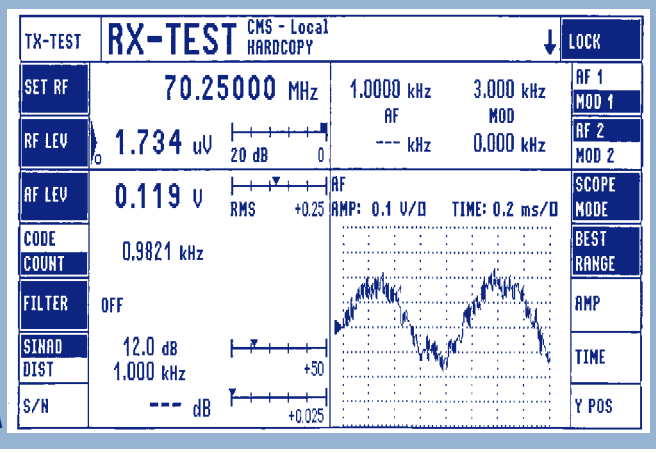
If the antenna is to be used out of doors, the junction of the copper pipe and the coax braid should be protected as well. This may be achieved by inserting the balun assembly in a length of 20-mm dia conduit and filling the lot with silicone sealant. All soldering to the coax cable should be done as quickly as possible to prevent deformation of the PTFE (Teflon) core and consequently creating 'impedance humps'.

Variations on the theme are possible, but be careful if you lack experience. You may, for instance, decide to use much thinner material for the dipole arms (for instance, lengths of welding rod), thinking it will make no difference as it is only the length that counts. Wrong, because your antenna will lose much of its broadband response and will present an acceptable VSWR at about 75 MHz ± 2 MHz only. A larger rod diameter increases the bandwidth so the antenna can be used for the entire VHF-Low band (68-88 MHz). That is why base antennas use for PMR services in the VHF bands are so thick!

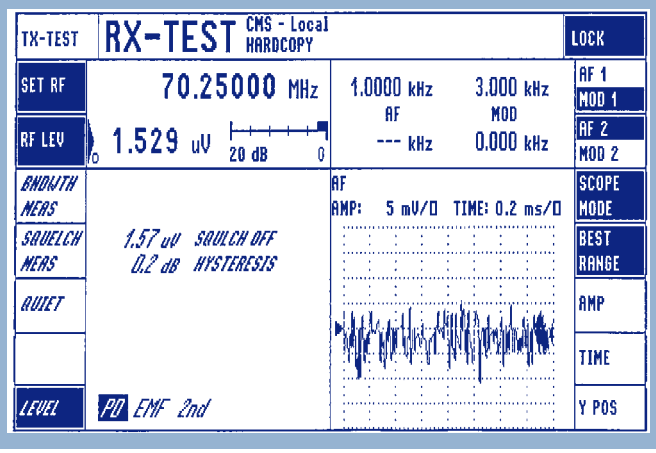
On the rack!

*While tidying up the design of the VHF-Low Explorer for publication in this issue, an opportunity arose to have our little receiver tested on a high-end instrument called the **Rohde & Schwarz CMS 54 Radio Communications Service Monitor**. This instrument (pictured here) can perform automated measurements across the frequency range 0.4 MHz to 1 GHz which is not normally within the capacity of the RF test equipment available in the Elektor Electronics design laboratory. The offer came from Mr. Ed Warnier PA1EW who is totally conversant with this piece of kit, handling and operating it as if he were driving his car to the supermarket. Not only the test results obtained from the receiver are worth telling you about, but also the basics of some of the specific tests the instrument can perform, as they may be unknown to many readers entering the radio hobby.*

*A few facts had to be established first. Our receiver is designed for narrowband FM (NBFM) reception between 68 MHz and 88 MHz, these values marking what is generally referred to as the VHF-Low communications band. As the receiver is of specific interest to radio amateurs, it was decided to tune it to **70.250 MHz** being the centre of the 4-metre band as defined by the IARU. The receiver being VFO tuned*



A



B

and lacking a frequency readout, reverse thinking quickly lead to the CMS 54 being set to 70.250 MHz and cheerfully tuning the receiver until the test signal was audible.

Hooray, it works!

Receiver sensitivity measurement

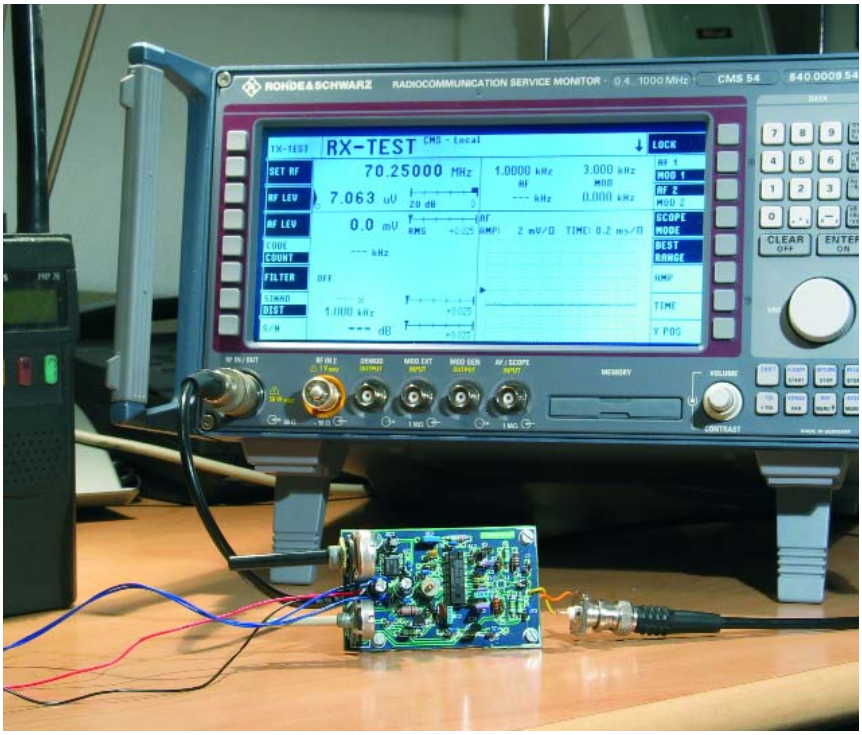
With the generator tuned to the receiver (or was it the other way around?) a great moment arrives — we're ready to decrease the RF output level on the generator until the receiver under investigation loses the signal. The VHF-Low Explorer not having an adjustable squelch or squelch defeat switch, the transition from 'very noisy signal' to 'muted' was found to be fairly abrupt (more about this further on).

Receiver sensitivity is defined as the RF signal level at which the receiver's audio output signal achieves a certain signal-to-noise ratio. For NBFM receivers, SINAD = 12 dB is considered the standard — meaning that the signal we would like to hear is 12 dB above the sum of noise and distortion (hence the acronym SINAD; an older measurement standard, S/N, employs noise only in the quotient). The RF signal level is usually given in microvolts pd (potential difference) although emf (electromotive force) is preferred by purists. The CMS 54, then, had its RF signal output connected to the receiver input (by a length of RG58 coax cable) and the audio input, to the receiver's audio output. After a manual adjustment of the volume control on the receiver, an automated measurement is started on the CMS 54 which steps down its RF signal level until it measures an audio signal of 12 dB SINAD. The RF signal level at which that happens is frozen and displayed — see ***screendump A****. In our case, a sensitivity of about* ***1.7 microvolts for 12 dB SINAD*** *was obtained which is not bad at all given the simplicity of the design. The test was carried out using a test tone of 1 kHz and a deviation of 3 kHz. As you can see in the screendump, the CMS 54 also displays a real-time image of the receiver's output signal.*

Absolute sensitivity and squelch action

Since the CMS 54 is capable of interpreting audio signals with such amazing precision it has no problems at all detecting when such a signal is passed or muted by the receiver. The latter action is taken care of by the squelch ('mute') function built into the TDA7000. An automated, stepped measurement was launched again, this time to establish the RF signal level at which the squelch closes. The result, about 1.6 microvolts, can be seen in ***Screendump B****. Ed kindly informed us that a squelch hysteresis of just 0.2 dB is not favourable for NBFM listening, a value of 2-3 dB being the standard. A bit more hysteresis ensures that stations dropping into the noise do not cause the squelch to close abruptly. Rather, the receiver will 'follow' such flutter-infested signals which, although barely intelligible in the noise will not cause the squelch to 'chatter'.*

The CMS 54 has a plethora of other functions for some really gruelling tests on radio communications equipment and in particular PMRs. We hope to be able to use it again some time in the future.



$$W \geq \frac{U+V}{1+U\frac{V}{C^2}}$$

Project c+

Data comms beyond light speed

Research papers by their nature are difficult for the average layman to understand. Here in the *Elektor Electronics* laboratory between the odd cup of coffee we spend much time studying scientific reports. A recent paper by a Japanese research group caught our eye because we were able to replicate the effects described using a very simple circuit and the results are so profound that it could change forever the way we send data.

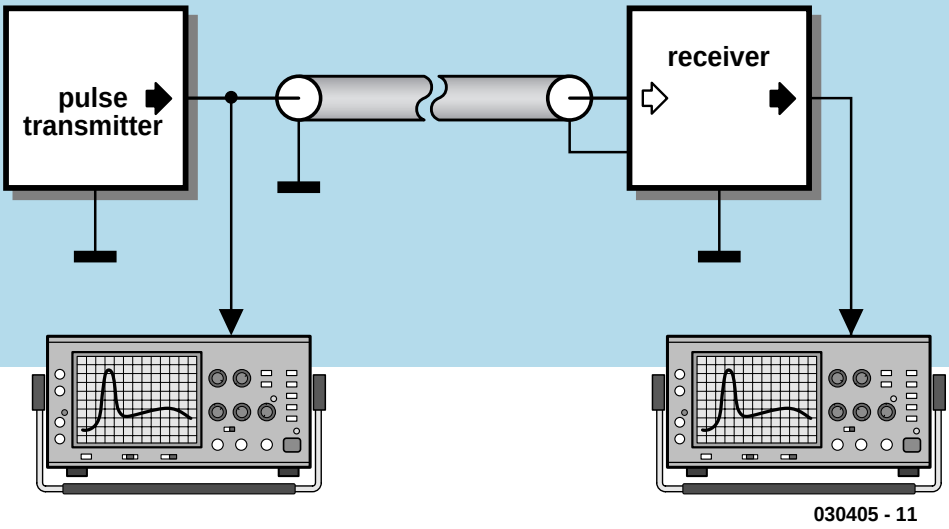


Figure 1. Experimental circuit for the Project c+.

The properties of light that most of you will be familiar with are that it travels in a straight line at a speed, *c*, of approximately 3 x 10^8 m/s. We should not forget to mention that this is only true when it is passing through a vacuum and not influenced by gravity. As soon as it enters another medium, light is slowed down and bent (refracted). In the latter part of 2003 M. D. Lukin and his colleagues at Harvard University announced in Nature [1] that they had successfully slowed down a pulse of light in a matrix of rubidium atoms such that it was stationary for a few hundred milliseconds before it was released and accelerated to regain light speed. The uses of this technology are manifold in the fields of quantum communications and photonics.
Drawing on these techniques a team of Japanese researchers have adapted the technology (termed *I-Ging*) to accelerate precise (low frequency) pulses of electromagnetic energy along transmission lines to achieve incredible propagation speeds.

Inverting causality?

The experimental set-up is shown in **Figure 1**. The pulse transmitter on the left of the diagram generates a signal pulse that is sent through the medium (either an optic fibre or simple cable) to the receiver circuit on the right hand side of the diagram. Two oscilloscopes are used to display the signal pulse when it is introduced to the medium and recovered by the receiver. The purpose of **Project c+** is to demonstrate that by careful matching of the pulse to the fundamental characteristics of the transmission medium it is possible to achieve vastly reduced propagation delays.

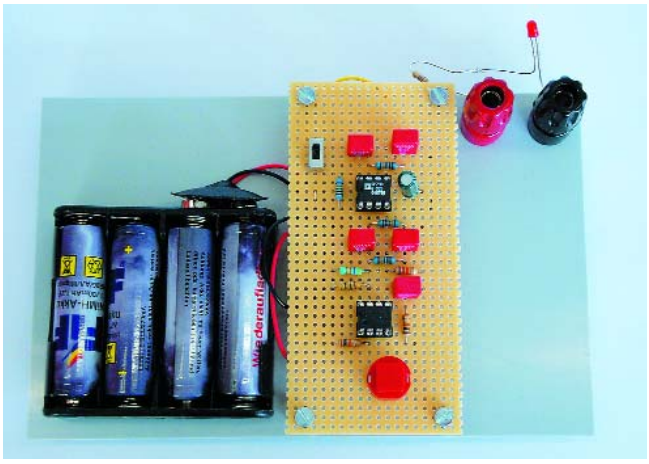
The breakthrough

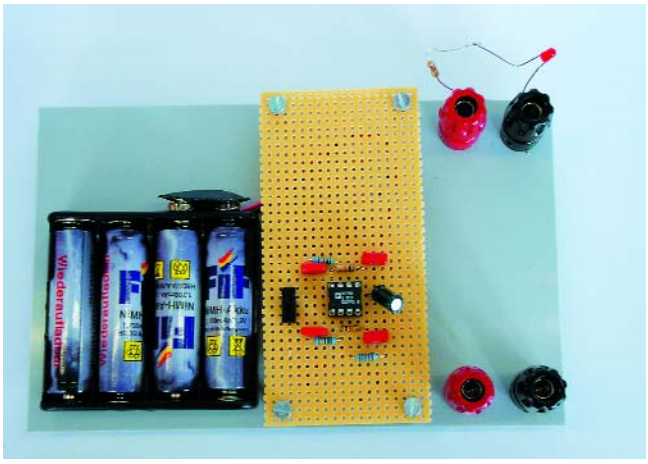
It is often the case that a scientific breakthrough occurs by accident where unexpected or anomalous results suddenly cause the investigation to veer off in a totally different direction. The story of this breakthrough is a case in point, it seems as though during routine network testing one of the research team mistakenly entered a frequency range of mHz (yes, that's millihertz) rather than MHz (megahertz) into a programmable waveform generator. The resultant effect on the test circuit was truly bizarre and may have easily been dismissed or overlooked had the researcher been less observant.
Normally in the field of high speed communications we are accustomed to propagation delays in the order of picoseconds or even femtoseconds but provided we use the correctly profiled pulse shape (a fundamental *I-Ging* pulse containing no higher order harmonics) coupled to a transmission medium originally intended for much higher frequency signals it seems as though it is possible to accelerate the pulse to such an extent that it undergoes localised temporal inversion in its passage through the medium and arrives much earlier than predicted (hence *c*+).

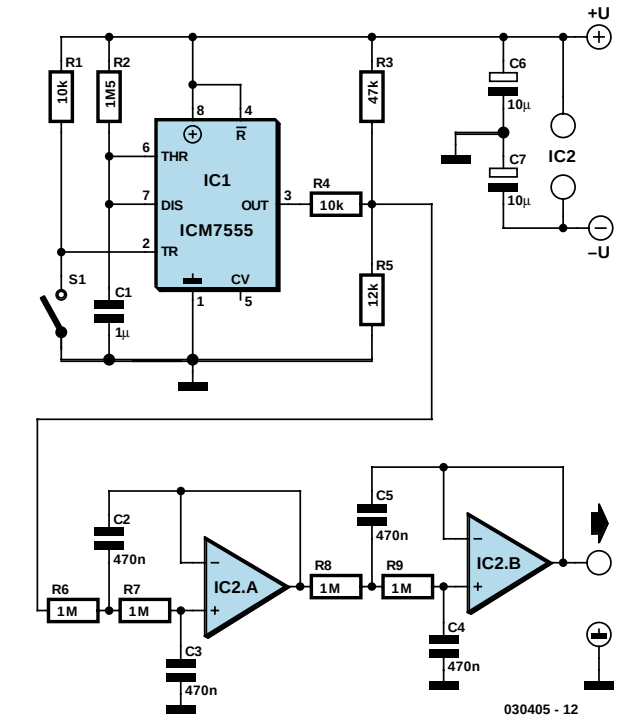
The experimental circuit

The pulse generator in **Figure 2** is quite straightforward and consists of a CMOS 555 timer IC together with two cascaded band-pass filters to produce the necessary pulse shape. The filter resistors are all 1% tolerance metal film types with a value of 1 MΩ (choose matched values where possible). Filter capacitors are low-loss foil types with 1% tolerance. The correct choice of components is critical to the success of the circuit, a fact to which any competent audio engineer will attest.

There are two possible power supply configurations for the circuit: either plus and minus 4.8 V consisting of eight NiMH AA type cells (if the TL082 op-amp is used) or a single-ended 4.8 V supply if a precision rail to rail op-amp like the OP290 is used. In this case the negative pole of the supply must be connected to the earth and all minus supply connections in the circuit diagram. To be scientifically rigorous the generator and receiver should be powered from separate power supplies to avoid cross coupling but a common supply will suffice for this experiment.
Once the circuit is completed the output waveform of the *I-Ging* generator can be observed on an oscilloscope. When S1 is pressed a transmission pulse similar to that shown in **Figure 3** will be generated and sent down the medium. Each square graticule represents 1 second.







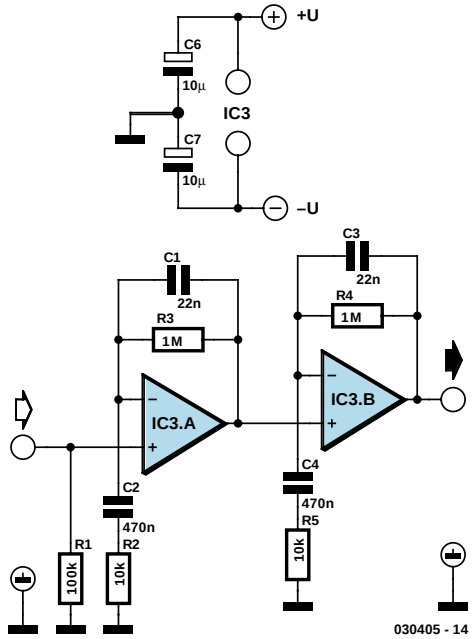


Figure 2. The I-Ging pulse generator uses high precision components.

Figure 4. The regeneration amplifier.

The medium

The choice of cable is vital in order to match the pulse profile to the characteristics of the transmission medium and thereby achieve the observed pulse acceleration. After much trial and error in the *Elektor Electronics* laboratory we concluded that a length of twisted-pair 100 Mbit/s Ethernet cable (CAT5) operated in the sub-hertz frequency range gave the most consistent results.

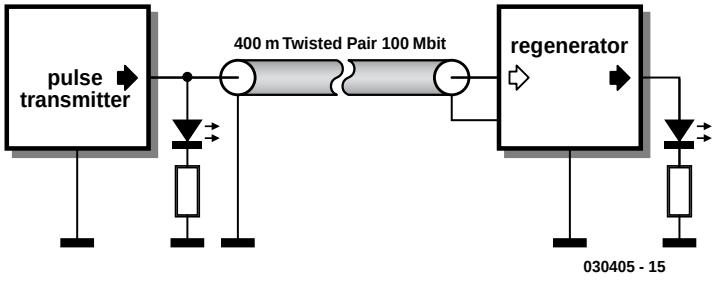
A cable length of 400 m was used in the prototype. In practice it is a good idea to cut it slightly longer so that it can be trimmed during testing to find its optimum length. The cable introduces a certain degree of attenuation to the low-frequency content of the pulse so the receiver circuit is provided with some amplification to compensate. The receiver circuit diagram is shown in **Figure 4** and uses a configuration similar to the transmitter. To operate reliably it is essential to ensure that the specified components are used throughout the circuit. The power supply can again either be single rail if you choose the OP290 opamp or split-rail for the TL082. Once the receiver has been carefully completed and checked we can begin with the experiment.

Back to reality

Figure 5 shows the test configuration for sending and measuring the high-speed pulses. With an oscilloscope connected to the transmitter output and another connected to the receiver output it can be clearly confirmed that when push button S1 is pressed, the transmitted pulse arrives at the receiver before it is detected as having left the transmitter! Even if you do not have an oscilloscope the effect is so pronounced that it can be demonstrated by substituting two simple LEDs at the measuring points and observing the events that occur when S1 is pressed. It is clearly observable that the receiver LED is lit slightly before the transmitter LED and then extinguishes before the transmitter LED! **Figure 6** shows the pulse waveforms as seen on a scope. It may be necessary to adjust the length of the cable slightly if the effect is not observed.

The revolution begins

The simple circuit described here replicates the findings of the original paper but only achieves a data rate of 0.5 b/s. To be truly useful this needs to be much faster, certainly in the region of Gb/s. To make the technology more practical a form of dynamically adaptive pulse shaping will also be necessary to compensate for the medium characteristics. Research into I-Ging is still in its infancy but these early results are so promising that we will undoubtedly see its adoption in more and more applications as the technology matures.



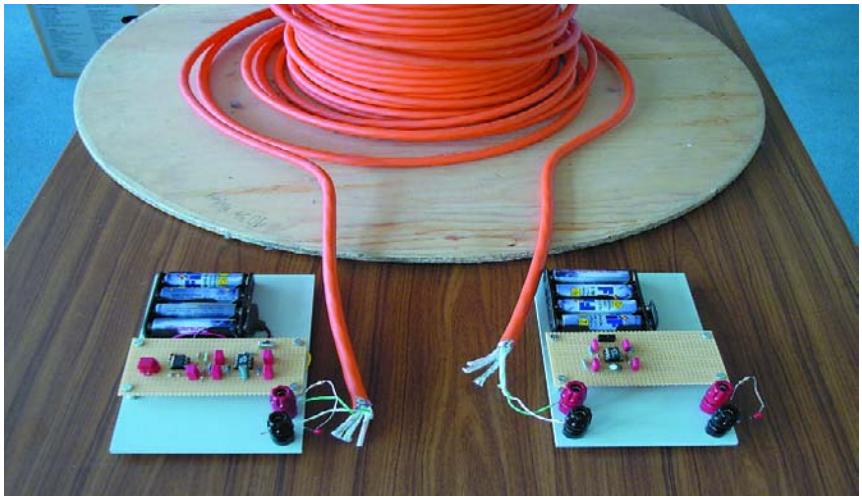


Figure 5. Test configuration with two LEDs to demonstrate the effect. A two channel scope can also be used in place of the LEDs.

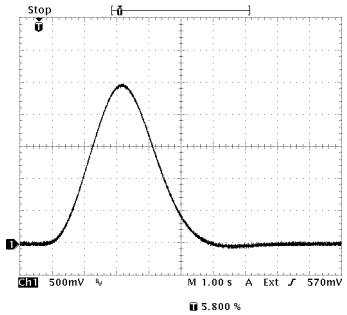


Figure 3. The transmission pulse.

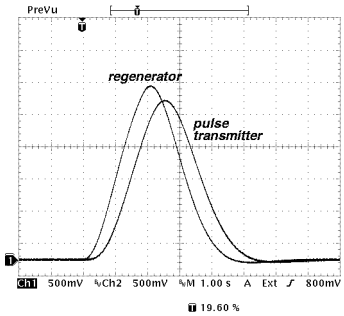


Figure 6. The impulse demonstrates localised temporal inversion.

The principle of causality is a cornerstone of our interpretation of the physical world but the observed negative propagation delay of approximately 0.4 s through the transmission medium and receiver combination seems to conflict with our understanding of this principle and will no doubt set alarm bells ringing for many *Elektor Electronics* readers, theoreticians and scientists.

In practice this technology would be used to send a stream of data at incredibly high speed rather than just a single pulse. The circuit was modified to send a pseudo random data sequence and the results are shown in **Figure 7**. The received signal shows very low levels of distortion and noise and more importantly still exhibits identical negative propagation delay through the medium.

The circuit described here is quite simple and we feel that many sceptics amongst you will not be silenced until they have built the circuit for themselves and witnessed what must surely be a new dawn for communication science and computing.

Can you explain the c+ phenomenon? If so, let us know as soon as you can so we can revert to the subject in next month's issue.

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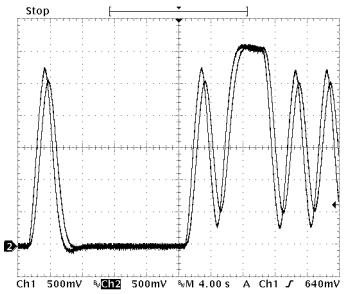


Figure 7. The technique applied to data transmission.

Reference

[1] http://lukin.physics.harvard.edu/Nature426_638.pdf

Burkhard Kainka

DREAM TEAM

Software DRM receiver

Low-cost digital radio receivers for short- and medium-wave are not yet on the market. There is, however, software available for processing DRM signals on a PC which can be used during the evaluation and changeover period. The two programs that have been developed differ in application, in features, and, not least, in cost.



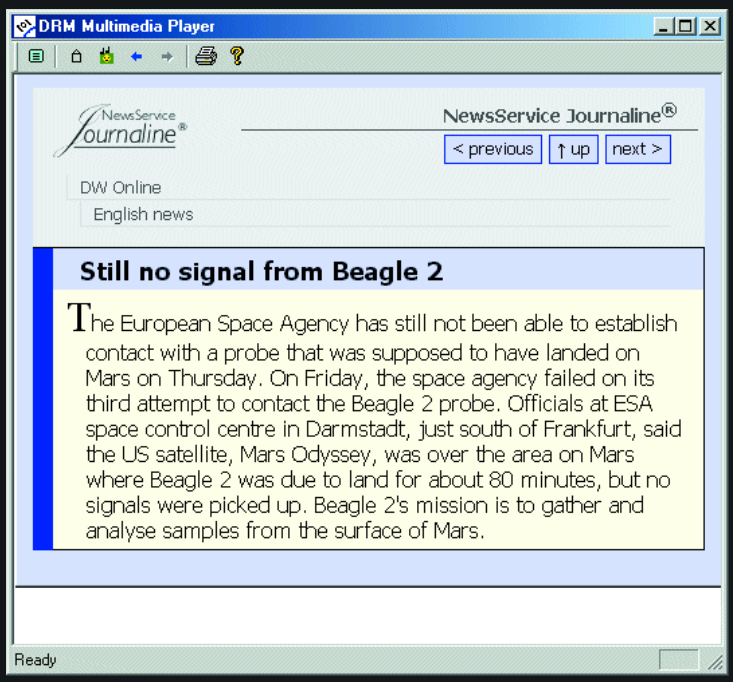


Figure 1. DRM can also carry text and images. Here is a example from the Deutsche Welle (German world service) news service, displayed using the DRM Software Radio program via a multimedia player.

Whether you use a modified world radio or the *Elektor Electronics* DRM receiver to surf the digital wave, you will need a software decoder running on a PC. This program takes the signal input to the sound card and decodes the data stream from it, which it returns as audio data to the sound card for output. There are at the moment two programs available: the 'DRM Software Radio' developed by Fraunhofer IIS, and the open source project 'DREAM', run by Volker Fischer and Alexander Kurpiers of the Communications Technology Institute at the University of Darmstadt.
A significant difference between the two programs lies in the requirements on the input signal. The standard intermediate frequency used is 12 kHz, putting the 10 kHz-wide band of the DRM signal between 7 kHz and 17 kHz. The DRM Software Radio allows a maximum deviation in the intermediate frequency of 500 Hz, whereas DREAM can decode DRM signals anywhere in the range from 0 kHz to 24 kHz.

DRM with text and images

'DRM Software Radio' is a commercial program and can be ordered online from www.drmrx.org for about 60 Euros. Each user is given their own software key and is automatically registered. The user can then take part in the DRM field trials and send in reception reports. Reception reports, especially from Europe, can certainly influence broadcasters' plans. The DRM Software Radio homepage gives information as to which stations can be received in which areas.
The function of the software as a DRM demodulator and decoder has already been described in the article 'Build your own DRM Receiver' project in the previous (March 2004) issue of *Elektor Electronics*.
DRM can carry more than just speech and music. As well as the 'audio service', information such as the name of the station and news headlines is also broadcast: these can be viewed as a scrolling or as a steady display. Some broadcasters also transmit images or other information using a format similar to the World Wide Web. The DRM Software Radio program makes these additional services available via a multimedia player. When a particular service is available, a click in the corresponding area in the window starts the player. After a delay while the requested data is collected, it is displayed. In this way, for example, the Deutsche Welle (German world service) transmitter in Jülich on 6140 kHz has recently started an additional service called 'Journaline', carried alongside the audio signal. A click in the audio window opens the multimedia player; in the background news summary information (in both English and German) is collected, and, after a delay, the pages of news can be viewed. This service works even when reception conditions mean that the audio signal breaks up: the news text still gets through, although it may take a little longer.
Figure 1 shows an example from the English-language 'Journaline' service.

Open Source Project

In contrast to tahe 'ready to run' DRM Software Radio, the DREAM open source project requires a little preparatory work. The authors have made the program available only as C++ source code (http://sourceforge.net/projects/drm/). Distributing the compiled version DREAM.EXE is not possible for copyright reasons, as patent-protected components are used. We shall look at these components individually, and present a step-by-step guide to compiling the project yourself, written by Thorsten Godau DL9SEC.
The complete project, along with advice on compilation and on the additional libraries required, can be found at www.tu-darmstadt.de/fb/et/uet/fguet/mitarbeiter/vf/DRM/DRM.html and at http://www.tu-darmstadt.de/fb/et/uet/drm.html.

The aim of the project was to develop an installable DRM software receiver with basic functionality. The project, in C++, was begun in June 2001 and version 1.0, which supports

Onward with fewer bits

Progress in digital audio

MP3

Way back in 1987 researchers at the Fraunhofer Institute started to develop an audio encoding method that made use of psycho-acoustic effects. The result was called **ISO-MPEG Audio Layer 3**, a.k.a. MP3 — and it became the standard for the 1990s. MPEG Layer 3 achieves CD quality in stereo at a data rate of **112-118 kbit/s**.

the new DRM standard using the FAAD2 library, has been available since 17 December 2003.
Although the software can be freely distributed under the GNU General Public Licence (GPL), that is not to say that there are no third-party rights attached to it. In certain countries use of the software by be in breach of patent law. The project is intended for those who want to find out how the DRM data stream is decoded, to learn about the software algorithms used, and then help to improve the software. If you only want to evaluate the quality of DRM transmissions, the authors recommend installing the commercial DRM Software Radio.

Microsoft Visual C++ V6.0 Service Pack 4 or Service Pack 5 is required for compilation, along with Trolltech QT 2.x. The following libraries are needed: FFTW, Qwt and FAAD2. We shall now look at these components in more detail:

Qwt (Qt Widgets for Technical Applications) is a library of GUI components, including graph plotting and controls. The entire C++ source code is independent of the underlying operating system and is used so that DREAM can run under Windows as well as under Linux. The library can be downloaded from http://qwt.sourceforge.net/.

FFTW (the Fastest Fourier Transform in the West) was developed at MIT by Matteo Frigo and Steven G. Johnson. The package can be downloaded from http://www.fftw.org/. DREAM uses the fast Fourier transform to analyse the individual carriers in the DRM signal in terms of amplitude and phase. The data obtained from these carriers is assembled to form the complete data stream, which includes both audio and multimedia components.

FAAD2, by the Dutch company AudioCoding.com, includes the DRM-specific algorithms for decoding the received digital data. AAC (Advanced Audio Coding) is the ISO high quality audio coding algorithm developed by leading companies AT&T, Dolby Laboratories, Fraunhofer IIS and Sony, which is also used in the Fraunhofer IIS DRM Software Radio. MPEG-2 and MPEG-4 AAC are also implemented. Version 2 also implements decoding of HE (High Efficiency) AAC streams. FAAD2 may be installed for private and scientific use under the GNU General Public Licence, although of course any patent restrictions must be observed. This is one of the reasons that a compiled version DREAM.EXE cannot be distributed without restriction. It is also not certain whether free use of FAAD2 will still be permitted as DRM is developed further. We should nevertheless warmly welcome this opportunity for those interested in DRM to gain access to this new technology.

Here are Thorsten Godau's comments on how to proceed.

The first requirement for compiling the DREAM source code for Windows is a working installation of:

– Microsoft Visual (Studio) C++ V6.0, and **either**
– Visual C++ Service Pack 4 and Processor Pack for SP4, **or**
– Visual C++ Service Pack 5 and Processor Pack for SP5

Then proceed as follows:

– **Download** the non-commercial version of QT (QT-Win V2.3 NC) from Trolltech at http://www.trolltech.com/download/qt/download_noncomm.html and install it (use the standard paths).

– **Download** the source code of DREAM V1.0 (or higher) from http://prdownloads.sourceforge.net/drm/drm_1_0.zip?download (select one of the mirrors).

– **Download** the precompiled FFTW package from http://www.tu-darmstadt.de/fb/et/uet/fguet/mitarbeiter/vf/DRM/download/WinFFTWInst.zip.

– **Download** the precompiled Qwt package from http://www.tu-darmstadt.de/fb/et/uet/fguet/mitarbeiter/vf/DRM/download/WinQWTInst.zip.

– **Download** the FAAD2 package from http://www.tu-darmstadt.de/fb/et/uet/fguet/mitarbeiter/vf/DRM/download/WinFAAD2SBRNewInst.zip.

Create directories and copy in the precompiled files:

– **Create** a directory called, for example, C:\vcprojects
– **Open** drm_1_0.zip and unpack it into subdirectories of C:\vcprojects.
– **Open** WinFFTWInst.zip. Copy the files libfftw.lib, fftw.h and rfftw.h from the subdirectory \lib into the directory

AAC
In 1997 the cooperation between the Fraunhofer Institute and large companies like AT&T, Sony and Dolby culminated in the **MPEG-2 Advanced Audio Coding** (AAC) method. AAC achieves high quality thanks to 1-48 audio channels and sampling rates between 8 kHz and 96 kHz.

mp3PRO
Using **Spectral Band Replication Technology** (SBR) developed by Coding Technologies the data reduction efficiency can be doubled without loss of quality. mp3PRO is the **MP3-compatible** implementation of SBR. It is for available as a plug-in for the current version of the famous Nero Burning ROM software.

aacPLUS/MPEG-4 HE AAC
The combination of AAC with bandwidth-extended SBR was originally marketed as **aacPLUS** by Coding Technologies. Having been standardised by MPEG as a one of the MPEG-4 audio profiles, the system became known as **MPEG-4 High Efficiency** (HE-AAC). This recently introduced standard allows full CD quality in stereo at 48 kbit/s, very good stereo quality at 32 kbit/s and 'parametric' stereo in FM-radio quality at just 20 kbit/s. The latter is employed for DRM since December 2003.
MPEG-4 HE AAC is also available as a plug-in for Nero Burning ROM.

C:\vcprojects\drm\libs.
– **Open** WinQWTInst.zip. Copy all the files (.lib und .h) from the subdirectory \lib into the directory C:\vcprojects\drm\libs.
– **Open** WinFAAD2SBRNewInst.zip and unpack it into subdirectories of C:\vcprojects.

Compile the FAAD2 package:
– Go to the directory C:\vcprojects\faad2\libfaad and double-click on libfaad.dsw (if Visual Studio is correctly installed the Visual C++ IDE will now open).
– In the IDE, select 'Build/Set Active Configuration' and in the window that appears select 'libfaad - Win32 Release', and then click 'OK'.
– In the IDE, select 'Build/Rebuild All'. Compilation begins, and should complete with 43 warnings, but no errors.
– Now, go to the directory C:\vcprojects\faad2\libfaad\Release and copy libfaad.lib into the directory C:\vcprojects\drm\libs. Likewise, copy faad.h from c:\vcprojects\faad2\include into the directory C:\vcprojects\drm\libs.

Build the QT MOC files:
– Go to the directory C:\vcprojects\drm\windows and double-click on MocGUI.bat to launch it. A number of files will automatically be created in the directory C:\vcprojects\drm\windows\moc.

Compile DREAM:
– Go to the directory C:\vcprojects\drm\windows and double-click on FDRM.dsw (if Visual Studio is correctly installed the Visual C++ IDE will now open).
– In the IDE, select 'Build/Set Active Configuration' and in the window that appears select 'FDRM - Win32 Release', and then click 'OK'.
– In the IDE, select 'Build/Rebuild All'. Compilation begins, and should complete with one linker warning, but no errors.

Run DREAM:
– Go to the directory C:\vcprojects\drm\windows\Release. Copy DREAM.EXE from here to a newly-created directory C:\Program Files\Dream.
– If DREAM.EXE is to be run on a computer where QT is not already installed, first copy the file qt-mt230nc.dll from the directory C:\qt\bin into C:\Program Files\Dream. Then run DREAM.EXE.

It's as simple as that.

Results

A click on DREAM.EXE starts the program. Once the receiver has been tuned to a suitable frequency, an audio signal will be produced after a short delay. The station being received and transmission information will be displayed on the screen (**Figure 2**).

Further information is available under 'View/Evaluation Dialogue'. This mode (**Figure 3**) shows the DRM spectrum and a large quantity of additional information including the current signal-to-noise ratio (SNR), the bandwidth, and the operating mode. It is also possible to experiment with a range of software options. A broken red line indicates the measured centre frequency of the DRM signal, which simplifies calibrating the receiver. DREAM is not restricted to a 12 kHz fre-

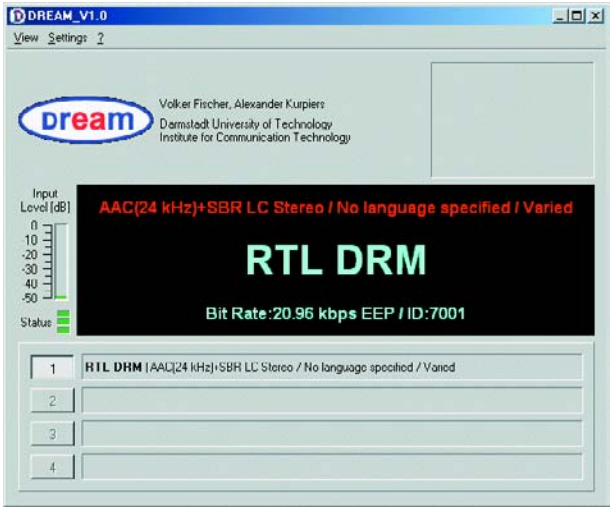


Figure 2. DREAM receiving RTL DRM.

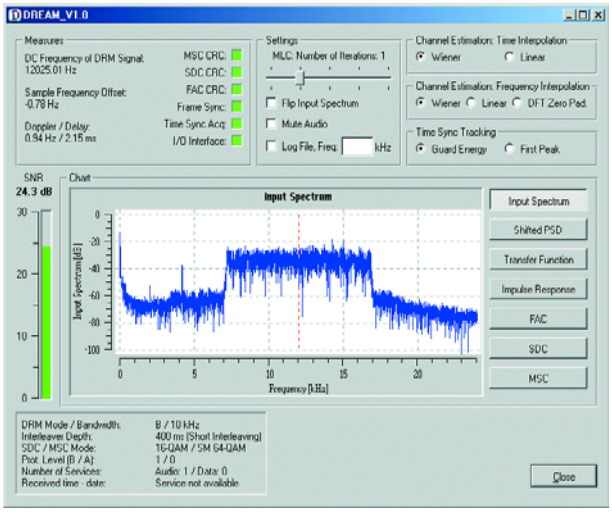


Figure 3. The DREAM evaluation dialogue displays the most important parameters.

quency band, but rather accepts the entire input band between 0 kHz and 24 kHz. This relaxes the constraints on the receiver and allows the receiver to be detuned by a few kilohertz, for example to avoid an unwanted interfering carrier on the image frequency.

AM reception

The receiver option Settings/Receiver mode/AM (analogue) allows DREAM to receive amplitude-modulated signals from normal AM broadcasters. DREAM looks for a strong carrier in the input spectrum and subtracts its frequency from the upper sideband of the received signal. The receiver thus operates as a single sideband (SSB) receiver. This allows both SSB, CW and utility signals such as weather fax and RTTY to be decoded.

In order to receive an AM station, first ensure that the frequency band to be received sits in the middle of the filter bandwidth around 12 kHz. Use DRM mode to search for the AM station: the carrier should stand out clearly from the spectrum at 12 kHz. Then switch to AM mode. In the evaluation dialogue the broken red line again indicates the detected carrier frequency at around 12 kHz. This setting remains fixed as long as AM mode is active. The receiver can therefore be retuned freely to bring in other stations.

The single sideband demodulation process gives a number of advantages over the conventional envelope detector. Selective fading normally causes severe distortion if the carrier is significantly attenuated. DREAM, however, can offer trouble-free reception in these conditions, although the sound quality may vary slightly. Also, DREAM includes a very good low-pass filter. The input signal is analysed by an FFT, the upper sideband moved down to zero frequency, and then the audio sig

nal re-synthesised by an inverse FFT covering the band from 0 kHz to 5 kHz. When the result compared with a conventional short-wave radio tuned to the same frequency, the delay introduced by the digital processing and data buffering is clearly apparent.

Single-sideband (SSB) reception also offers the possibility of obtaining better reception from severely distorted AM stations. Since short-wave broadcast channels are at multiples of 5 kHz, it is often the case that there are pairs of stations just 5 kHz apart: the upper sideband of one then completely overlaps the lower sideband of the other. With DREAM it is possible to receive only the undistorted sideband. In some cases this will mean inverting the input spectrum (option ‘Flip Input Spectrum’ in the evaluation dialogue). This is also required in order to receive an SSB transmission using the lower sideband. When the *Elektor Electronics* DRM receiver is being used, fine adjustment of its intermediate frequency is possible using the program DRM.EXE. Reception results are better than when using direct conversion, since the software completely discards the unwanted sideband.

Conclusions

DREAM_V1.0 is certainly a serious alternative to the DRM Software Radio. The program is absolutely stable in use and requires less processor power than the earlier versions. It is possible to receive images, and the program can create a log file with reception results. As already noted, DREAM is very tolerant of the position of the DRM baseband signal within its input bandwidth, scanning the entire range from 0 kHz to 24 kHz. It also offers an analogue AM reception mode, allowing it to be used with the *Elektor Electronics* DRM receiver for standard broadcasts.

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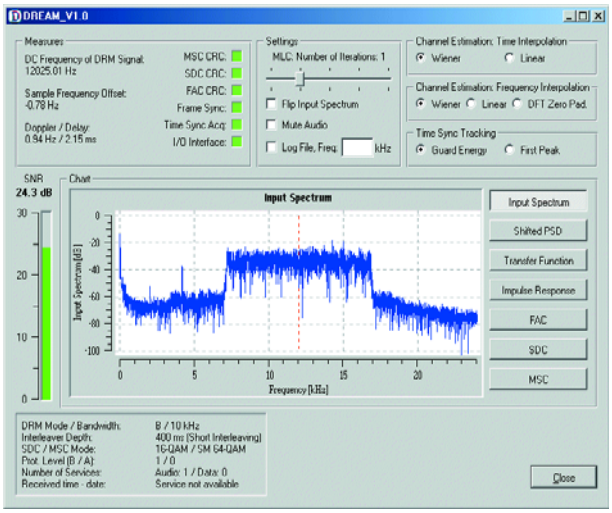
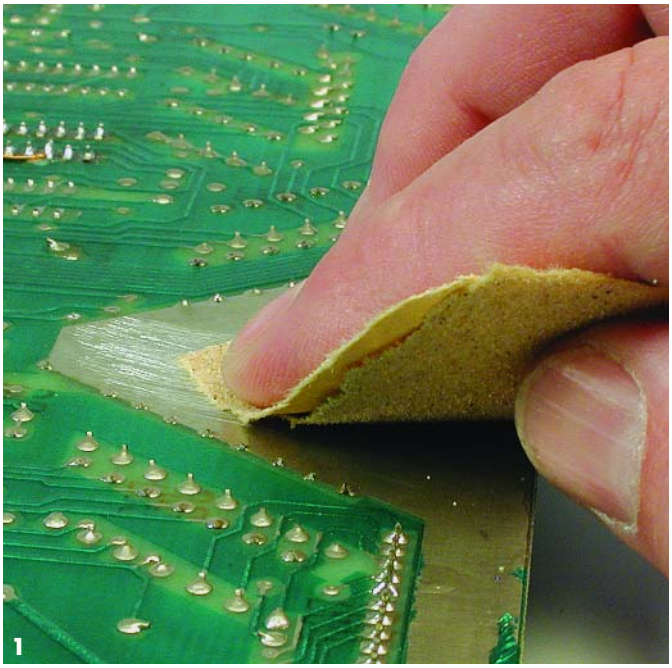
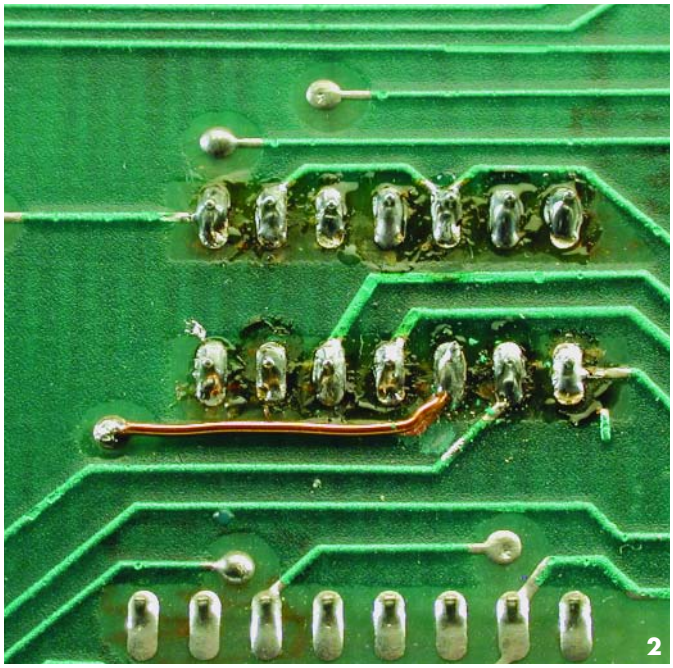


Figure 4. In contrast to the DRM software radio, DREAM can also demodulate normal AM transmissions on the PC.





Repairing the damage caused by leaking batteries

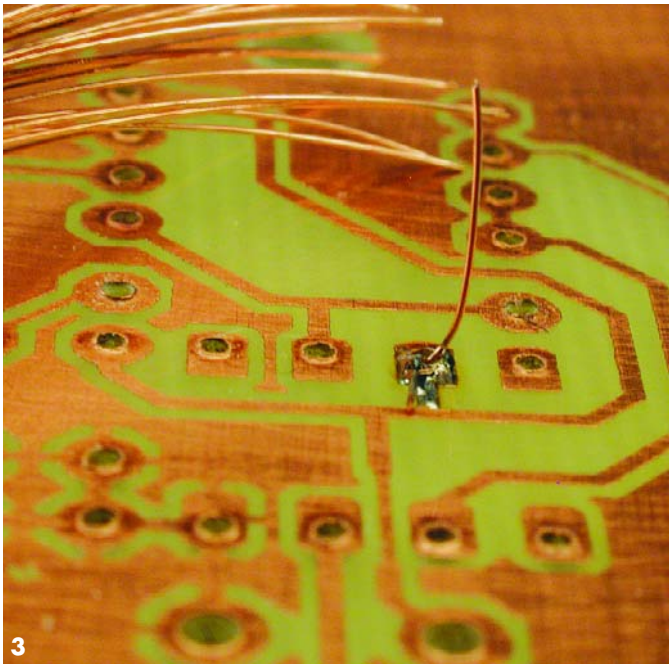
Most of you will have suffered from this at one time or another: a torch or other battery-powered equipment has been neglected too long and the batteries started leaking. After all, these cells contain a cocktail of corrosive chemicals that will eventually find their way through the battery housing and end up in all those places where they're not wanted.
NiCad cells that are overcharged can vent the electrolyte in gas form. The metal parts in particular will be damaged by this, with disastrous consequences.
It becomes even worse when the batteries, whether in a holder or not, are mounted on a PCB. This can be the case with (older) microprocessor boards, which have, for example, a real-time clock that has to keep running, or a volatile memory that stores settings that have to be maintained when the power is turned off or lost. A leaking battery on a PCB leads to corrosion, which has both visible and invisible consequences.
When repairing this it is pointless just to polish those areas that are visibly affected; any repair has to stand the tests of time. The electrolyte can penetrate into certain materials and continue with its destructive process underneath the surface. Just cleaning the surface is therefore not enough; the only remedy is the complete removal of every bit of electrolyte.
On a PCB the corrosion can be seen in various ways. Firstly as a simple mark where the battery is mounted. Next, it can be seen as a green or white deposit on the pins, components and solder joints.

We start with the removal of all components that are on the affected part of the PCB. Use a good quality solder sucker and/or desoldering braid and keep in mind that in many cases, such as for resistors, capacitors, IC sockets and standard logic ICs, the PCB is more important than the component that has to be removed. In those cases, you can cut the connections and remove them one by one.
One problem that occurs when desoldering is that solder affected by corrosion often won't flow properly. In that case you should add some fresh solder and try it again. If you are still unsuccessful, you should sand the joint until the corrosion has gone, and try again with the solder sucker. In extreme cases you may have to use a PCB drill to restore the connections.

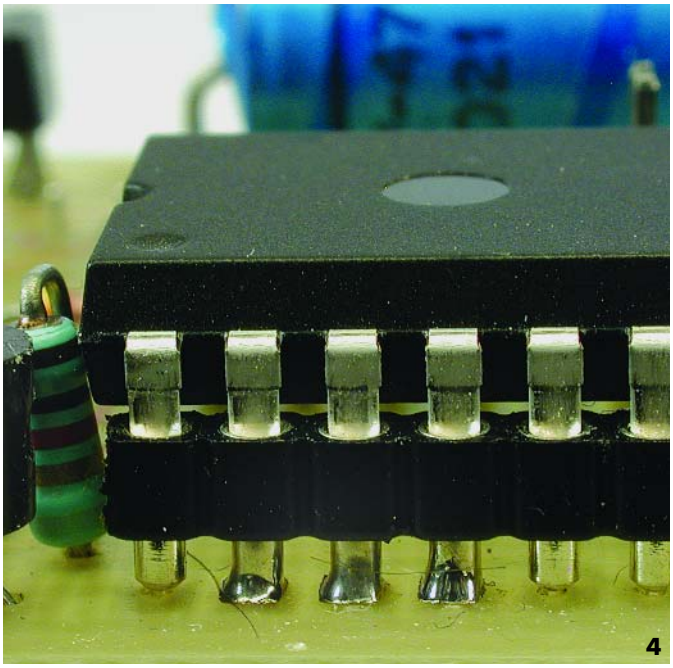
When all components have been removed from the corroded part of the PCB, it should be sanded with fine wet-and-dry sandpaper. Any solder resist should be removed as well; you just keep sanding until all tracks become shiny again.

To finish the job properly any remaining electrolyte has to be neutralised by rinsing the PCB in a 1:1 mixture of water and vinegar. It is often thought that all batteries contain acids, but Nicads and alkaline cells have KOH as electrolyte, which is an alkaline. Hence we need to use an acidic solution to neutralise it. After this it should be rinsed with water, dried and then cleaned with alcohol (white spirit is much cheaper, but less pure). The PCB should then be sprayed with a solder lacquer, which protects the tracks from oxidation and lets the solder flow cleanly when the PCB is repopulated.
The PCB now has to be inspected carefully, and any broken tracks have to be repaired using enamelled copper wire, cut to length. The components can then be soldered back onto the PCB.
It's quite a time-consuming job, but if it is done properly the PCB should give many years more service, unless the batteries start leaking again...

(040025-1)



3



4

Through-hole 'plating' double-sided PCBs

Making a double-sided PCB is far from easy. It is quite a task to make sure that the artwork films with the tracks and pads are perfectly aligned on both sides. And it becomes even more difficult when both sides have to be connected at various points.

The professional PCB manufacturer uses through-hole plating: the board is drilled and a thin layer of copper is deposited in the holes that require a connection between the top and bottom side. An electrochemical process is then used to make the actual connection. This however requires specialist equipment, which won't be available in many electronics labs. Fortunately there are other methods, although they are somewhat laborious. These methods are also useful when repairing damaged double-sided boards.

The simplest solution is to solder the pins or leads of the components on both sides of the board. This is easily accomplished with conventional components (resistors, diodes, transistors, etc.), but with radial capacitors and ICs (in sockets) this is more difficult.

The second method uses Litz wire, which is often on hand in the electronics workshop. This wire consists of many thin copper strands, which are highly suitable for making connections between two sides of the board and still leave enough room in the holes for the component leads (**Figure 3**). The thin strands are soldered on the component side of the board, taking care that the solder joint doesn't rise above the board too much, and that the hole doesn't become blocked by the solder. When all connections have been made, the components can be placed on the board and soldered on the solder-side. We recommend that you use two strands per connection, thereby reducing the risk that a connection is damaged when the component pin is stuck through the board.

For DIL-IC sockets there is a much simpler method: there are contact strips that have the same type of contacts as turned-pin IC sockets. These have the advantage that they are raised slightly further above the board, providing enough room to solder the pins on both sides of the board (**Figure 4**). These strips are preferable to ordinary turned-pin sockets because the pins on the component side of the board are easier to get to with a soldering iron from both sides.

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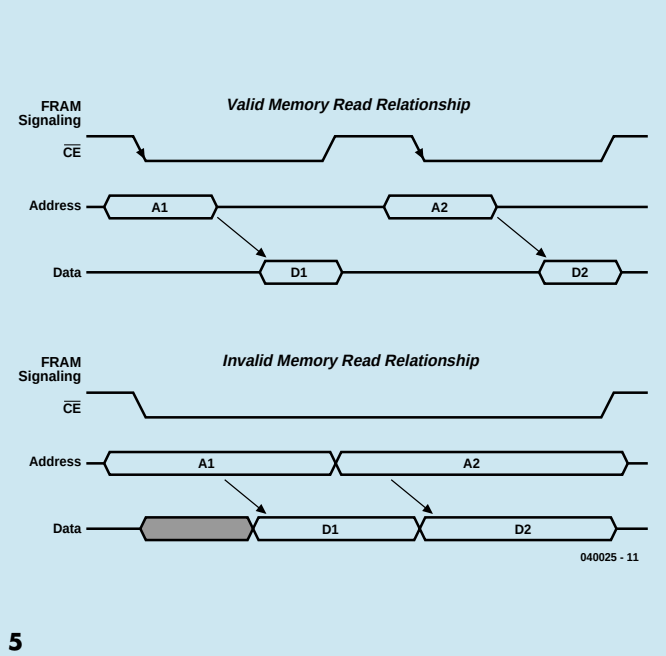
Alternatives for battery-backed RAM

Many microprocessor boards include a static RAM that stores a variety of system settings. It is often desirable to keep these settings when the power supply is turned off or interrupted. For this reason a backup battery is mounted on these boards, which provides standby power to the SRAM.

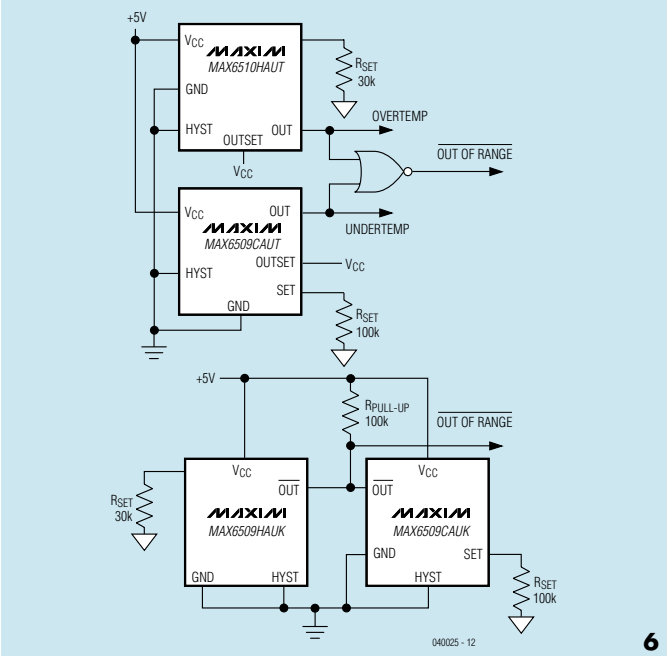
At first sight, this seems a simple solution, but it is far from ideal. Batteries take up a fair amount of space on the PCB. They also need to be checked regularly for any possible leakage and that they supply the correct voltage. On modern processor boards a flash memory or EEPROM is often used, neither of which require a backup battery. Older boards can be given a new lease of life by replacing the SRAM and batteries with a modern alternative such as FRAM.

Ramtron International Corporation has introduced two ICs, the FM1608 and FM1808, which are ideal for removing the need for backup batteries in older systems, and which are of course very useful in new designs too. These are a special type of non-volatile memory, Ferroelectric RAM, which is being developed further by the manufacturer. A detailed description of the technology can be found on their website at www.ramtron.com/aboutfram. The manufacturer guarantees at least 10 billion read/write cycles and a 10-year data retention for these ICs. The best characteristic of these memories is that they are pin compatible with standard 8 K x 8 (FM1608) and 32 K x 8 (FM1808) SRAMs and EEPROMs, and can therefore be used in existing designs without having to make any drastic modifications.

There is just one aspect of the design that has to be taken into account when a 'normal' SRAM



5



6

is replaced by an FRAM: the $\overline{CE}$-signal of an FRAM is used to clock an internal address latch, as shown in the timing diagram. This means that the IC requires a falling edge on this pin after a valid address has been put on the address bus, otherwise the address won't be recognised!

(040025-3)

Temperature switches in SOT packages

The MAX6509 and MAX6510 made by Maxim are very small temperature switches, which can be configured with a single external resistor to trip at any temperature between -40°C and +125°C. The accuracy of the trip point is typically ±0.5°C and ±4.7°C maximum. They have a pin that sets the hysteresis to either 2°C or 10°C.
The MAX6509 has an open-drain output. The MAX6510 uses the OUTSET pin to choose between an output that is active high, active low, or an open-drain with a pull-up resistor. The output could for example drive a reset or interrupt to a microcontroller system, switch a supply or activate an external alarm. The current through these components should be kept as small as possible, to limit any temperature variation due to internal dissipation.

With these components you should pay particular attention to the suffix (the letters immediately following the part number) and specifically the first letter!

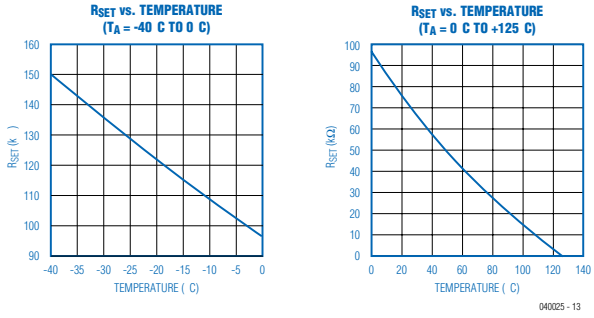
An 'H'-type ('Hot') switches the output when a rising temperature exceeds the trip point and switches back when the temperature falls below the trip point, **minus** the hysteresis. A 'C'-type ('Cold') in contrast will switch when the temperature falls below the trip point and switches back when it rises above the trip point **plus** the hysteresis. In other words: the hysteresis is either above or below the trip point, depending on the type of sensor.
By combining both types in one design, it is very simple to monitor a temperature between lower and upper limits. Two possible configurations are shown in **Figure 6**. The lower circuit uses the open-drain outputs of the MAX6509 as a wired-OR, with a common pull-up resistor. All of the temperature switches in this example have a hysteresis of 2°C (input HYST connected to ground).
When the trip temperature has been chosen, the corresponding value for R_{set} can be read from one of the graphs in **Figure 7**. For an exact value, one of the following formulae should be used.

For temperatures between –40°C and 0°C use:
$R_{set} = [(1.3258 \times 10^5) / (T+1,3)] - 310.1693 - [(5.7797 \times 10^6) / T+1.3)^2]$
For temperatures between 0°C and +125°C use:
$R_{set} = [(8.3793 \times 10^4) / T] - 211.3569 + [(1.2989 \times 10^5) / T^2]$

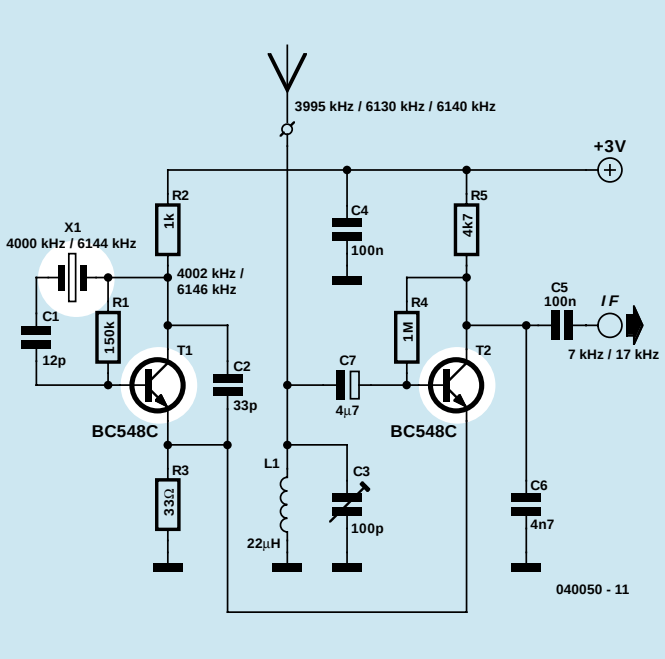
where T is in K, R_{set} is in kΩ.

From the graphs you can determine that the switches in the examples have trip points of about 0°C (R_{set} = 100 kΩ) and 65°C (R_{set} = 30 kΩ).

(040025-4)



7



2 Transistors, 1 Crystal — that's it!

Poor Man's DRM is here

Burkhard Kainka

Almost too good to be true but tried & tested: a totally undemanding receiver giving you an opportunity to experience DRM digital broadcasts on shortwave. Digital SW almost for free!

There are a large number of shortwave stations broadcasting not only conventional analogue signals but also digital programme material. This raises the exciting question if there might be stations that 'fit' standard quartz crystal frequencies and so can be captured using the ultra-simple receiver presented in this short article. Ultra-simple? Yes, the receiver consists of nothing more than a direct mixer and a crystal oscillator.

As you could have surmised there are off the shelf crystals that fit the bill (else there would not have been an article). Three shortwave frequencies used by Deutsche Welle (German world service), 3995 kHz, 6130 kHz and 6140 kHz, match the common microprocessor quartz crystal frequencies of 4.000 MHz and 6.144 MHz — that is, with a trick! In each case, the oscillator should run about 2 kHz above the nominal crystal frequency. In many cases, a crystal frequency can be 'pulled' a little by using a load capacitance which is slightly larger than the usual 20 to 30 pF (assuming the crystal operates at fundamental resonance). In our circuit a series capacitor of 12 pF does the job, forcing the crystal to resonate a little above its nominal frequency. It is not terribly important if the actual frequency is 4002 kHz or 6146 kHz.

Inside the receiver the digitally modulated signal received at the DRM station frequency is mixed down into the audio range. This simple operation causes the DRM baseband (which is still digitally modulated) to appear at 7 kHz for the station frequencies 3995 kHz and 6140 kHz, or 17 kHz for the transmitter at 6130 kHz. Decoding these signals on the PC will only succeed if you use the DREAM software for DRM, because this program is tolerant of input frequencies between 0 and 24 kHz. Luckily, DREAM is an open-source program hence does not incur any costs.

It should be noted that the received DRM baseband will need spectrum-inversion. This is because high-side injection is used, that is, the local oscillator operates at a frequency above the station frequency. In the program, activate the 'Flip Input Spectrum' option.

Oscillator and mixer

From a point of view of modern electronics the lower parts of the shortwave bands can hardly be called 'high frequency'. Consequently there are no problems building the receiver from AF / fast switching transistors like the ubiquitous BC548C or BC549C. The oscillator around T1 feeds its output signal directly onto the emitter of mixer transistor T2. Mixing takes place on the curved part of the transistor characteristic. The base gets the RF signal directly from a heavily damped hence wideband input circuit that's tuned for maximum RF with the aid of a trimmer. The upshot is that the DRM baseband is available at the collector. Depending on the available level, the DRM signal coupled out via C5 is fed to the Line or Microphone input of your soundcard for processing by the DREAM software.

Under favourable circumstances (including propagation and the absence of man-made noise), an indoor wire antenna of about 3 metres will be sufficient. Better results are achieved with an outdoor antenna of about 10 m. The Deutsche Welle (DW) transmitters put up usable signal strengths across most of Western Europe. However, if DREAM starts to indicate a poor signal/noise (S/N) ratio (less than 15 dB), look for wideband noise as a cause for signal corruption. Under good conditions, our little receiver supplies an S/N greater than 20 dB. The DRM broadcast schedule on DW may be gleaned from, among others, Stefan Mahn's website at www.drm/info.de

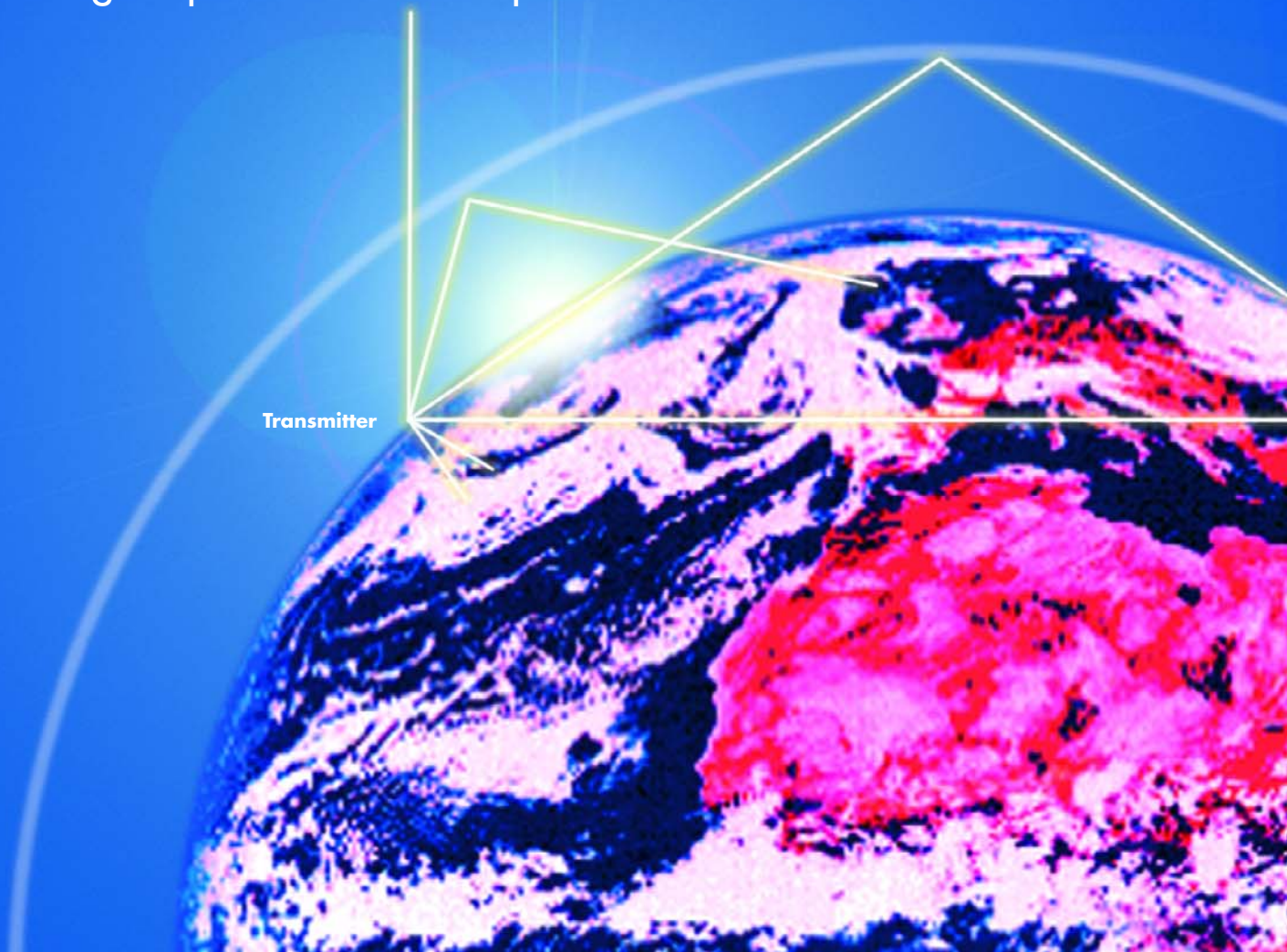
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Burkhard Kainka

WAVECATCHER

Aerials and preselectors for AM and DRM

For initial experimenting and radio listening with a DIY short-wave or DRM receiver, a simple rod aerial or piece of wire is sufficient. But it's only natural to want something better. This article gives you the right tips for better reception.



In the UHF band, radio waves propagate essentially in a line-of-sight manner, similar to light, and this limits their reception to a range of around 100 km, depending on aerial height. However, radio waves at frequencies below 30 MHz show completely different behaviour, which makes a significantly larger reception range possible. Of course, the complex propagation mechanisms in this frequency range also lead to special problems, such as dependence on the time of day, field strength fluctuations (fading) and selective fading.

Radiation and propagation

The decisive role in the propagation of radio waves in the short-wave bands is played the ionosphere, a set of weakly conductive layers in the atmosphere at great height that are formed by solar ionization. The ionosphere results from collisions between particle radiation and gamma radiation from the sun and air molecules, which become ionized with the release of electrons. Within certain frequency bands and at certain incident angles, the free electrons generated in this manner act like a mirror. At large incident angles and high frequencies, the ionosphere is instead transparent.

A short-wave transmitter can be received in the nearby region via ground-wave propagation over a range of only around 30 to 100 km, depending on the height of the aerial. At greater distances, the transmitter disappears behind the horizon due to the curvature of the earth, making a direct link impossible. However, radio waves reflected from the ionosphere can reach receivers located in a region starting at certain minimum distance away for the transmitter, as illustrated in **Figure 1**.

Between the limit of reception of the ground wave and the start of reception of the reflected signal, there is a region called the 'skip zone' where the signal cannot be received. The reflection angle that would be necessary for the signal to be received in this region is too large. Of course, the transmitter also radiates energy at this angle, but it leaves the atmosphere without being used, perhaps to be received sometime later on by one of our neighbours in outer space.

The minimum distance between the transmitter and the receiver varies with the time of day for each frequency, and it also depends on the level of solar activity, which varies over time.

High frequencies can only be reflected at very shallow angles. Consequently, the ground distance spanned by the reflected signal is generally greater in the higher-frequency bands. The skip zone is also correspondingly larger; during the day it ranges up to around 200 km at 6 MHz and around 1000 km at 15 MHz.

The skip zone expands at night, and with it the reception range. As a result, it often happens that a particular transmitter can be initially received quite well near dusk but then suddenly disappears, since in a manner of speaking it has slid into the skip zone. If the same programme is also being broadcast on other frequency bands (as is customary with BBC, DW, etc.), it is recommended to change to a lower frequency when this happens.

Generally speaking, radio waves arrive at a receiver via more than one path. The differences in the path lengths give rise to phase differences, which lead to partial reinforcement or cancellation of the waves. Especially in the short-wave band, rapid fluctuations in field strength are common. This frequently causes selective fading, which is particularly noticeable with AM transmitters in the form of unpleasant distortion resulting from nearly total loss of the carrier signal, causing it to be overshadowed by the sidebands. DRM (the new digital transmission technique) is also affected by this fading, but the modulation and coding methods used for DRM are especially robust and can tolerate partial loss of data. Thanks to effective error handling, even deep dropouts in the DRM spectrum, such as those caused by cancellation at certain frequencies, generally do not interfere with reception.

DRM restores excitement to radio listening on the medium- and short-wave bands. There are already quite a few stations available (see **Table 1**), and new transmitters are constantly being added to the list.



Figure 1. Signal reflection and the skip zone for propagation in the short-wave bands.

A longish wire

Strong short-wave transmitters can be received using an 'aerial system' (rod aerial or piece of wire) with a length of less than a metre. For long-distance reception, it is naturally much better to use a 'longish' wire aerial, which preferably should be strung up outdoors – as widely separated from other objects as possible, as high as possible and sufficiently far away from the house, in order to avoid the 'noise cloud' emanating from the house.

In theory, a wire aerial has a resonant frequency at one quarter of the signal wavelength, although a good earth connection acting as a counterpoise is important for this. In practice, wire aerials with lengths of around 10 metres have proved to be satisfactory.

If the receiver is located close to a window or the outside wall of the house, it is sufficient to connect the end of the wire directly to the inner contact of the aerial socket. However, if the distance between the aerial and the receiver inside the house is relatively long, the connection should be made using coaxial cable, with the opposite pole being provided by an earth connection close to the feed point of the aerial (see **Figure 2**). Here it makes no difference whether you use 50-Ω cable or 75-Ω cable. After all, the base resistance of the aerial varies with frequency, and it has a complex impedance with

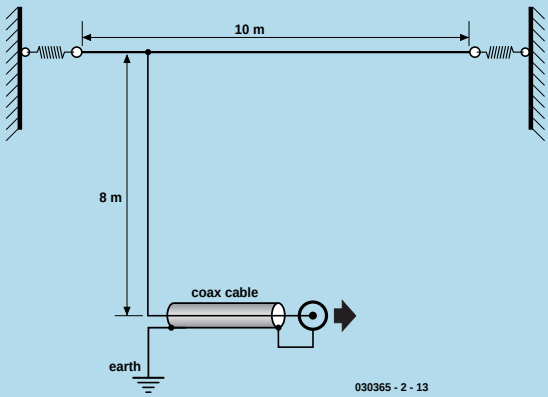


Figure 2. An outdoor wire aerial connected via a coaxial cable.

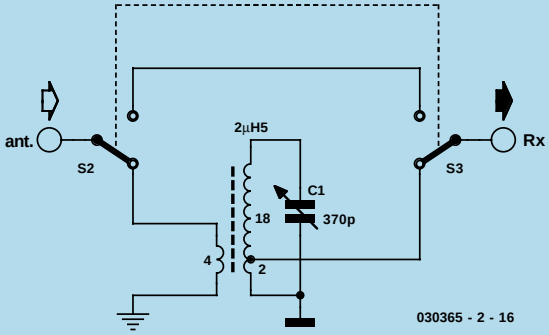


Figure 3. A freely tuneable resonant circuit used for preselection.

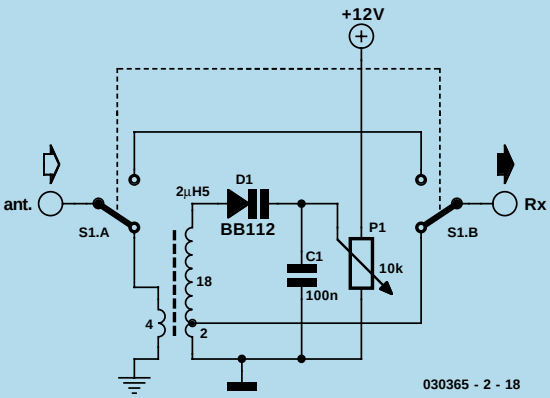


Figure 4. Tuning using a variable-capacitance diode instead of a rotary variable capacitor.

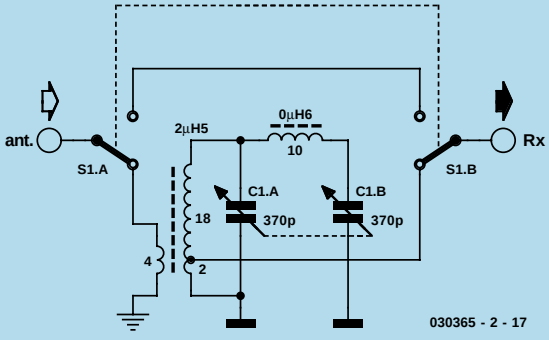


Figure 5. Tuning from 3 MHz to 30 MHz using a dual variable capacitor.

alternating capacitive and inductive components. The coaxial cable also has its own resonances, since it is not being used at its characteristic impedance, and it transforms the aerial impedance, with the net result that resonances other than those to be expected from the length of the aerial can also occur. However, this does not have much of a noticeable effect at the receiver, since signal level variations of around 10 dB hardly matter with DRM.

Outdoor wire aerials are normally made from stranded wire with a sufficiently large cross-sectional area, in order to obtain good mechanical strength as well as low ohmic losses. A lead from a standard mains cable or loudspeaker cable with a cross-sectional area of 0.75 to 1.5 mm_ is suitable for this purpose. It is also possible to use significantly thinner wire. A test using 10 m of thin magnet wire (0.3 mm diameter) yielded usable results for DRM reception, and it has the advantage of being quite inconspicuous.

If you shy away from constructing your own aerial, in many cases you can make use of existing systems or cables. A typical aerial system provides not only television and UHF signals, but also the entire AM range from long-wave to short-wave. It's certainly worth trying.

In many cases, better results can be obtained with a rooftop aerial than with an indoor aerial. Old rooftop aerials are often no longer in use, but the aerial cables leading to the roof are still in place. The cable by itself can also be useful. Such a 'forgotten' aerial cable that has been converted into a 'single lead', which means with the inner conductor and the screen shorted together, forms a usable vertical aerial. The cable usually runs all the way to the roof of the building and thus reaches a greater height than a horizontally strung wire aerial. Especially at relatively high frequencies (above 15 MHz), better results can be achieved using such an aerial than with an outdoor wire aerial.

Preselection

A 'longish' wire aerial has a broadband characteristic and receives the entire AM radio band between 0.5 MHz and 22 MHz relatively uniformly, so no additional tuning is need for the aerial itself. However, preselection is worthwhile if reception is degraded by mirror frequencies.

Practically every receiver that works on the superhet principle has two reception frequencies: the intended frequency and the mirror frequency, which is separated from the intended frequency by twice the intermediate frequency. With the usual IF of 455 kHz (as used in the DRM receiver published in the March 2004 issue), this means the mirror frequency is located 910 kHz above the tuned frequency. For receivers having a switching mixer (such as the diode-ring mixer of the DRM receiver), the received signal is mixed with not only the fundamental frequency of the oscillator signal, but also (with a certain amount of attenuation) with all odd harmonics of the fundamental oscillator frequency. Primarily in case of reception in the medium-wave band, this can lead to interference from mixer products formed by harmonics of the mixer oscillator and strong short-wave signals. Consequently, a medium-wave preselector often provides significant improvement.

A preselector connected between the aerial and the receiver is most commonly used, and such preselectors are generally tuneable. Suitable preselectors are available from specialist (amateur) radio shops, but you can also build your own.

Tuneable

The standard approach to building a preselector is to use a tuneable resonant circuit (**Figure 3**). The coil can be wound as an air-core coil, which means without using an actual core. The wire diameter is not all that important for such air-core coils. For small coils, you can use enamelled copper wire with a diameter of 0.3 to 0.7 mm; somewhat heavier wire should be used for larger coils for the sake of mechanical stability.

A coil having a diameter of 8 mm and 20 turns over a length of 10 mm has an inductance of 2.5 µH. In combination with a 370-pF variable capacitor, it has a lower resonant frequency of approximately 5 MHz. This circuit can thus be tuned across the 49-m band and the higher-frequency bands up to around 16 MHz. A tap at the second turn provides the proper impedance for connection to the receiver.

The aerial can be connected using a coupling coil with two to four turns. If you make the coupling coil such that it can be moved back and forth, the degree of coupling can be adjusted. You can then experimentally determine the best adjustment. Tighter coupling yields a higher signal voltage, but it decreases the Q factor of the resonant circuit and thus reduces the attenuation at the mirror frequency. If it is necessary to use a short aerial (such as a rod aerial), the coupling must be designed to be relatively tight . In this case, the aerial can be connected directly to the hot end of the resonant circuit.

The resonant circuit shown in the figure has a high Q factor (typically 50). This yields a bandwidth of 120 kHz at 6 MHz. As a result, the variable capacitor must be tuned relatively precisely. For DRM reception, the optimum tuning can be recognised by a maximum signal level in the spectrum display generated by DRM software, but the delay in processing the date for this display makes adjustment difficult. It is easier to use direct acoustic monitoring by connecting the receiver output directly to the Line In socket of the sound card. It's then easy to tune for maximum noise volume for the DRM transmitter using the speakers of the PC.

If the preselector is built into an enclosure, the most important frequencies should be marked on a scale.

Figure 4 shows the same type of resonant circuit with the rotary variable capacitor replaced by a high-value variable-capacitance diode, such as the type BB112 (available from Geist Electronic). Here it is important to use a stable, well-filtered voltage for the tuning potentiometer, since otherwise reception can be degraded by phase modulation of the aerial signal.

The tuning range of a simple resonant circuit does not exceed 1:3 with a standard variable capacitor. One solution is to use several coils that can be selected using a rotary switch.

Another approach is used by radio amateurs, who are faced with the same problem in the standard amateur radio bands (80 m to 10 m, which corresponds to 3.5–29.7 MHz). This requires a preselector with a tuning range of 1:10. The solution is to use coupled circuits with two fundamentally different resonant frequencies. **Figure 5** shows a proven circuit using a dual variable capacitor and a second coil with 10 turns. Although there are two 'bad' pass frequencies for every setting, they are well separated from the mirror frequency of the receiver.

Good air-dielectric variable capacitors are no longer easy to come by. It is often possible to scavenge them from old

Table 1 DRM transmitters and frequencies.

Time	Frequency	Station	Language
06.00–24.00 Tu–Su	6095	RTL	various
08.00–14.00 daily	15440	DW	English
10.00–12.00 daily	6140	DW	English
10.00–12.00 daily	9850	RNW**	English
10.00–15.00 daily	7320	BBCWS	English
11.00–13.00 Sa & Su	9410	BBCWS	English
12.00–12.57 daily	9850	RNW**	Dutch
12.00–13.00 daily	6140	DW	German
13.00–15.00 Mo–Fr	9410	BBCWS	English
14.00–15.59 daily	6130	DW	German
15.00–16.00 daily	9490	VoR*	English
16.00–17.00 daily	9490	VoR*	German
16.00–17.29 daily	3995	DW	German
16.00–18.00 daily	6140	DW	English
16.00–19.15 daily	1296	BBCWS	English
17.00–18.00 daily	9490	VoR*	French
18.00–19.00 daily	6140	DW	German
19.00–20.00 daily	11925	BBC	Russian
21.00–22.55 daily	11730	RNW**	Dutch
21.15–24.00 daily	1296	BBCWS	English

* Voice of Russia

** Radio Netherlands World Service (Wereldomroep)

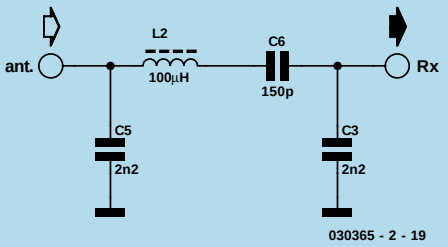


Figure 6. A 1296-kHz aerial filter.

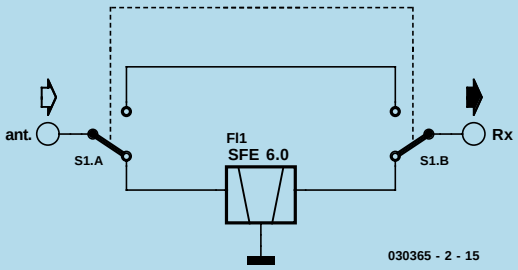


Figure 7. Using a ceramic IF filter for preselection.

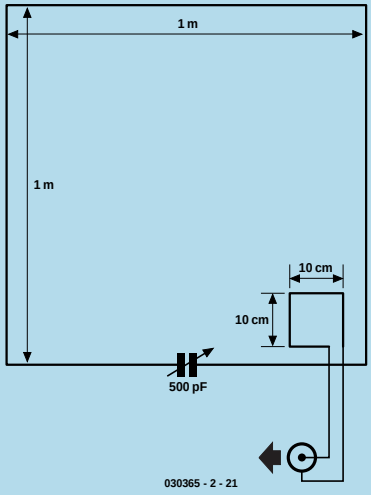


Figure 8. A magnetic-loop aerial.

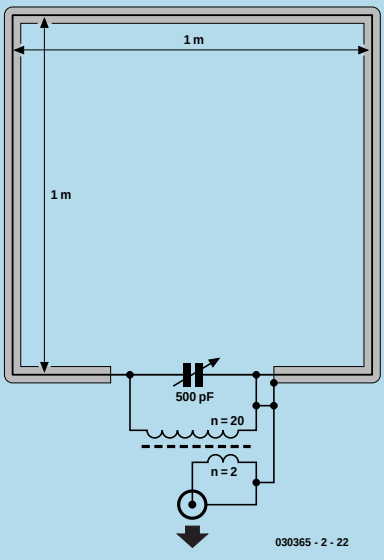


Figure 9. A tuned and screened loop aerial made from coaxial cable.

radios. One source of suitable multi-gang variable capacitors from current production is Geist Electronic (www.geist-electronic.de).

Fixed-frequency

An alternative to a circuit with adjustable tuning is to use a fixed-frequency filter. In the medium-wave band, there is often only one usable reception frequency. Although the relative bandwidth of the input circuit is relatively large, good selectivity is possible thanks to the low frequency. A circuit such as that shown in **Figure 6** can thus manage with a fixed inductor. This fixed-frequency filter for 1296 kHz improves reception of the BBC transmitter in the evening hours. It primarily reduces interference from multiples of the oscillator frequency in the short-wave bands.

For the 49-m band, a quite simple solution is possible using the type SFE 6 ceramic IF filter, which is used for the sound IF in television sets. **Figure 7** shows how to connect the filter to the receiver input so it can be switched in or out as necessary. The specified 3-dB bandwidth of around 100 kHz, which is actually too narrow, is broadened by the low impedances of the aerial and receiver to such an extent that the measured 6-dB corner frequencies are located at 5850 kHz and 6150 kHz. In practice, this filter is primarily useful in the 49-m band when interference occurs from strong transmitters in the 40-m band. At 7 MHz, the filter provides an attenuation of around 40 dB.

Magnetic-field aerials

A ‘longish’ wire aerial receives electrical energy from both the electrical and magnetic components of the radio waves. By contrast, short aerials such as rod aerials predominantly receive energy from the electrical component. This results in a higher level of interference, especially inside a building. The coupling between the receiver aerial and electrical equipment or the mains wiring is primarily capacitive. Better results can thus be obtained by receiving the magnetic-field component. In principle, all that is necessary for this is a wire loop or a coil. Commonly used solutions are loop aerials with a few turns of wire or single-winding loops, sometimes called ‘magnetic loops’. Tuned loops with high Q factors are particularly effective. For instance, you can use a length of copper pipe formed into a circle with a diameter of one metre. However, a broad piece of aluminium foil wound around a cardboard box with suitable dimensions also provides good results.

When connected to a variable capacitor with a maximum capacitance of 500 pF, the loop forms a high-Q resonant circuit and thus delivers a significantly higher aerial voltage than would be expected from an aerial of this size. The receiver should be loosely coupled using a small coupling coil, in order to avoid excessively damping the resonant circuit (see **Figure 8**). The optimum size and position of the coupling coil is best determined experimentally. Due to the high Q factor of the aerial, an additional preselector is in any case unnecessary.

A magnetic loop aerial can also be constructed using normal wire, although this yields a lower Q factor and thus a lower aerial voltage, as well as a larger bandwidth. If it is necessary to make the aerial smaller, two or more turns of insulated wire can be used.

An especially effective solution is an electrically screened loop

aerial, which in its simplest form can be made from a length of coaxial cable. Such an aerial can be inconspicuously fitted in a bookshelf, and it provides a relatively good signal to noise ratio. The resonant frequency depends on the size of the loop and the capacitor setting. If the total length of the coaxial cable is four metres, a resonant frequency range extending to below 6 MHz can be obtained with a 500-pF variable capacitor (see **Figure 9**). The primary inductance of the wideband transformer should be greater than that of the wire loop. Good results can be obtained with 20 turns on a ferrite core or toroidal core. In the interest of obtaining a high Q factor, the tuned circuit should not be too heavily damped. Consequently, the secondary has only two to four turns. The best value for the coupling coil should be determined experimentally.

In the medium-wave band, ferrite rod aerials have long since proven their worth. Like magnetic loops, they are relatively insensitive to electrical interference. **Figure 10** shows a ferrite rod aerial followed by an impedance converter. With a 10-mm diameter ferrite rod, the coil requires 70 turns of stranded RF wire ('litz wire') or 0.3-mm enamelled copper wire. Approximately 100 turns are necessary on a thinner rod (8 mm diameter).

Relatively large resonant voltages arise across the tuned circuit, even with distant transmitters. For example, at the author's location in Essen, Germany, at peak reception times an open-circuit voltage of 50 mV can be measured across a ferrite rod aerial with a length of 20 cm for the BBC signal at 1296 kHz. At the low-impedance receiver input, this still amounts to 5 mV, which is more than enough steam for our new digital steam radio.

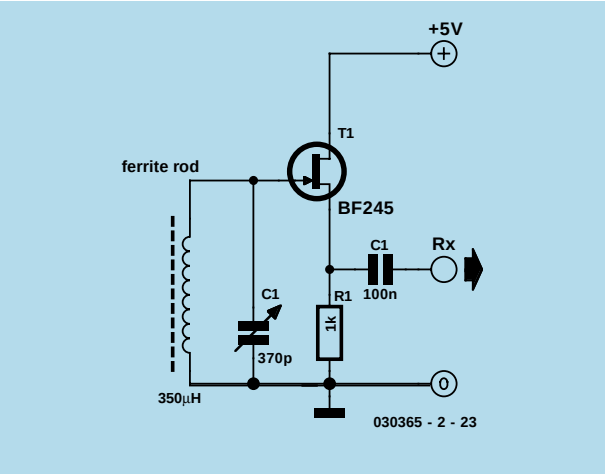


Figure 10. A ferrite rod aerial with an impedance converter.

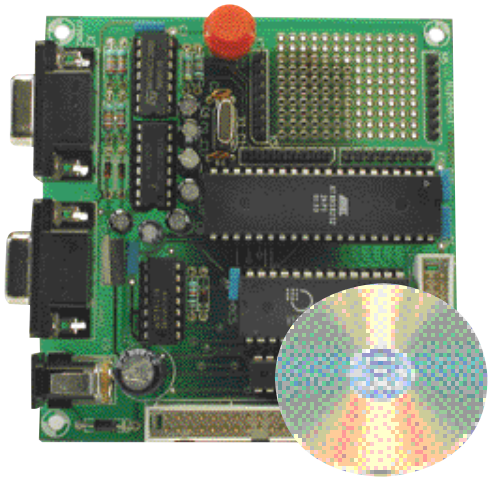
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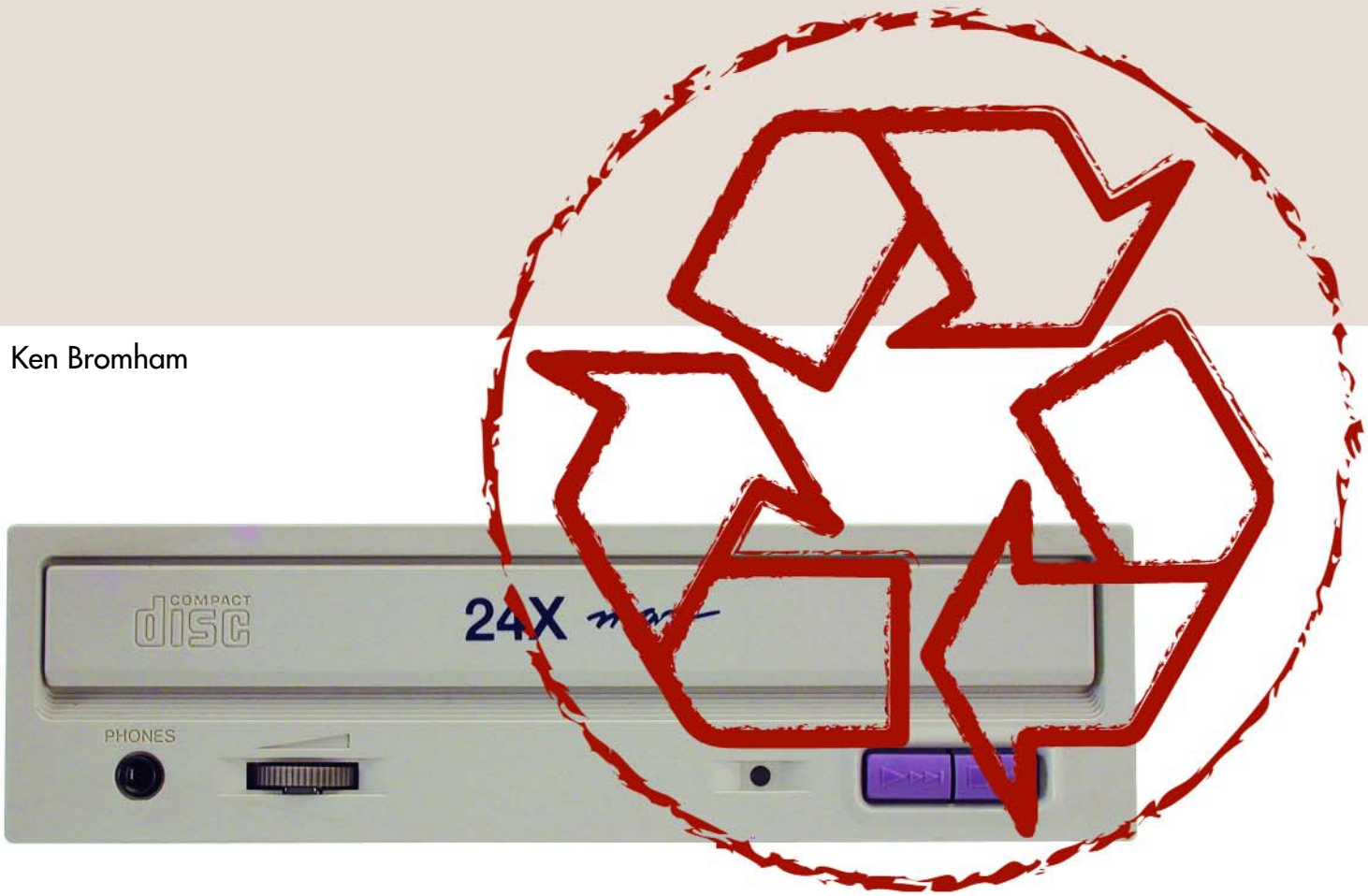
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Ken Bromham



Pandora's Sound & Music Box

Recycle an old CD-ROM drive for (triggered) playback

This project was originally conceived for model making applications as a cheap way to provide high quality audio playback in response to a trigger signal (typically from a pushbutton or PIR detector), but we're pretty sure Elektor readers can find many other uses.

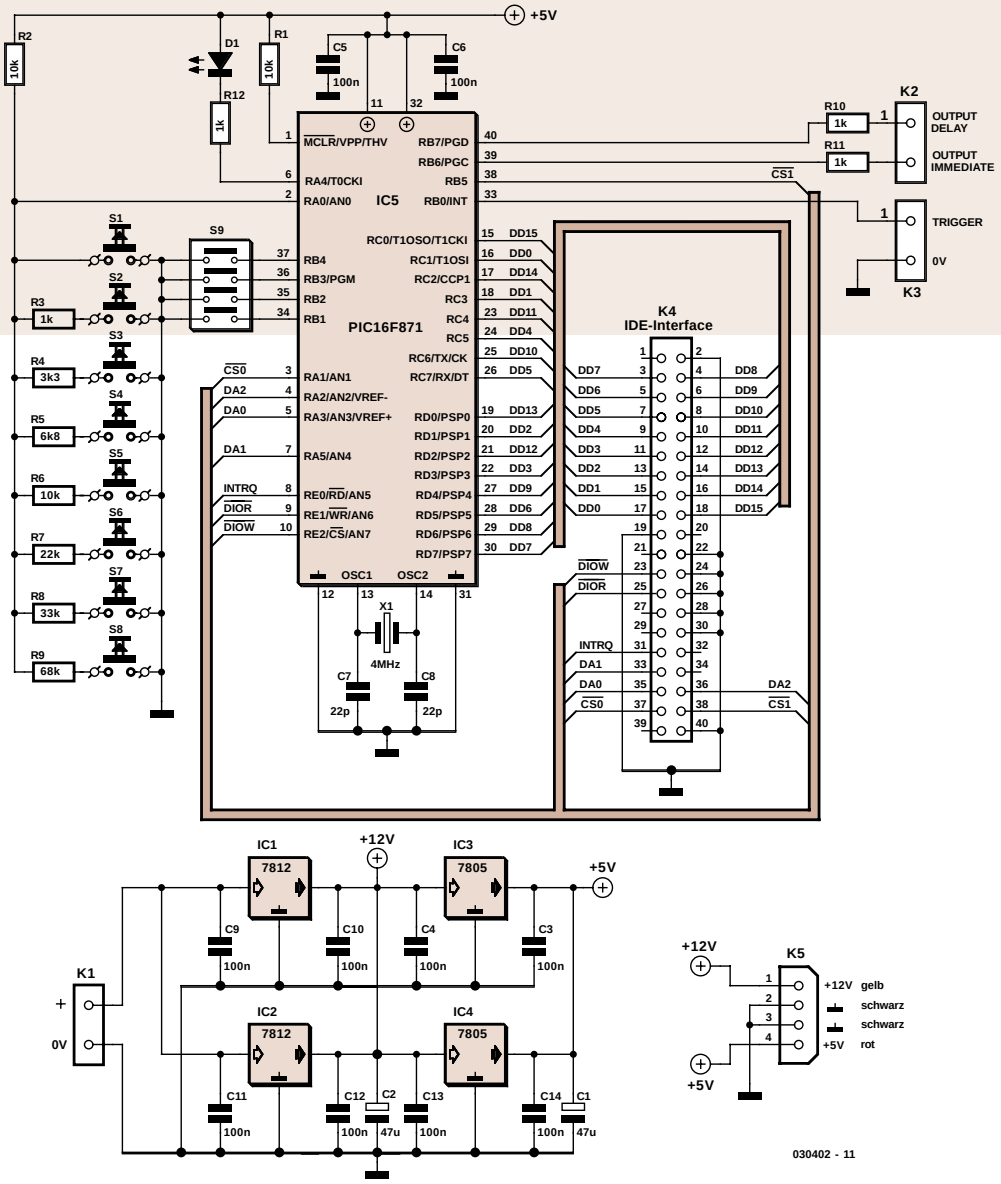


Figure 1. The modern version of Pandora's Box is a black one governing triggered response from an old CD-ROM drive playing back music or sound samples. You determine what 'horrors' (if any) lurk inside the box.

The standard low cost solution to creating auto-playback sounds is to use a record/playback chip (say, the ISD25XX series) but these devices are limited to about 3 kHz audio bandwidth, not to mention a lot of hiss and noise, so the sound quality is not the best unless you're specifically after for voice messages of the Stephen Hawking type.

The Elektor approach

Over the past few years we have received many requests from readers asking for a circuit that would allow them to use an old CD-ROM player for the sole purpose of playing back music CDs. In many cases, that is possible just by connecting a power supply and headphones to your drive, inserting a CD and pressing the Play button. Simple as this may sound, there are pitfalls, particularly with later CD-ROM drives that do not have a music playback button. Also, we really could not stand the thought of being unable to select and skip tracks, etc. In true Elektor fashion we wanted to be in control of things and make the old CD-ROM drive do something really useful. The alternative approach developed by Ken Bromham and described in this article makes use of a 40-pin PIC 16F87X microcontroller which, helped by a small number of external components, provides a versatile controller module that will interface to any old ATAPI CD-ROM drive. Mind you, ATAPI is not a brand, but a connectivity standard for 99% of all CD-ROM drives in PCs for home and office use. Don't worry about it, just start rummaging around in the attic or cellar, dig a CD drive out the IT skip at work, or commandeer the oldest (usually read-only) CD drive from the kiddies' PC leaving a note saying '*drive removed for scientific purposes*'. Never tell them you got the idea from Elektor, instead, tell them to keep using the CD/RW drive which is much faster, better, etc. or better still buy them a mini MP3 player and headphones.

COMPONENTS LIST

Resistors:
R1,R2,R6= 10kΩ
R3,R10,R11,R12 = 1kΩ
R4 = 3kΩ3
R5 = 6kΩ8
R7 = 22kΩ
R8 = 33kΩ
R9 = 68kΩ

Capacitors:
C1,C2 = 47µF 16V radial

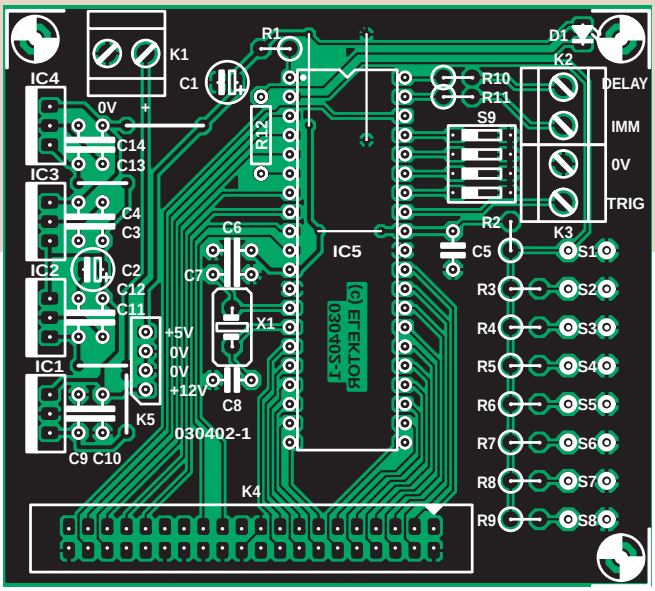


Figure 2. The printed circuit board is single-sided and contains seven wire links.

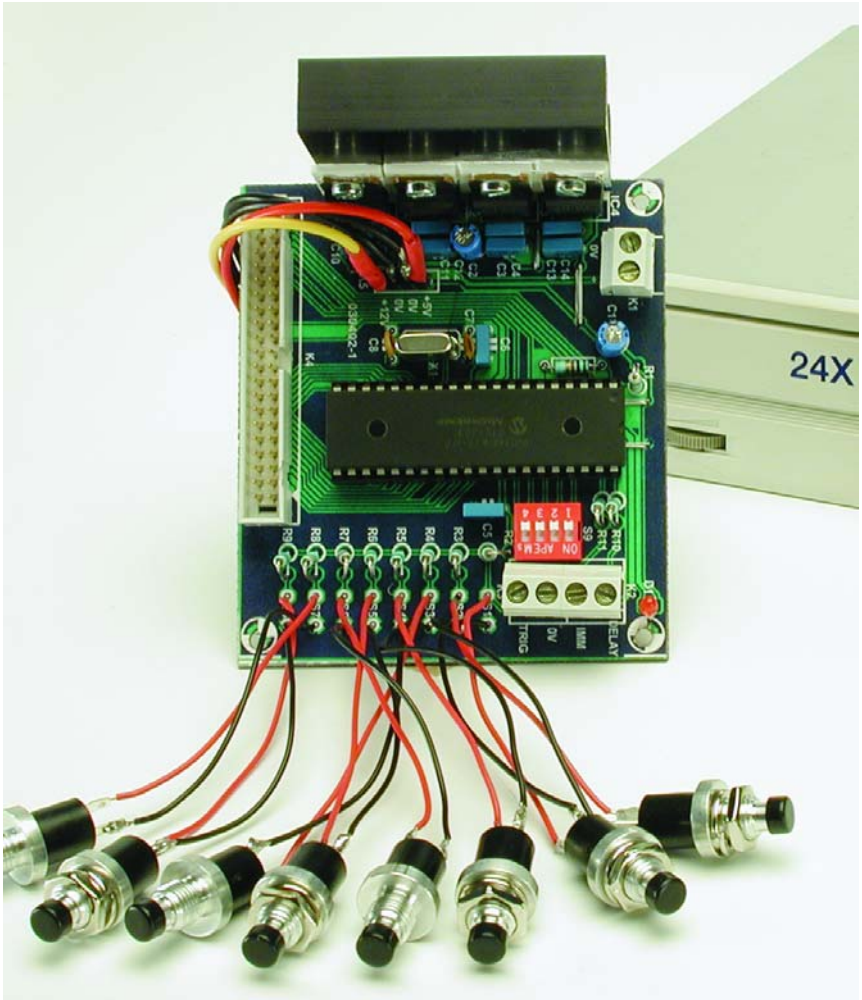


Figure 3. When in doubt about any constructional aspect, just use this photograph for guidance.

What can it do?

The controller module has two main modes of operation, single trigger use or multiple-trigger use, with further options selected by a 4-way DIL switch (psst… in fact both modes run simultaneously). There are also two digital switching outputs, asserted during playback, to allow other features (for example, lights or a motor) to be automatically switched on for the duration of the audio playback.

Single-trigger mode

The function here is very simple. Activating the single trigger input will cause playback of one track from the CD. There are four options (selected with a DIL switch), as follows :

DIL switch # 1
off = normally open contact for single trigger
on = normally closed contact for single trigger

DIL switch # 2
off = no response to trigger until end of track
on = respond anytime

DIL switch # 3
off = random track selection
on = sequential track selection

DIL switch # 4
off = keep disc continually spinning (defeat drive's inactivity timeout)
on = allow the CD-ROM drive to power down.

The latter options may require some elucidating. In the first case the playback will always start almost immediately as the disc is always spinning, but this may have some impact on the drive MTBF value (mean time between failure — you'll find it hard if not impossible to find data on this). In the second case, if the disc has stopped spinning, there will be a short 'spin up' delay before playback starts. No problem for applications that are likely to

C3..C6,C9-C14 = 100nF
C7,C8 = 22pF

Semiconductors:
D1=LED, red, low current
IC1,IC2 = 7812
IC3,IC4 = 7805
IC5 = PIC16F871/P, programmed, order code **030402-41** (see Readers Services page)

Miscellaneous:
S1-S8 = pushbutton, 1 make contact
S9 = 4-way DIP switch
X1 = 4.000MHz quartz crystal
K1,K2,K3 = 2-way PCB terminal block, lead pitch 5mm
K4 = 40-way boxheader
K5 = power supply plug for CD-ROM drive
PCB, order code **030402-1** (see Readers Services page)
Heatsink, e.g., Fisher SK59 (6 K/W)
Disk, PIC source and hex code files, order code **030402-11** or Free Download
CD-ROM drive

Free Downloads

PIC source and hex code files. File number: 030402-11.zip

PCB layout in PDF format. File number: 030402-1.zip

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

spend a lot of time 'doing nothing'.
There may be up to 24 tracks on the CD (the PIC chip is storing the table of contents data in its limited RAM). The multiple trigger inputs should be left open.

Multiple-trigger mode
In this mode you can have a maximum of eight separate triggers (for instance, push buttons). Pressing button 1 will always play track 1, button 2 will always play track 2, etc., up to and including track 8. It can be used with normally-open contacts only. If, for example, there are only three tracks on the CD then pressing buttons 4 – 8 will have no effect.

DIL switch 1 must be configured as normally open and the single trigger input left open.
DIL switch 2 is not relevant.
DIL switch 3 has the same function as above.
DIL switch 4 has the same function as above.

Digital switching outputs
Pandora's Sound & Music Box has two digital outputs for control of external devices like sounders, lamps, amplifiers, signal routers, door locks, you name it, anything can be controlled as long as it has a simple 0/5 V (TTL) digital control input, or can be switched on and off with a few mA of drive current.
The 'Output Immediate' output goes high immediately after triggering and remains high until end of playback.
The other output called 'Output Delay' goes high only after playback has started (that is, after any spin-up delay) and remains high until end of playback.

Circuit and construction

The circuit diagram shown in **Figure 1** has few surprises, basically showing a microcontroller sitting between a bunch of switches and some connectors. The heart of the circuit is a 40-pin PIC16F871 microcontroller which fortunately has enough input/output pins to connect to all of the necessary ATA interface lines with enough left over to handle the trigger inputs, option selections and the switching outputs. As can be seen from the schematic, only a handful of extra components are required. Note that the single trigger input RB0 (K3) and the option select inputs RB1-RB4 (S9) make use of internal, that is, invisible, pull-up resistors. All other port lines of the PIC16F871 are connected to the drive's IDE (ATAPI) interface via connector K4. Together with R2, resistors R3–R9 provide a simple potential divider network connected to input A0 on the PIC. The upshot is that a different voltage is applied to A0 depending on which button is pressed (multiple-trigger mode). This voltage is read by an internal A/D converter. It is assumed here that it is not necessary to distinguish multiple simultaneous button presses. If multiple trigger mode is not required then resistors R3–R9 and switches S2-S8 can be omitted, but R2 must be retained to keep input A0 pulled high.
The PIC ticks at 4 MHz as determined by quartz crystal X1 and its usual pair of small satellite capacitors, here identified as C7 and C8. The user-defined settings are read from DIL switch S9. One LED, D1, has been included to acts as a 'PIC awake' indicator (very useful!). Capacitors C5 and C6, finally, ensure the 5-V supply voltage to the PIC remains as clean as possible.
The circuit has been designed to operate from a single 15-18V DC supply, which should be 'heavy' enough to also supply the drive's 12-V line. Two paralleled 7805 fixed voltage regulators, IC3 and IC4, provide +5 V for the PIC and the CD-ROM drive's 5 V line. The 12-volt supply is realised in a similar way by two 7812s in parallel. Alternatively, an old PC power supply can be used to power the CD-ROM drive directly. In this case it is recommended to retain the 7805s and use 12 V from the pc power supply for the controller board. A heatsink will still be necessary.
Resistors R10 and R11 provide current limiting for the digital outputs and have been given the nominal value of 1 kΩ. The PIC chip can source/sink an absolute maximum of 25 mA for each pin, so the value of these resistors can be changed so long as this maximum is not exceeded. In any case, it is sufficient to drive a transistor/relay combination for example.
The PCB shown in **Figure 2** was designed for ease of use by you, the constructor. It is available ready-made through our Readers Services under number **030402-1**. Alternatively, you may decide to make your own board using the artwork file that can be downloaded free of charge from our website.
As there are only regular components to fit on the board we doubt the construction will present any problems. The simplest and cheapest component on the board, however, is often the one that's forgotten, causing major headaches and dozens of unnecessary emails of the 'Help it don't work' type! We're talking about the infamous **wire link**. There are seven of them on the board and they are best fitted before any other component so they're not forgotten. Bolt the voltage regulators onto a common heatsink (see **Figure 3** and the parts list), insulating washers are not required as all metal tabs are connected to ground. The PIC being the most expensive part, it deserves to be fitted into a 40-way DIL socket with good quality contacts.
Although the circuit diagram suggests that there are rather a lot of wires and other things to connect to the board, in reality the situation is not that bad as you can see from **Figure 4**. The cables

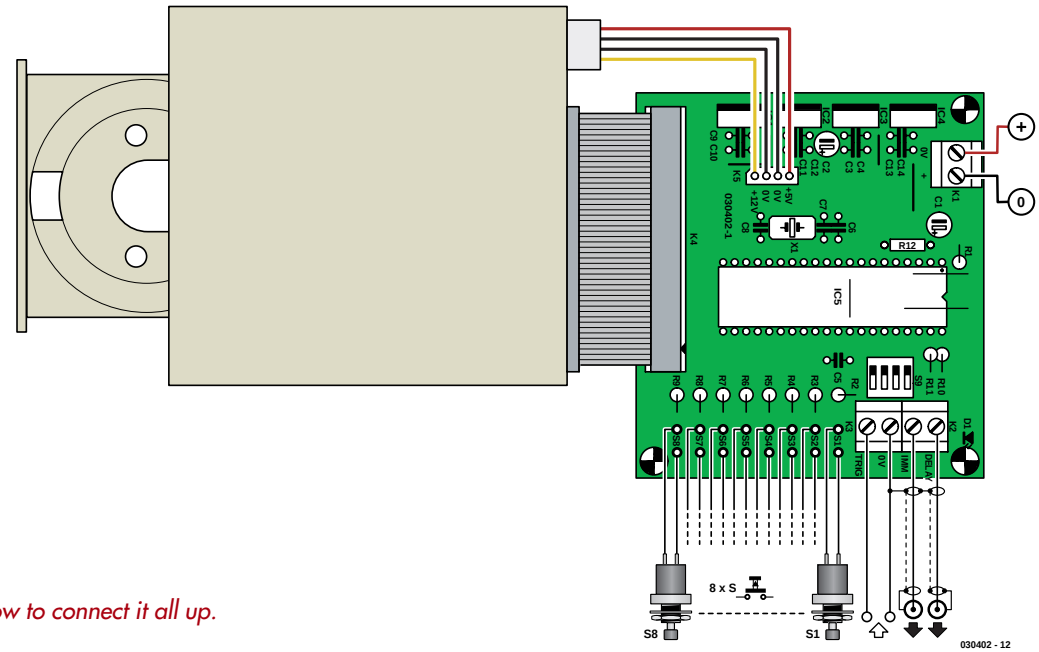


Figure 4. How to connect it all up.

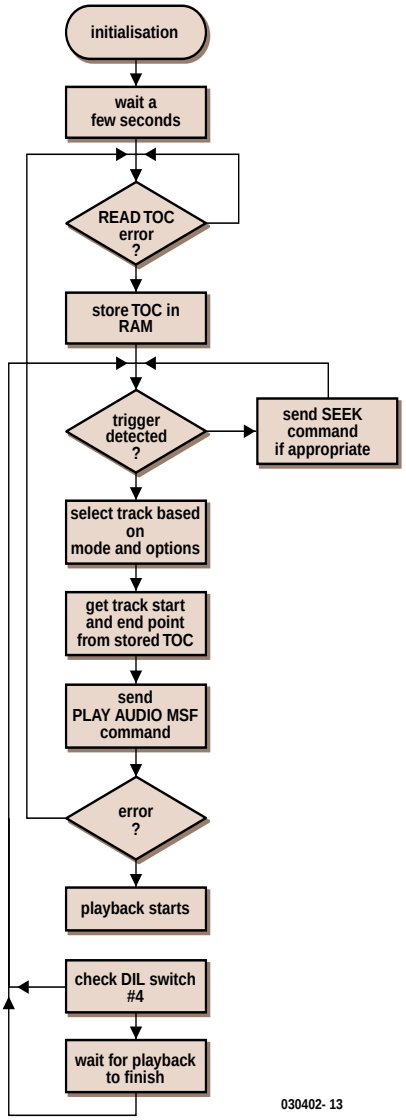


Figure 5. Use this flowchart of the main program if you don't fancy reading assembly code but still want to understand how the software works.

between the board and the CD-ROM drive, for example, are ready-made ones pulled from the junkbox or an old computer.

Program

The program that runs inside the PIC micro has been written in assembly language. The source code and hex file are available on floppy disk (anyone out there still using these?) or as free downloads from the Publishers' website. If you wish to program your own PIC, please feel free to do so using the files supplied. Alternatively a pre-programmed PIC is available from Readers Services, the order code being **030402-41**.

The source code supplied by Ken Bromham is well worth studying, even if you do not build the project. Ken succeeded in including plenty of comments so if you are familiar with this assembly language it should be possible to follow the program, despite some classic spaghetti code. If not, you may still want to grasp the 'broader lines' offered by the flowchart of the main program shown in **Figure 5**.

The actual ATAPI commands used by the PIC firmware are:

PLAY AUDIO MSF (play from specified start to end location. MSF = Minutes, Seconds, Frames, 75 Frames = 1 Second).

READ TOC (get the table of contents).

READ SUBCHANNEL (used to get the current audio status).

SEEK (position the head at start of track 1, but will also cause the disc to spin up, so used to defeat the CD-ROM drive inactivity timeout).

The READ SUBCHANNEL command is used at various points in the above sequence, whenever the program needs to know if playback is currently in progress or if playback has finished.

Testing

As usual, check for the presence of 5-V before inserting the PIC chip. The module can then be tested without connecting to a CD-ROM drive. Simply power up and check that the LED on pin A4 flashes a few times. Nothing else will happen, but this confirms that the PIC is up and running. Switch off, connect to CD-ROM drive, power up. The LED should flash a few times and then continue to flash until a disc is inserted, the tray is closed and the TOC (table of contents) successfully read. When the LED stops flashing, the module is ready to respond to a trigger and the different options can be experimented with.

Note that the CD-ROM drive must be configured as a MASTER device, and pin 1 on the PCB socket (K4) must go to pin 1 on the CD-ROM ATA interface socket (usually indicated by a red wire in the ribbon cable). When used with a single 12-V DC supply make sure this supply can provide a minimum of about 1.2 A. The audio output can be taken from the analogue out on the back of the CD-ROM drive or from the headphone jack a the front. To continue the low-cost theme, we recommend a cheap pair of active 'multimedia' speakers (whatever that means), unless, of course, you really want to build your own amplifier!

ATAPI protocol in brief

ATAPI (AT Attachment Packet Interface) devices use the same physical interface as ATA (AT Attachment) devices such as a hard disk drive, so it is necessary to understand how this works first. In summary, the ATA device has a limited number of 8-bit registers (for example,. COMMAND, STATUS) and a single 16 bit DATA register. The interface is of the parallel type with 16 bi-directional data lines, where only the lower 8 data lines are used for reading/writing the 8 bit registers. ATAPI devices use the same register set, although some registers have been renamed and serve a different function. Unfortunately this does not provide enough flexibility for the increased range of commands required, so the concept of a command packet was introduced along with a new ATA command, the 'ATAPI packet command'. To send a command to an ATAPI device the general procedure is to first write the (generic) ATAPI packet command to the device's COMMAND register, and then send the command packet by writing multiple times to the device's DATA register. For CD-ROM drives the command packet is 12 bytes in length and so 6 consecutive writes to the DATA register are required, sending 2 bytes each time. The command packet contains an opcode for the specific ATAPI command along with any additional parameters that are required.

Here is an example of the packet for the PLAY AUDIO MSF command:

Byte 0:	**Operation Code (0x47)**
Byte 1:	**Reserved**
Byte 2:	**Reserved**
Byte 3:	**Start Location Minutes**
Byte 4:	**Start Location Seconds**
Byte 5:	**Start Location Frames**
Byte 6:	**End Location Minutes**
Byte 7:	**End Location Seconds**
Byte 8:	**End Location Frames**
Byte 9:	**Reserved**
Byte 10:	**Reserved**
Byte 11:	**Reserved**

You can see that the command packet is padded with spare (reserved) bytes if necessary to give the 12-byte length.

Detailed documentation can be found on the web, www.t13.org is a good place to start looking, just be prepared for some serious bed-time reading.

ATA Interface pinning

Pin no.	Label	Description
1	$\overline{\text{HRESET}}$	Reset
2	GND	Ground
3	HD7	Data bus bit 7
4	HD8	Data bus bit 8
5	HD6	Data bus bit 6
6	HD9	Data bus bit 9
7	HD5	Data bus bit 5
8	HD10	Data bus bit 10
9	HD4	Data bus bit 4
10	HD11	Data bus bit 11
11	HD3	Data bus bit 3
12	HD12	Data bus bit 12
13	HD2	Data bus bit 2
14	HD13	Data bus bit 13
15	HD1	Data bus bit 1
16	HD14	Data bus bit 14
17	HD0	Data bus bit 0
18	HD15	Data bus bit 15
19	GND	GND
20	N/C	Key pin
21	DMARQ	DMA request
22	GND	Ground
23	$\overline{\text{HWR}}$	I/O write
24	GND	GND
25	$\overline{\text{HRD}}$	I/O read
26	GND	Ground
27	IORDY	I/O channel ready
28	SPSYNC:CSEL	Spindle sync or cable select
29	$\overline{\text{DMACK}}$	DMA acknowledge
30	GND	Ground
31	INTRQ	Interrupt request
32	$\overline{\text{IOCS16}}$	16 BIT I/O
33	HA1	Address bus bit 1
34	$\overline{\text{PDIAG}}$	Passed diagnostics
35	HA0	Address bus bit 0
36	HA2	Address bus bit 2
37	$\overline{\text{CS1FX}}$	Chip select 0
38	$\overline{\text{CS3FX}}$	Chip select 1
39	$\overline{\text{DASP}}$	Drive active/drive 1 present
40	GND	Ground

With compliments, Pandora

(030402-1)

There's only one limit to the applications of Pandora's Sound & Music Box: your imagination. Just couple the two notions 'some kind of trigger' to 'an audible, pre-recorded response' and away you go. Here are some possible applications to get you going, in fairly random order:

– a voice message system employing the public address sound equipment in a large building. Very useful for guiding the public to fire exits in case of an emergency, when written notices are hardly ever seen, let alone read.

– an electronic dog barking in response to your doorbell; a track with More & Fiercer Dogs for really persistent callers that push the bell a second time (Pandora must have heard of Cerberus), gunfire for a third time, then police sirens, and so on.

– a door- or doormat-triggered muzak or 'welcome' generator.

– a voice guide in museums, triggered by visitors approaching an exhibit.

– a low-cost jingle-and-tune box for quizmasters and deejays.

– a language training aid.

– a spoken Callsign / 'CQ Contest' generator for radio amateurs.

For all of the above applications, you will need to burn your own music CD. Programs to compile music or sound samples onto your own CDs abound in PC land, CoolEdit being one of the best known. Note however that you can't use MP3 files just like that — a suitable decoder will have to be added.

out inside out inside out inside

of the past. None the less, there are (or were) disadvantages. The first optical mice were slow to react, inaccurate and power hungry. These problems have now all been solved, so only the advantages remain. There is still one small weakness: an optical mouse does not work well on some surfaces, because it cannot find enough identifying marks. However, this problem is easily solved with another mouse mat.

As a result of falling sensor prices and the simple construction of the sensor (practically everything, except the LED, is accommodated in one IC) the mechanical mouse is now slowly disappearing from the retail shelves.

Camera and DSP

At first glance, it would seem easy to detect the movement of the mouse using an optical system. But behind this simple idea hides some complex electronics that's comparable to a simple video camera combined with an intelligent digital movement detector.
Figure 1 shows a cross-section of the mouse near the optical sensor. All the electronics, including the 'camera' part and the lens, are contained in a single IC labelled *sensor* in the diagram (**Figure 2** shows a block diagram of the IC). An LED next to the IC provides for sufficient illumination of the surface over which the mouse is moved. The sensor catches some of this reflected light and translates it into an image. The camera section is relatively small, usually only 20_20 or 30_30 pixels. These generate a black-and-white image for the image-processor (a DSP), which repeatedly performs a pattern analysis and compares this with the previous pattern. Using patented technology, the DSP can deduce in an instant in which direction the pattern is moving and at which speed. **Figure 3** shows two images separated in time by about 0.7 ms. The processor recognises similar patterns in each of these images and uses these to calculate the displacement in the X- and Y-direction.

Modern sensors have a resolution of 400 or 800 CPI (counts per inch) and reach unbelievably high processing speeds of around 2500 frames per second. This makes movement speeds of up to 12 inches (approx. 30.5 cm) per second possible! Wireless optical mice use special techniques to reduce the power consumption as much as is possible. For example, the frame-rate is reduced when little movement has been detected.

Finally, a quick look at the output signals produced by the mouse sensor. The cheaper sensors are limited to a serial output, which transmits the X- and Y-displacement data to a processor in the mouse for further processing. Larger sensors often have additional quadrature outputs, which provide signals similar to a mechanical mouse. In principle you could use such a sensor to modify an existing mouse yourself…

(040026-1)

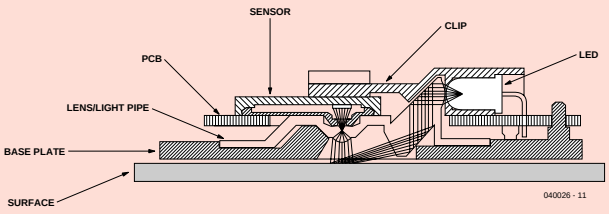


Figure 1. This drawing shows a cross-section of a mouse near the sensor and LED. (source: Agilent)

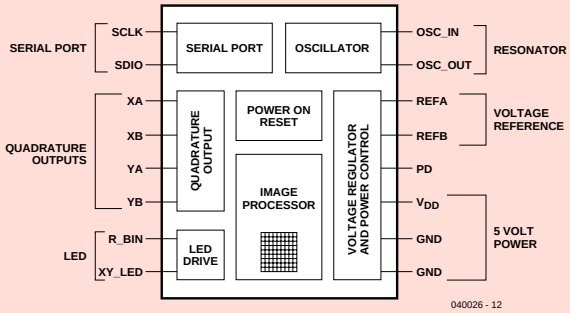


Figure 2. Block diagram of the design of an optical mouse sensor, in this case an ADNS-2051. (source: Agilent)

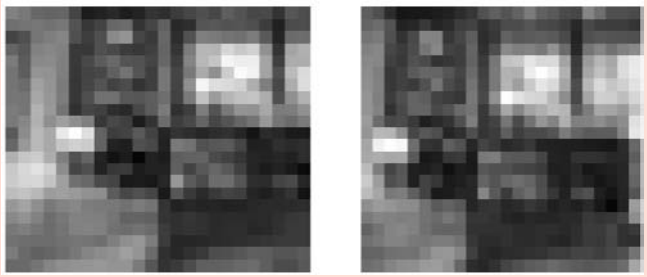
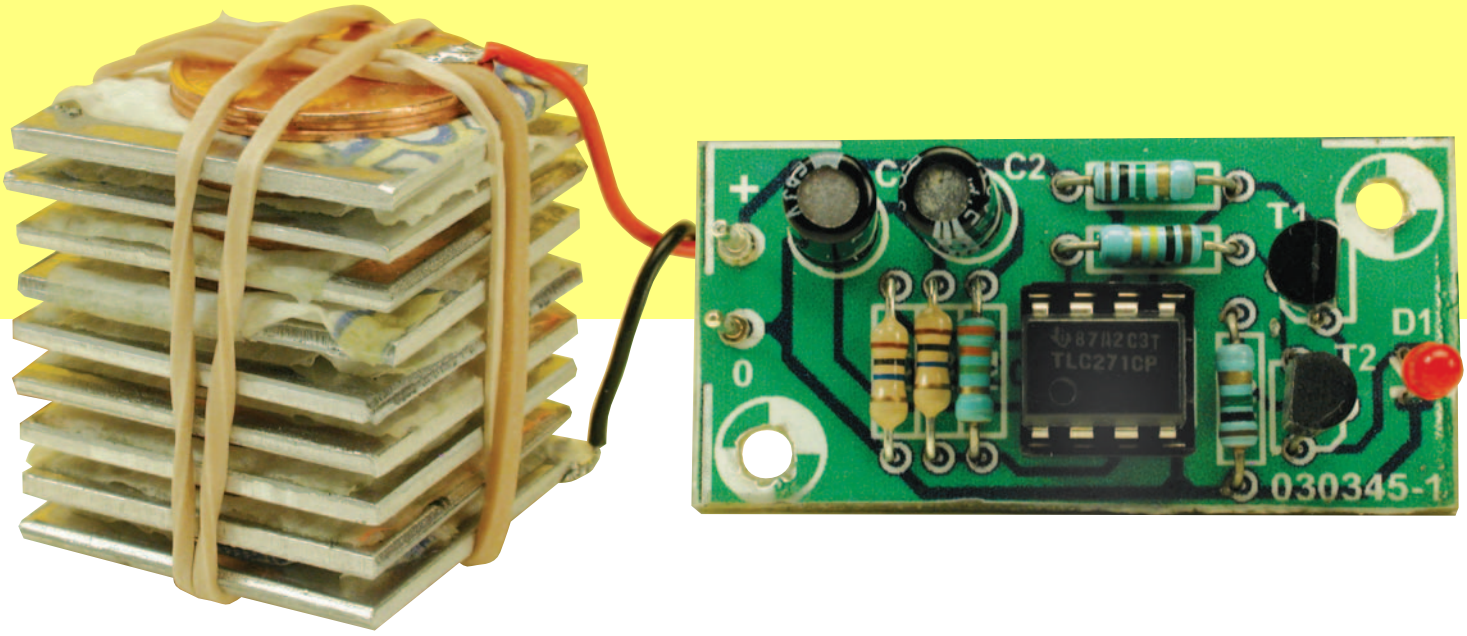


Figure 3. These are two images recorded by the sensor 0.67 ms apart. From this information the DSP calculates the displacement in the X- and Y- direction. (source: Agilent)



Prof. Dr. Martin Ohsmann

Low-Power LED Flash
powered by lemon juice

This circuit will need just a few micro-amps to make an LED flash. The energy comes from a home made 'vegetarian' battery.

We have to admit that the circuit shown in **Figure 1** appears a bit over the top for an on/off indicator, as the same functionality could have been achieved from a classic combination of an LED and a series resistor. However, the circuit is unbeaten we feel in that it shows how a visual on/off indication may be realised in battery-powered equipment, while using a minimum of energy.

It is of course impossible to make an LED light continuously with very little current. The solution is to make it flash briefly. The requirement, then, is for a low-energy oscillator capable of creating an extremely low duty factor (on/off ratio). A large time constant is realised by capacitor C1 in conjunction with charging resistor R1. Micropower opamp IC1 acts as an inverting comparator, monitoring the voltage on C1 and comparing it with a threshold level defined by resistors R2 and R3. Once capacitor C1 contains sufficient charge, it has to be discharged again. The energy stored in the capacitor is of course not wasted but put to good use — in this case powering the LED.

In order to create a relatively intense current peak through the LED, the capacitor is discharged through a 'thyristor tetrode' formed by T1 and T2. This composite device is triggered by the comparator via R5. Because the thyristor tetrode does not consume current as C1 is being charged and the Micropower opamp itself is an extremely energy-wise component, the total current consumption remains extremely low — just 15 µA at 12 V. The circuit will function from a supply voltage as low as 3.5 V, although the LED in that case will not light brightly anymore. On a positive note, the current consumption is then reduced to just 4 µA!

COMPONENTS LIST

Resistors:
R1,R6 = 1MΩ
R2 = 3MΩ3
R3,R4 = 9MΩ1
R5 = 100kΩ

Capacitors:
C1,C2 = 1mF 16V (see text)

Semiconductors:
D1 = LED, red, low current
IC1 = TS271C (see text)
T1 = BC557C
T2 = BC547C

Miscellaneous:
K1 = 2 solder pins
2 solder tags
9 copper or coppered discs (e.g. coins, see text)
9 aluminium discs (see text)
Absorbing kitchenpaper (see text)
1 lemon (or pure lemon juice, see text)
1 rubber ring
PCB, available from The PCB Shop

Construction

The construction on the printed circuit board shown in **Figure 2** is not critical in any respect. Capacitor C1 must be a foil-dielectric type (for example, Siemens MKT). C2, too, should exhibit a low leakage current because otherwise the current consumption will be far higher than desired. The LED should be a 'high-intensity' or a 'low-power' type.

The introductory photograph shows an initial attempt at construction using Vero board.

The circuit employs a micropower operational amplifier type TS271C from ST (formerly SGS-Thomson). This is **not** a direct replacement form the TLC271, although the latter device may also be used in this circuit if you remember to link pin 7 to 8 in order to preserve the low-bias setting. Resistor R4 which defines the bias current with the TS271 is then omitted.

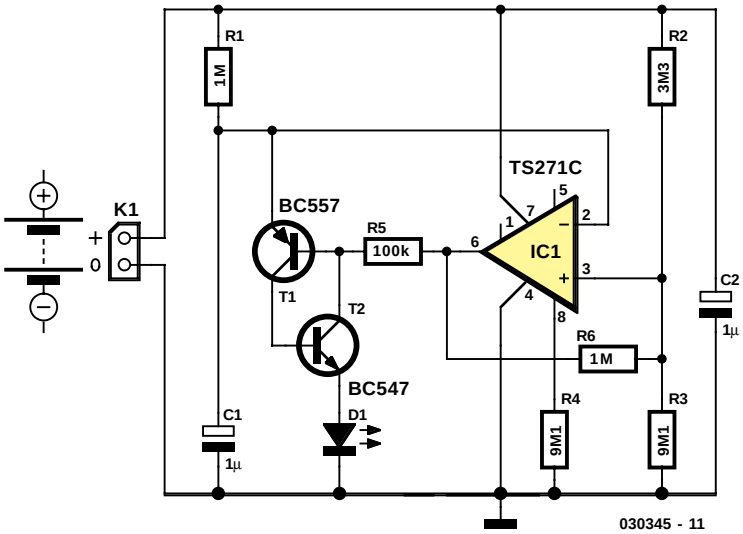


Figure 1. Circuit diagram of the energy-wise LED flasher employing the TS271C micropower opamp.

Add some lemon juice

The extremely low current consumption at reasonable supply voltages actually allows our little circuit to be powered from a home-made battery. The general idea is depicted in **Figure 3**.

The battery consists of a nine individual elements connected in series by stacking. Each element is made up from a (coppered) coin with a diameter of 10-15 mm, a separating sheet made from kitchen paper soaked in lemon juice, and a piece of aluminium foil. The components that constitute the 'veggie battery' are also visible in the introductory photograph.

Several experiments were carried out and the following procedure was found to give best results: fold the aluminium foil eight times and use the circumference of the coin to cut out eight round packets in one go. Next, cut nine square pieces of kitchen paper, the edges should be slightly longer than the diameter of the coin. This is necessary to prevent a short circuit between copper and aluminium. Then saturate the paper with lemon juice. Freshly pressed juice will work better than juice from a bottle!

The lower part of the battery assembly is formed by a slightly larger piece of aluminium foil — this will act as the negative terminal. On it lies a paper disc. Then stack on a coin, an aluminium disc, a paper disc and so on, right up to the top coin that acts as the positive terminal. If you are satisfied with your creation, the presence of a voltage may be proved with the aid of a high-impedance digital voltmeter. The battery terminal voltage should be between 3 V and 4 V. Connect the battery to the circuit and you'll be able to watch the LED flash for quite a while getting its energy from… lemon juice!

(030345-1)

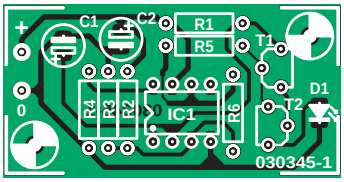


Figure 2. A PCB to try it all out.

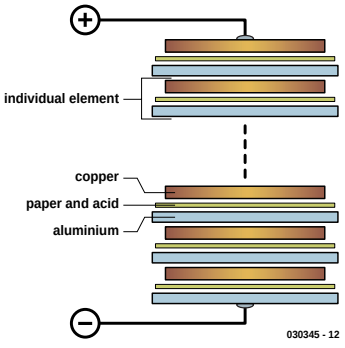
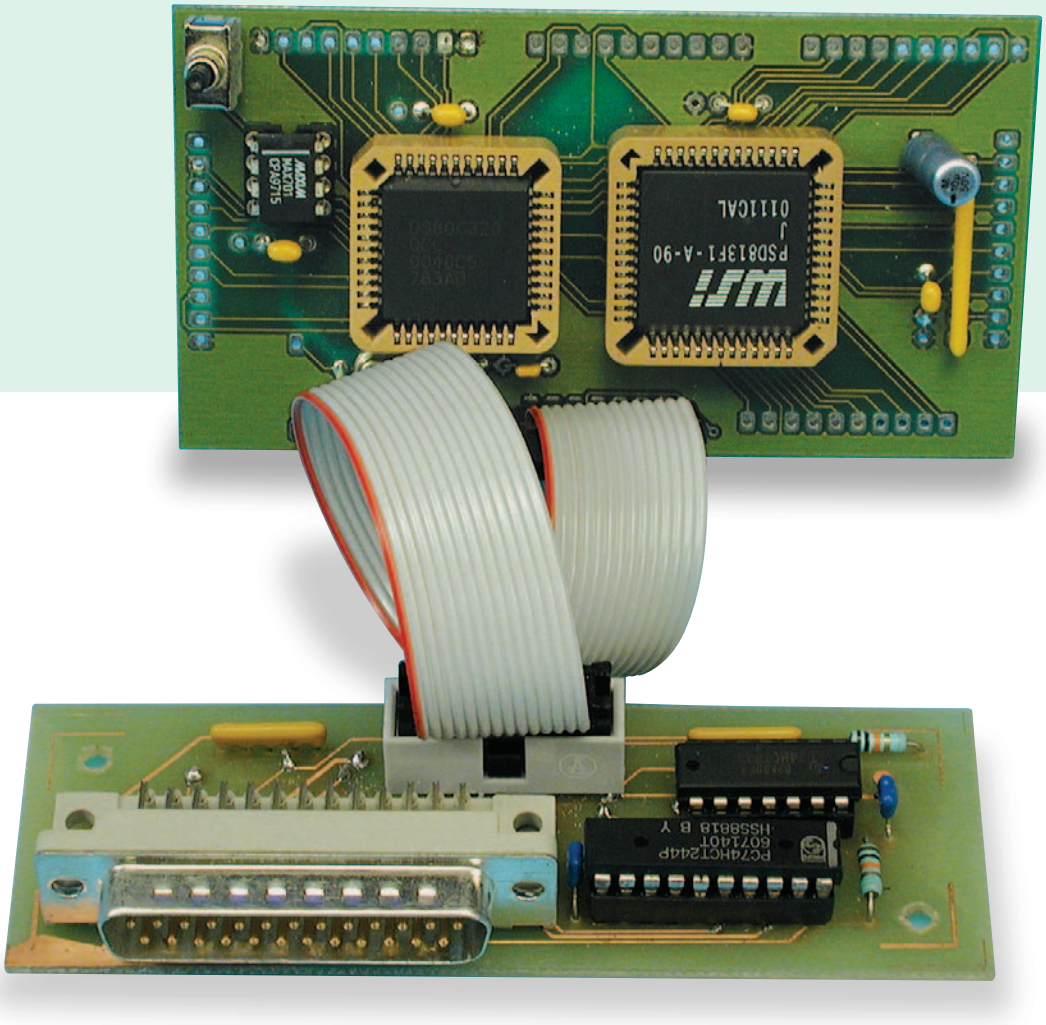


Figure 3. Schematic representation of the battery.

Søren Petersen



Drop-in Microcontroller Board

An 80C32 jack-of-all-trades

At the heart of this remarkably compact microcontroller board you'll find an ingenious bit of integration technology. The combination acts as a versatile plug-in module for lots of applications requiring some form of intelligent control. Thanks to the extensive software that comes with the board and the presence of a PC interface, our drop-in board is excellent for hardware development, too.

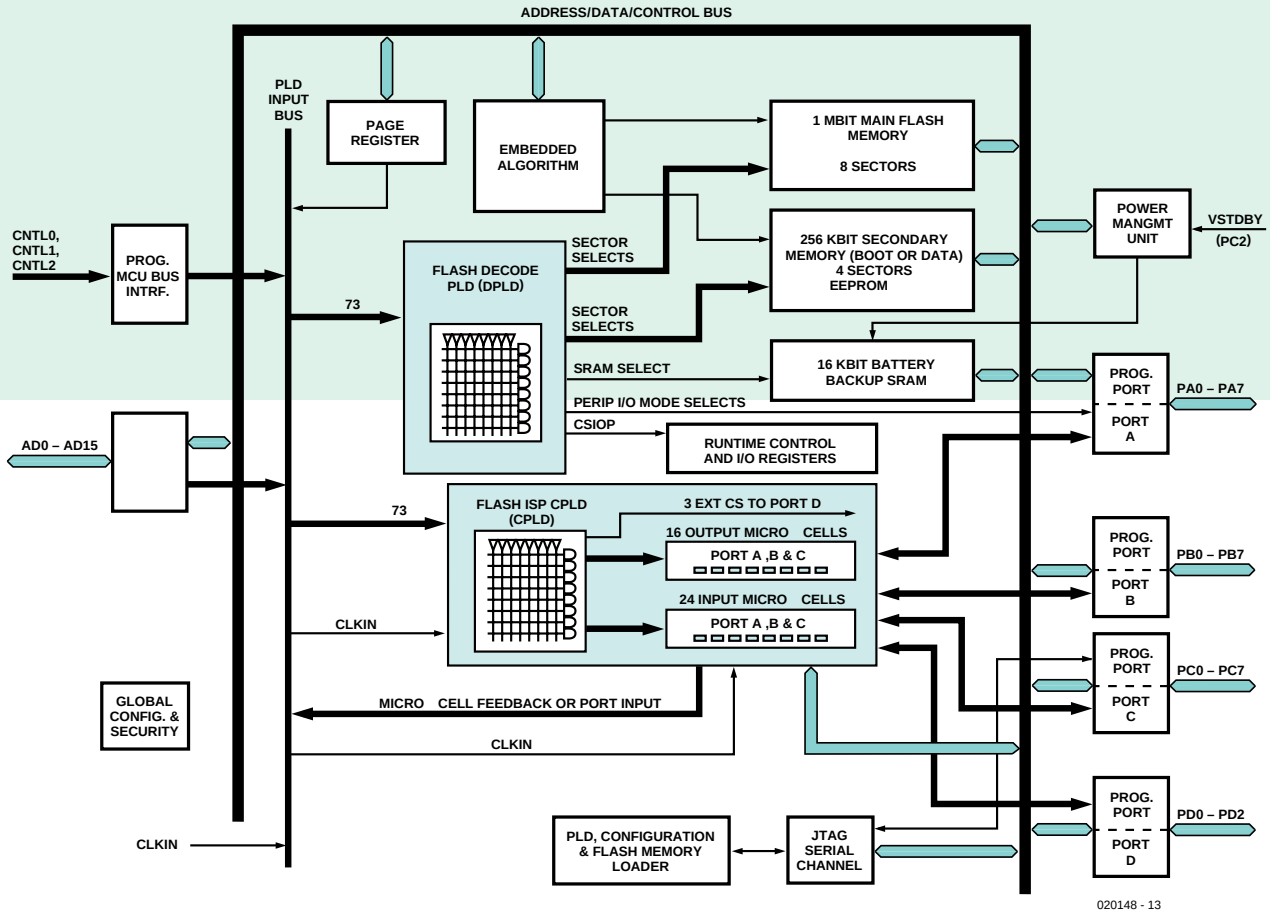


Figure 1. Internal structure of the smart 'all-in-one' chips from the PSD813xxx series.

As many of our readers will be able to confirm, the majority of microcontroller boards look very similar. If the microcontroller used has no program memory, an EPROM is added. Similarly, if there is a lack of RAM, the problem is solved by adding an external RAM device like the 62256. The lot is then complemented with a few logic gates and *hey presto* we have a basic microcontroller system. In some cases the designer will go one step further by employing a little more than just the microcontroller I/O pins. Some more digital electronics is then in order, but that really wraps it up as far as the circuit is concerned.

PSD813 series

Okay, if all these microcontroller circuits are so similar, why not design a single integrated circuit that contains all functionality? That way, circuits can become much smaller and the designer could concentrate on those points that are unique to the circuit. This, we figure, must have gone through the heads of several designers at STMicroelectronics (formerly SGS Thomson), the general idea having culminated in a series of ICs called 'PSD813xxx'. The individual components inside the chip are found back in **Table 1**. Obviously, this IC offers far more possibilities for a wide variety of microcontroller circuits.

Figure 1 shows a block diagram illustrating the internal structure of the PSD813xxx chips.

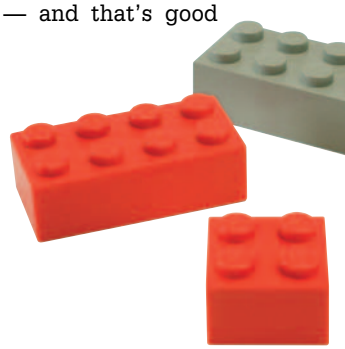
The memories may be mapped on a segment by segment basis. This creates many different configurations, for example, first 16 K Flash, followed by two 8 K EEPROM segments and then another 32 K of Flash memory.

The memory management is also very handy. For example, using the page registers allows you to increase the controller's address range. Another possibility is to configure the memory in a different way depending on the page register contents. The samples show how this is done in practice.

The CPLD section in principle obviates the logic that's normally required to drive peripheral devices, but only if the designer is satisfied with 27 I/O pins. For a controller with a multiplexed bus, this allows a de-multiplexed address to be created by the chip and, if necessary, a tri-state databus. Other applications include generating chip select signals for the peripheral devices or even creating extra I/O ports.

The PSD813 series has not been designed for one specific family of microcontrollers. In fact, it is so versatile that it can be used with several microcontroller families. These include not only micros with a standard address bus and databus, but also types with a multiplexed data/address bus like the 80C32, 8051XA and the 68HC11.

One final point to mention in this short profile is the JTAG port that allows all functions of the chip to be programmed in-circuit. In other words, it is not necessary to buy an expensive programmer — and that's good news.



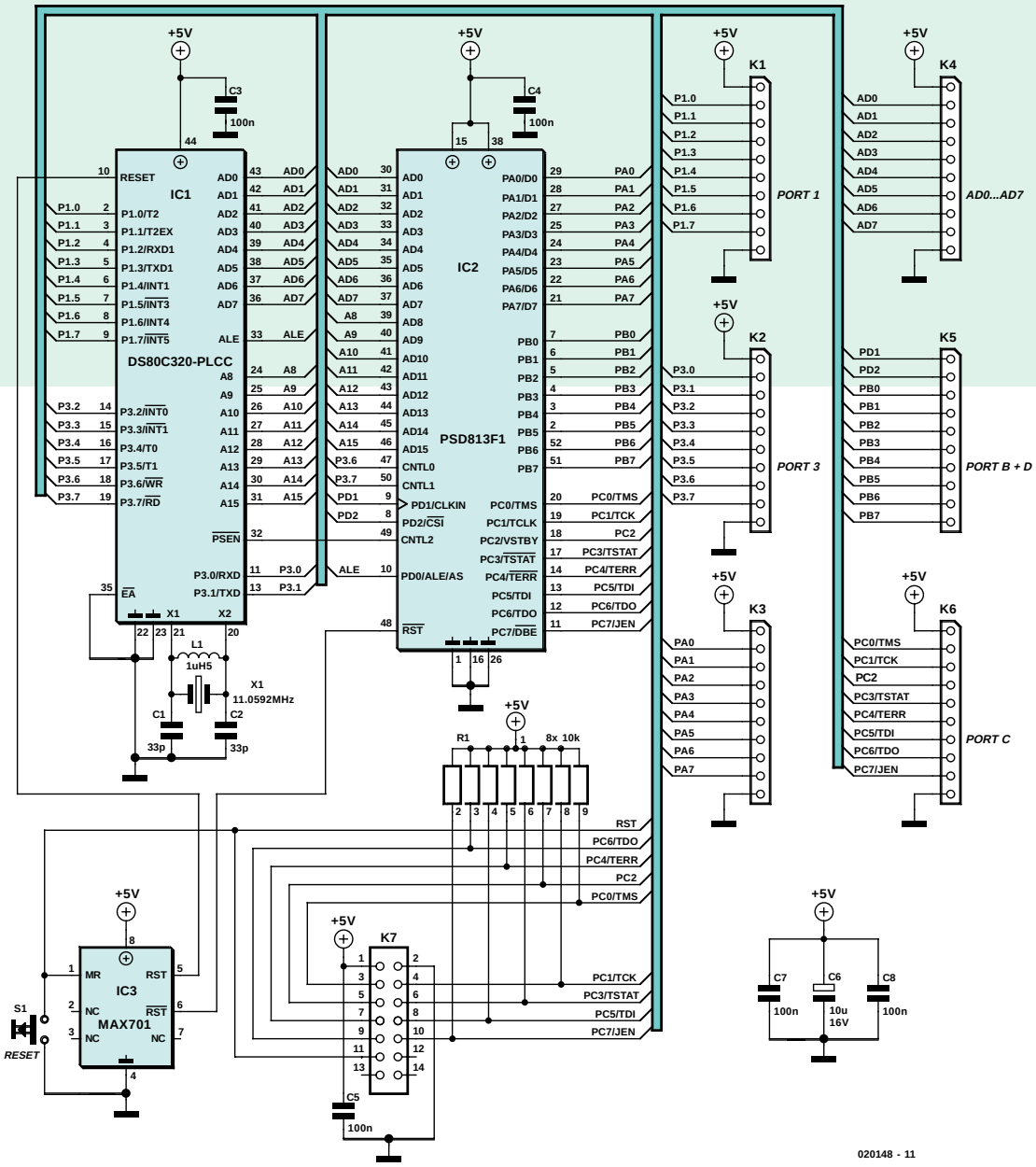
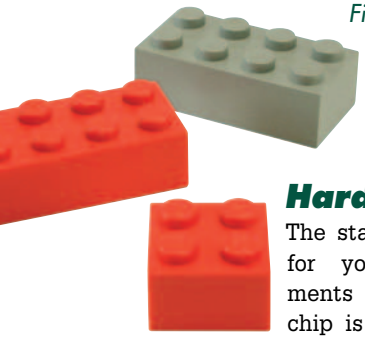


Figure 2. The application of a PSD813 results in a considerably smaller and simpler circuit diagram than you have come to expect from controller boards.



Hardware

The starting point for your experiments with this chip is a piece of hardware we developed that could be 'dropped' into several circuits. The circuit diagram of the drop-in module may be found in **Figure 2**. Even a cursory glance at the drawing should convince you that simplicity is trumps. Thanks to the multifunctional character of the PSD813 chip, this bit of hardware may be used in combination with a microcontroller in many different application circuits. What's more, the JTAG interface makes the total configuration great for use as an experimental or development system.

For the heart of the circuit we went for a DS80C320 (IC1). This is an 80C32 compatible micro that's much faster than the ordinary 80C32, hence will not let easily let you down in regard of processing speed. Its main pins are all connected to the PSD813F in position IC2. Furthermore, the circuit contains a microprocessor supervisor circuit built around IC3 and S1, some components to complete the clock oscillator and a handful of small caps to clean and buffer the supply voltage.

An on-board power supply is not provided because this module will typically act as an add-on to an existing circuit supplying +5 V, rather than a stand-alone unit. We did, however, bond out all relevant signals to pins on connectors K1 through K6. Connector K7 is the JTAG interface that allows the chip to be programmed. This interface has been described several times already in this magazine.

A suitable interface to connect the parallel port to the JTAG interface is shown in **Figure 3**. This programmer is directly accessible from the associated program called PSDSoft and that is why we believe it's really indispensable. The programmer is connected to the PC's parallel port via K8 and a cable with 1-to-1 pin correspondence. K9 is the JTAG port — this is con-

Table 1. PSD813 function overview (PSD813F1)

- Memory management
- 8 Flash RAM segments (128 kBytes)
- 4 EEPROM segments (32kBytes)
- SRAM (2 kBytes)
- CPLD (more than 3000 gates)
- 27 configurable I/O ports
- JTAG port
- Programmable power management
- Address and data demultiplexer

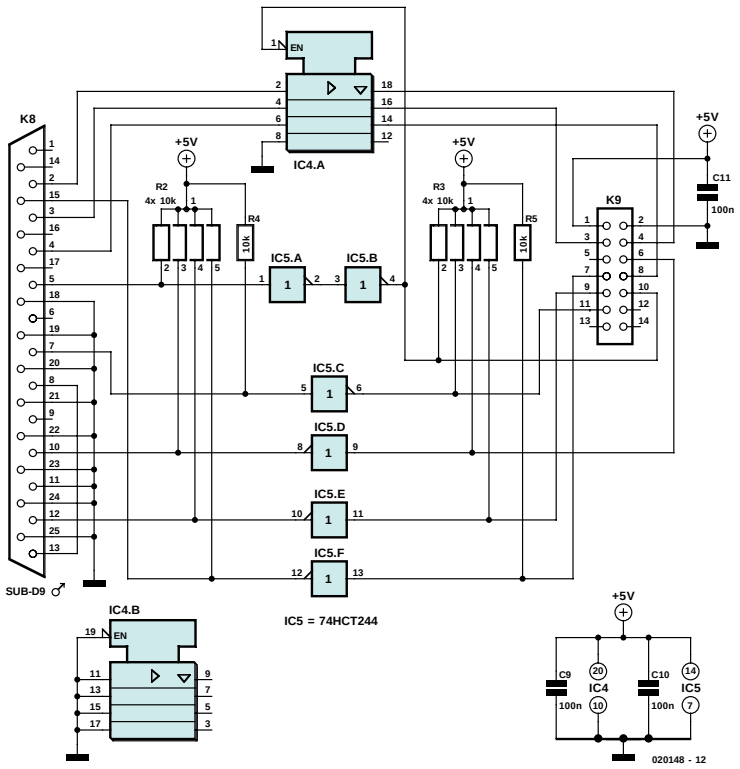


Figure 3. 'Glue' to connect the PC's parallel port to the JTAG interface..

nected to the plug-in module via a 14-way flatcable. IC5 affords buffering and inverting of the signals. The enable input is controlled by the PC, ensuring that the relevant signals do not load the circuit connected when in the 'off' state.

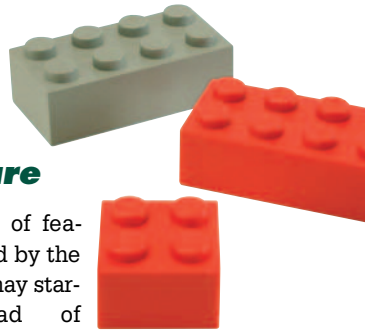
Building it

Having purchased the printed circuit board and all components you are ready to start building the circuit. As shown in **Figure 4**, the PCB consists of two sections that are easily separated using a hacksaw. One section is for the interface with the PC, the other, for the drop-in module proper.

In view of the relatively small number of parts, building up the two sub-boards is unlikely to take much of your time. As usual, we recommend you start with the low-profile parts. For the rest, hardly anything that can go wrong if you stick to the component overlay on the PCB and, of course, the parts list.

Connectors K1-K5 have to be soldered at the underside of the board, their task being to link the module to the host equipment in which is to function. **Figure 5** shows our ready-assembled and tested board.

Before we can test the circuit, we need to look at the software.



Software

The galaxy of features offered by the PSD813F1 may startle instead of encourage. A chip with so many functions is usually difficult to configure and you may quickly feel lost in bits and bytes. Fortunately, STMicroelectronics have written a clearly structured program called PSDsoft Express that may be obtained free of charge from their website at www.st.com. The documentation with the PSD813 series is impressive to say the least.

The program comes with a wizard guiding you through the configuration process in a step-by-step manner, using unambiguous questions. You'll quickly get the hang of the program simply by using it a few times.

To enable everyone to make a head start with the PSD813 chip we produced a small example program that may be downloaded free of charge from our website. The program demonstrates how extra ChipSelect signals and I/O ports may be generated and created respectively.

Having launched PSDSoft Express you are ready to create a new project. Programming the PSD chip could not be easier using the wizard. The individual pins of the PSD chip connected to the JTAG interface and the controller are immediately assigned names by the wizard. **Figure 6** illustrates the ease of defining an extra CS signal. Once all steps have been completed, the PSD chip is ready for programming through the JTAG interface.

Components list

Resistors:
R1 = SIL array 8 x 10kΩ
R2,R3 = SIL array 4 x 10kΩ
R4,R5 = 10kΩ

Capacitors:
C1,C2 = 33pF
C3,C4,C5,C7-C11 = 100nF
C6 = 10µF 16V radial

Semiconductors:
IC1 = DS80C320QCL (44-pin PLCC)
IC2 = PSD813F1 (RS Components # 417-5726)
IC3 = MAX701CPA
IC4 = 74HCT244P
IC5 = 74HCT04

Miscellaneous:
K1-K6 = 10-way SIL-header
K7,K9 = 14-way boxheader
K8 = 25-way sub-D plug (male), angled pins, PCB mount
L1 = 1.5 µH miniature choke
S1 = pushbutton, 1 make contact
X1 = 11.0592MHz quartz crystal
44-way PLCC socket
50-way PLCC socket
PCB, order code **020148-1** (see Readers Services page)
Disk, example project file, order code **020148-11** or **Free Download**

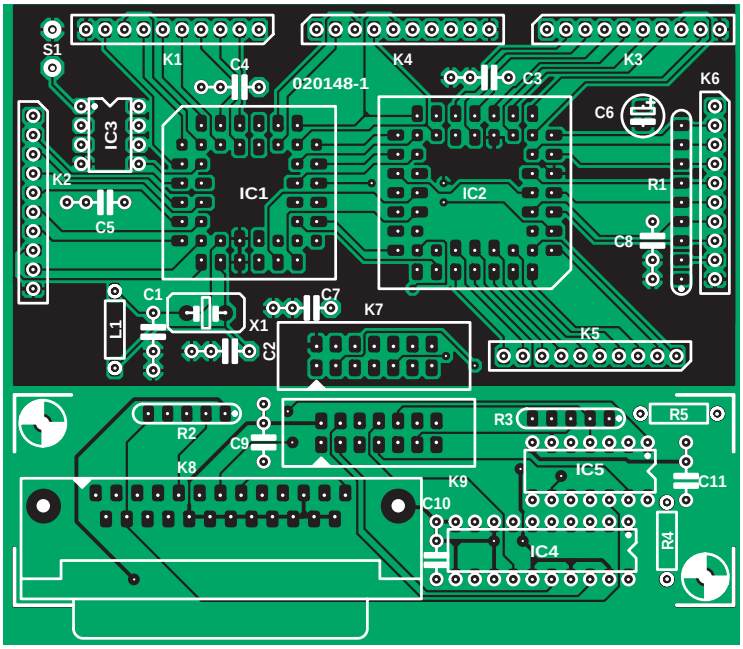


Figure 4. The PCB consists of two sections that are easily separated by sawing.

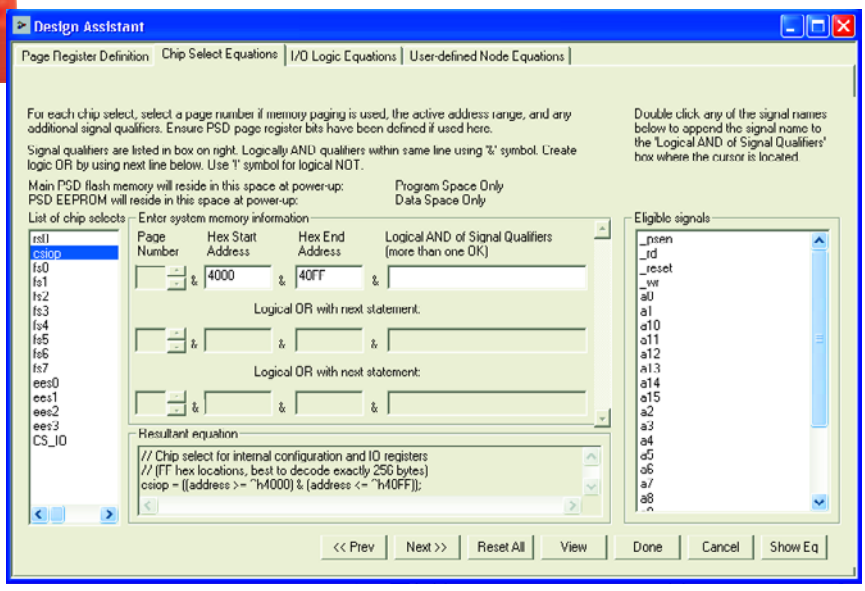


Figure 5. Using ST's PSDsoft Express to assign a software address range.

Finally

The PSD831 chip offers far more possibilities than those used in our simple example. None the less, the drop-in module discussed in this article will not fail to underline the versatility and multifunctional character of the PSD813xxx chips. Together with a suitable microcontroller, these chips form a solid base for a wide range of applications. The PSD813xxx allows much space to be saved in circuits normally requiring an EPROM, Flash memory and a handful of discrete logic.
Thanks to the extensive software support and the ability to be programmed through a JTAG interface, the chip will easily prove worth its salt during circuit development and debugging. Updates to the chip software and hardware (in the CPLD parts) are also easy to implement using the same JTAG interface.

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Free Downloads

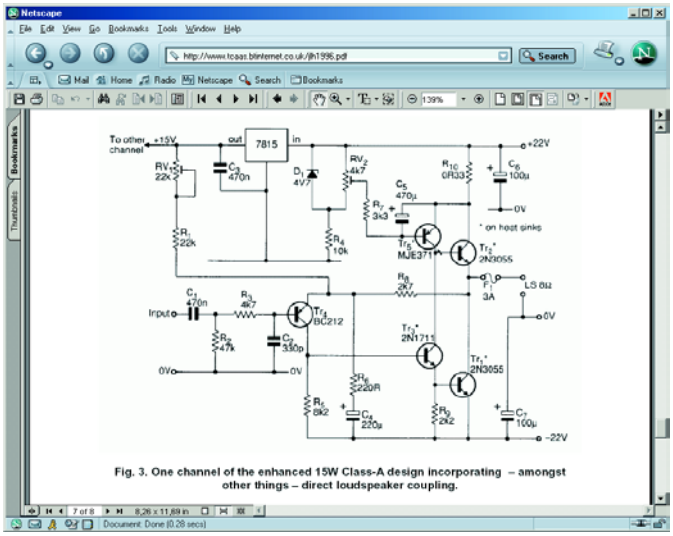
– PSD813 example project file.
File number: 020148-11.zip

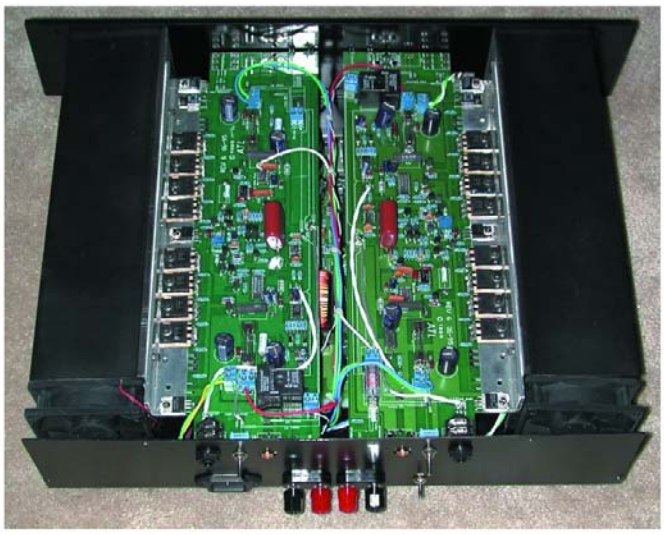
– PCB layout in PDF format.
File number: 020148-1.zip

www.elektor-electronics.co.uk/dl/dl.htm,

select month of publication.

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Audio Power Amps on the Internet

Fewer components, better sound?

Harry Baggen

Audio lovers will no doubt drool over the new Elektor preamp with digital control published in this issue. This, of course, calls for a matching power amplifier! Sure, there are enough high-quality Elektor designs to choose from, but it may also be interesting to trawl the Internet for alternative designs from all over the world.

Elektor Electronics is renowned when it comes to high-end audio designs. Our magazine frequently runs projects for pre-amplifiers and final amplifiers that are built by thousands of readers. The designs achieve excellent sonic quality at a fraction of the price of commercial products. It is in this tradition that we are proud to present, in this issue, our new digitally controlled preamp. The audio section of this preamp is marked by a minimalist design, guaranteeing superb sound reproduction while being successfully combined with a microcontroller and a display for extensive yet manageable control of it all.

Such an optimised design calls for a power amplifier with matching performance. Excellent candidates from our own stable include the IGBT Power Amp (June 1995), the Compact Amp (May 1997) and even the Crescendo Millennium Edition (April 2001).

Audiophiles may however also find interesting designs on the Internet, allowing them to 'follow their faith'. Some people will insist on keeping things as simple as possible, while others will not rest until their equipment is technically perfect.

The first category certainly embraces the legendary 'Zen' amplifier brought to us by American designer Nelson Pass, with follow-up versions carrying names like 'Son of Zen' and 'Bride of Zen'. Nelson is a master of electronic minimalism — the amplifier stage proper having just one active element, a power MOSFET. This is complemented by no more than a current source built around a power MOSFET and a transistor. The Zen amp delivers a modest 10 watts in Class A. Over the years a number of variations on the theme have emerged, many of these showing great activity from Nelson Pass himself.

Nelson Pass runs a website, **Pass DIY** [1], containing home-brew designs and covering all Zen projects in great detail.

Another website describing interesting audio power amplifier projects is called **ESP, Elliott Sound Products** [2]. One of the projects is ominously called 'Death of Zen', this is a Zen variant employing bipolar transistors. A version actually built from this circuit diagram is found at **Bebe** [3]. Believe it or not, it is built around the legendary (and still available) power transistor type 2N3055!

Another well-known and highly respected designer of, among others, class-A power amplifiers is John Linsley Hood, who in the 1960s became famous through his designs published in *Wireless World* (now *Electronics & Wireless World*). A large number of these 'evergreen' articles may be found at the **Class-A Amplifier Site** [4]. The site also supplies circuit diagrams of a headphone amplifier and a preamplifier as well as power amplifier projects from other designers.

Electronics enthusiasts who would rather rely on a more extensive design with a symmetrical layout and higher output power may be interested in the well-known power amplifier proposed by W. Marshall Leach. An extensive description is supplied by the designer himself at **The Leach Amp** [5].

A well elaborated design for an amplifier with relatively high output power (2 x 350 W into 8 Ω) may be found at a website called **A and T Labs** [6]. The power output section is built around complementary power MOSFETs. Another peculiarity of the design is the application of a switch-mode power supply unit (SMPSU) with a whopping output of 1 kW, allowing the entire assembly to be housed in a relatively low 19-inch rack.

To close off this article we'd like to draw your attention to **Schematic.info** [7], a website packed with circuit diagrams of twentieth-century audio classics like Dynaco, SAE, Quad, NAD and Crown. Take your pick!

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Internet Addresses

[1] **Pass DIY**:
www.passdiy.com/legacy.htm

[2] **ESP, Elliott Sound Products**:
http://sound.westhost.com/index.html

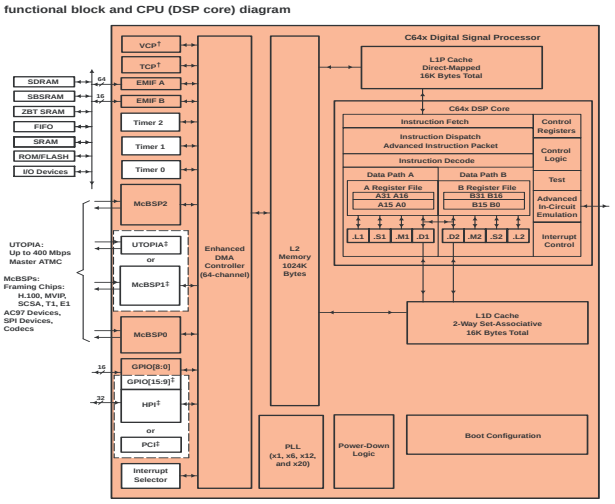
[3] **Bebe T**:
http://diyaudio.8m.com/Bebet/bebet.html

[4] **The Class-A Amplifier Site**:
www.tcaas.btinternet.co.uk/

[5] **The Leach Amp**:
http://users.ece.gatech.edu/~mleach/lowtim/

[6] **A and T Labs**:
www.a-and-t-labs.com/K6_Sw_Amp/

[7] **Schematic.info**:
http://schematic.narod.ru/a_index.htm



DSPs at 1 GHz

Top performance from 90-nm technology

The 90-nm process technology employed by Texas Instruments gives a spectacular boost to DSP performance. For the first time, DSPs pass the 1-GHz barrier.

DSPs (digital signal processors) type TMS320C6414/ 15/16 sample at 1 GHz, providing stunning throughput levels. They provide the user with 8 GigaMACs for 8-bit data width for video applications, or 4 GigaMACs for 16-bit data commonly used in speech and audio processing. This sort of throughput not only improves the bandwidth and channel capacity of existing real-time applications like base stations for mobile phones and radios, IP-based video, fast wideband networks, medical diagnosis equipment and radar, but also opens the way to new applications ranging from adaptive antenna arrays and intelligent vehicles right up to artificial vision.

A single 1-GHz component, for example, can now manage real-time transrating of eight MPEG-2 channels at D1 resolution (720x 480) or the processing of 55 GSM channels using AMR (adaptive multirate) speech encoding in a cellphone base station. In this way, designers no longer have to implement complex circuits using multiple processors.

The speed breakthrough has been achieved thanks to an innovative design method and improved manufacturing processes employing 90-nm technology, making it possible to produce faster and smaller ICs than ever before. TI's ability to pack transistors in an extremely tight space has resulted in dramatic increases in speed and chip density. The change to 90-nm technology reduces the chip size and with it the production costs, allowing almost 50% more chips to be made from a single wafer. This not only had a positive effect on the new chips, but also allowed the price of the existing 720-MHz 'C64x series to be reduced by more than 50%. The 90-nm technology also simplifies the integration of System-on-a-Chip architectures by rationalising the communication between the DSP core, memory, peripherals, RISC processors and analogue components. These innovations comprise a command pipeline for efficient use at frequencies greater than 1 GHz, a dual data architecture with 32 32-bit registers per data path and optimized implementation of data functional units required for the execution of critical program paths.

The new components are software compatible with the earlier generations — the 'C64 command set was not changed, so the design and process improvement are virtually invisible to software developers. The new DSPs being pin compatible with the earlier versions, they can easily be dropped into existing designs, boosting the data throughput, lowering energy consumption and keeping system costs within reason.

The 1-GHz DSPs types C6416, C6415 and C6416 are currently only available as samples. All three components have an on-chip 1-MB memory and only differ in respect of their integrated peripheral devices. The DSPs will be available in volume quantities by Autumn 2004. Unit prices start at US$ 189.

To prevent time loss, software engineers can avail themselves of the C64x Starter Kit and make a head start with the design of key software elements of a target system, including algorithms, command codes and initial software system integration. The C64x kit contains didactic exercises, reference framework support, the integrated Code Compose Studio development platform, the DSP/BIOS real-time kernel and a number of standard algorithms. The Kit is already available at a cost of US$ 395.

Further information on this interesting starter kit may be found at www.ti.com/c6000dsksp.

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Contact information:

European Product Information Center (EPIC)
Texas Instruments Germany
Haggertystrasse 1 - D-85350 Freising - Germany.
Tel. (+49) 8161 803311
Email: epic@ti.com
Internet: www.ti.com/sc/epic/
www.ti.com/1ghzsamplingp

64-k 80C552 Flash Board Dear Editor, regarding the above project (January 2004, Ed.) I am at a loss as to which software to obtain for the board and where to find it. Can you help me?
Mike Miller (by email)

Well Mike the article clearly states at several points, including the Free Downloads inset at the end, that the same software is used as with the Precision Measurement Central (a MSC1210 development system; cover item, July/August 2003, Ed.). Not to worry, though.
Go to www.elektor-electronics.co.uk/ dl/dl.htm, then to issue 324 (September 2003), then select and download the large (!) file 030060-1b.zip. The folder structure of this 'beast' is shown in Elektor Electronics September 2003 on page 54. The 80C552 Flash board also requires file number 030042-11.zip which may be found under issue 328 (January 2004). This zip file contains the source code files of the OS552 and Hello programs, as well as Flash552.hex and the GAL source and JEDEC files. As you can see, we've got it taped!

Long(ish) wire Dear Sir, the Longwire Match for SW Receivers article on page 111 (July/August 2003, Ed.) has a couple of errors. A long wire antenna is generally considered to be long with respect to the wavelength received, that is, at least 10 times the wavelength. The 3-m wire mentioned is actually a short antenna at HF!
A balun's primary function is to convert a balanced line to an unbalanced one, not to step down an antenna's impedance.
The circuit is simply an impedance converter for matching a short antenna to 50 ohms, not a balun equivalent for a long wire.
Leon Heller, G1HSM (by email)

So it is true — length does matter! We stand corrected yet hope the inappropriate use of the term 'long wire' does not detract too much from the usefulness of the circuit.

CoolRunner-II — not so cool Dear Jan, I was surprised and somewhat disappointed to see that the Xilinx CoolRunner-II Development Kit reviewed in your February 2004 issue could not actually be ordered from the Xilinx website using the hyperlink you printed. The product seems to have been discontinued. As I am still interested in making a head start with CPLDs, I'd like to know what, if any, replacement product you or Xilinx recommend?
Stuart Galbraith (by email)

Things are moving fast in the silicon chip industry — faster than we can print pages and get them into the newsstands or your mailbox. Literally minutes after hitting the Return key and sending our magazine to the printers we too discovered that the product had been removed from the Xilinx website. Fortunately, within days Xilinx advised us that the CPLD Design Kit (part # DO-CPLD-DK) is a perfect substitute. We were glad to see that the new kit contains even more goodies than the Coolrunner-II while the price has remained the same at $ 49.99 plus shipping.

Where to start? I am currently travelling the world and have just bought your magazine in OZ (is that Denmark? Ed.). I come from the UK but have never seen your magazine there.
I wonder if you could help me. I would very much like to learn about electronics but after years of trying to find evening courses, I gave up. Do you have any suggestions as to how a complete idiot to electronics can learn about electronics. I am not very good at learning from books unless it has a large practical content.
Tim (by email)

Hello Tim, and welcome to Elektor Electronics. We envy you, being confined to our offices and answering readers' letters. Amazing you have never seen our magazine in the UK as it has been around in vast quantities since 1975. May I suggest you follow our Mini Project article series? The complete series is available free of charge from our website, simply click on Mini Projects Online. We're sure you'll find the content far from bookish.

Mirror mirror on the wall Dear Editor, I have a problem with the PCB layouts in PDF format. I need to print a mirror image because I use the 'print-it-on-a-laser-printer-and-iron-it-on-the-copper' method of making PCBs. The Minolta Pro 6 driver for Win 2000 has no provision for printing an mirror image, and Acrobat Reader does not seem to do it by itself. Moving the design from Acrobat Reader to PaintShop Pro or some other graphics program seems to destroy the fine lines by pixellation, if you know what I mean. Can you help?
Per Troelsen, probably the most faithful subscriber in Denmark (15+ years) (by email)

Per, despite being a faithful reader you may have missed that since mid 2001 our PCB downloads contain pdf files with non-reflected as well as reflected artwork. See also 'Printing PCB Artwork' in the May 2003 issue.

Alarm! Clock deviation! Gentlemen, the timing error admitted in the Digital Alarm Clock article (February 2004, Ed.) — 0.256 ms slow per minute — amounts to just over a third of a second per day, or two and a quarter minutes per year. However, looked at another way, it is equivalent to the 4 MHz master oscillator running 17.067 Hz slow. In practice, this is far less than the likely frequency error for an unadjusted oscillator of this type — possibly by an order of magnitude! Might I suggest replacing one of the fixed capacitors C1 or C2 with a small preset trimmer? Then adjusting the oscillator to 4,000,017.067 Hz will allow the clock to keep 'perfect' time.
Mathematically, at least, it is also easy to correct the division ratio so that an accurate 4,000,000.00 MHz clock (adjusted as above) will keep time, but whether this is actually possible with the PIC only you can answer. Apparently you use the overflow of TIMER_0 to increment a 16 bit counter. If this 8-bit TIMER_0 counter could be preset so that it divided by 160 instead of 256, and the 16-bit counter set for a maximum count of 46875, this would give a precise 60 second interval. If TIMER_0 can only divide by 256, could the prescaler be set to divide by 5 instead of 8? That would achieve an identical result.

It would be ideal if the division ratios could be configured accurately as described, then the 4 MHz oscillator could be more easily adjusted; a frequency reference would be available for other uses, and there would be the possibility of using an external 1 MHz clock. However, failing that, simple adjustment as described in the first paragraph would at least give accurate time-keeping.
Peter Vince (by email)

Te designer of the circuit, Manoel Conde de Almeida, replies: You are right. The 0.256 ms error is the admitted error considering that the crystal oscillator will run at the nominal crystal oscillation frequency (in this case 4 MHz). Of course, we all know that off the shelf crystals have a tolerance of around 100 ppm.
The idea of replacing one of the capacitors by a trimmer looks good and seems to be the most effective one. But we have to remember that it would require special equipment (a calibrated frequency meter, for instance) for the adjustment process. Some readers interested in building the clock may not have that kind of infrastructure available.
Using the firmware (mathematics) is an alternative but, since we never know the exact frequency the oscillator will be running at (assuming that the user does not have equipment to measure/adjust the frequency), it becomes more difficult to develop a 100% accurate solution.

Anyway, along that line of thought, I have worked on a firmware solution that tries to minimize the issue by giving the user the opportunity to change the maximum count of the register that keeps track of the TIMER_0 interrupts and updates the clock's seconds counter.
The user may select 1 of 3 operating modes depending on the clock's behaviour (slow, OK, fast). Each mode establishes different maximum counts for the TIMER_0 counter.
It doesn't solve the problem entirely but minimizes the error when the oscillator is running near the limits of the crystal tolerance. The solution, though, works with the PIC16F628 and not the PIC16F84A because of programming memory space. If you're interested I can share more information and exchange more ideas with you and other Elektor readers.

Self-discharging NiMH batteries Dear Jan, in a recent article on digital cameras published by a renowned computer magazine I stumbled on the following rather bold statement: "NiMH batteries suffer from energy loss at a rate of about 1% per day". I am not an electronics buff and although the statement does not come from any of your articles I would still invite your comment?
G. Lisabeth (by email)

The statement is correct. All cells lose energy owing to self-discharging and NiMH batteries are no exception. Although the exact rate of discharge depends on the cell type (i.e., technology) '1% per day' is a generally accepted value.

ECD CD-ROM Dear Editor, I recently bought a copy of your ECD CD-ROM. I have a few questions on the product:
1. Reference is made to an elusive 'booklet' explaining the installation and use of the CD-ROM. What I was able to find was a leaflet (inlay) giving just the minimum system requirements.
2. Is it possible to use the CD-ROM without having to copy the lot to hard disk? If so, how is it done? I was unable to find any guidance.
3. Eventually I did install the product on hard disk and found it a pleasure to work with. My only objection is that a subject has to be closed before another can be selected, or am I missing something? Otherwise, full marks for the ECD!
Robert Fruytier (Netherlands)

Thanks for your positive criticisms Robert, and glad to read that you like the product. As the installation is mostly self-evident, any sort of description is superfluous and would not fill a booklet anyway. There's no way you can avoid copying all data on the CD-ROM because the immense size of the component database makes it impossible to handle from CD-ROM particularly on slower PCs.
There's no need to close subjects every time. Several subjects may be opened in sequence and you may switch between them using the ALT+TAB key combination (yes the method dates back to Windows 3.1). The selection menu is also accessible by right-clicking on the icon in the status bar.

Economic forces Dear Publisher, thanks for your notice to renew my subscription to Elektor Electronics this year. Your information that you have to increase the subscription rate yet again this year due to economic reasons, I fully understand. Unfortunately, due to the strong British Pound and your increases of subscription over the last three years, the cost of your magazine is no longer affordable for myself. This is very disturbing for myself, but having been a reader and subscriber of this magazine for over 26 years, I have, with a heavy heart, decided not to continue my subscription to your wonderful magazine. The main reason is that, being a pensioned person and still active in various fields of electronics, I just cannot afford it anymore!
I wish you all the best and hope you will continue to produce a wonderful and A-1 magazine!
Sejjad Salam (Slovenia)

Sejjad, we were sorry to read your email but fully appreciate your problems and thank you for having been a subscriber for so long. Despite the recent price increases we can still claim to offer the best cost per circuit figure of all electronics magazines published in the UK.
We hope that in the future you will still be able to pick up the odd copy of Elektor from a newsstand if a subject to your liking is published (do check our website — it is free).

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It would be ideal if the division ratios could be configured accurately as described, then the 4 MHz oscillator could be more easily adjusted; a frequency reference would be available for other uses, and there would be the possibility of using an external 1 MHz clock. However, failing that, simple adjustment as described in the first paragraph would at least give accurate time-keeping.
Peter Vince (by email)

Te designer of the circuit, Manoel Conde de Almeida, replies: You are right. The 0.256 ms error is the admitted error considering that the crystal oscillator will run at the nominal crystal oscillation frequency (in this case 4 MHz). Of course, we all know that off the shelf crystals have a tolerance of around 100 ppm.
The idea of replacing one of the capacitors by a trimmer looks good and seems to be the most effective one. But we have to remember that it would require special equipment (a calibrated frequency meter, for instance) for the adjustment process. Some readers interested in building the clock may not have that kind of infrastructure available.
Using the firmware (mathematics) is an alternative but, since we never know the exact frequency the oscillator will be running at (assuming that the user does not have equipment to measure/adjust the frequency), it becomes more difficult to develop a 100% accurate solution.

Anyway, along that line of thought, I have worked on a firmware solution that tries to minimize the issue by giving the user the opportunity to change the maximum count of the register that keeps track of the TIMER_0 interrupts and updates the clock's seconds counter.
The user may select 1 of 3 operating modes depending on the clock's behaviour (slow, OK, fast). Each mode establishes different maximum counts for the TIMER_0 counter.
It doesn't solve the problem entirely but minimizes the error when the oscillator is running near the limits of the crystal tolerance. The solution, though, works with the PIC16F628 and not the PIC16F84A because of programming memory space. If you're interested I can share more information and exchange more ideas with you and other Elektor readers.

Self-discharging NiMH batteries Dear Jan, in a recent article on digital cameras published by a renowned computer magazine I stumbled on the following rather bold statement: "NiMH batteries suffer from energy loss at a rate of about 1% per day". I am not an electronics buff and although the statement does not come from any of your articles I would still invite your comment?
G. Lisabeth (by email)

The statement is correct. All cells lose energy owing to self-discharging and NiMH batteries are no exception. Although the exact rate of discharge depends on the cell type (i.e., technology) '1% per day' is a generally accepted value.

ECD CD-ROM Dear Editor, I recently bought a copy of your ECD CD-ROM. I have a few questions on the product:
1. Reference is made to an elusive 'booklet' explaining the installation and use of the CD-ROM. What I was able to find was a leaflet (inlay) giving just the minimum system requirements.
2. Is it possible to use the CD-ROM without having to copy the lot to hard disk? If so, how is it done? I was unable to find any guidance.
3. Eventually I did install the product on hard disk and found it a pleasure to work with. My only objection is that a subject has to be closed before another can be selected, or am I missing something? Otherwise, full marks for the ECD!
Robert Fruytier (Netherlands)

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