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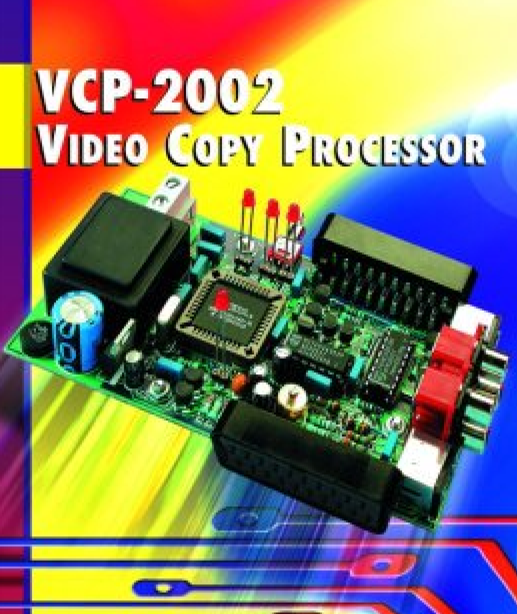
VCP-2002
Video Copy Processor



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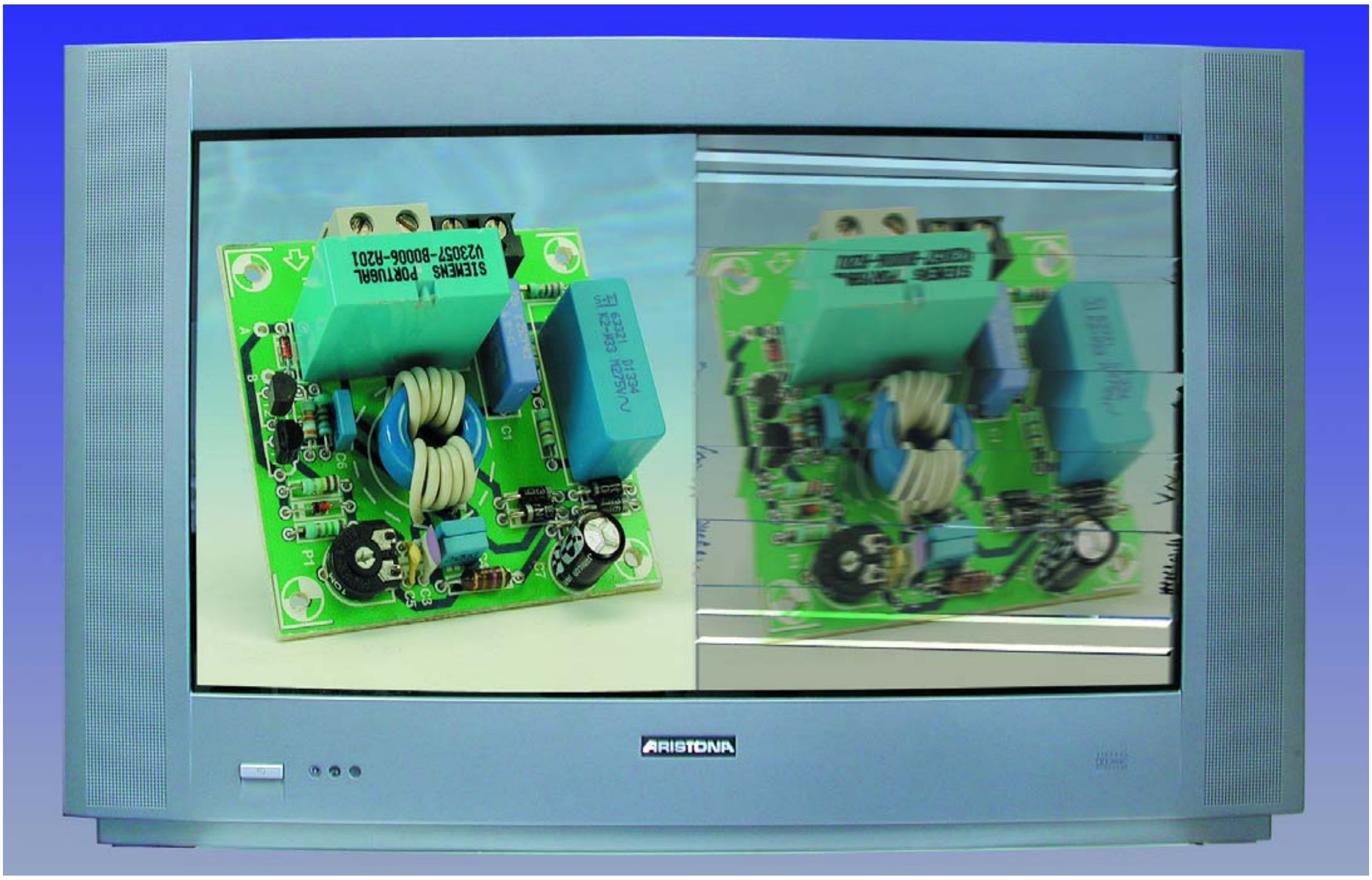
VCP-2002 Video Copy Processor

for DVD, S-VHS and VHS

Design by W. Foede

Due to lack of standards, the muddled situation regarding copy protection and the relatively high price, buying a DVD recorder is not yet recommended. Analogue video recorders are sure to fill this gap for quite some time to come. However, they can only be used for tape-to-tape or DVD-to-tape copying if certain interference signals are removed.

The Publishers dissociate themselves from any unlawful use of this circuit involving the infringement of copyrights resting on video media such as magnetic tapes and DVDs.



Analogue picture recording is slowly but gradually being replaced by its all-digital equivalent. We should emphasize 'slowly' because the changeover is hampered by the cost of DVD recorders, blank DVD disks and the insecurity about recording formats to be adopted worldwide. It would be gross mistake to claim that analogue video recording is dead, on the contrary! Excellent S-VHS recorders can be bought for around £ 150 and even if you take into account the slightly higher cost of tapes, the total cost/performance balance of a modern S-VHS recorder is still very attractive.

Even with DVD, analogue signals are still used, mainly because all-digital copying of video signals will take some time to fully mature. If and when that happens, you can be sure that a very powerful copy protection system will be built into the hardware and software to be developed in the near future.

The output signals of a modern DVD player include the following:

- composite video (CVBS — colour, video, blanking, sync) for connection of VHS recorders via the SCART socket;
- Y/C (luminance/chrominance) for S-VHS via the Hosiden socket (a 4-pin DIN socket);
- stereo audio (L/R) via cinch sockets.

Analogue copy protection

Both video outputs feature copy protection using the Macrovision system. The only difference as compared with a VCR is the source of the purposely introduced interfering signals. On tapes, the Macrovision signals is present in each picture blanking pulse. By contrast, the DVD player features a Macrovision IC that's activated by the DVD's internal command set and actually adds the interference to the output signals. Because these signals are not degraded by analogue means such as a magnetic tape, their quality is very high.

The basic functions mentioned above are illustrated in **Figure 1**. As opposed to a TV set, the recording device features an automatic video recording circuit. The circuit strives to maintain a constant input level for full contrast, independent of picture contents. The reference used is the difference between the 'black' level immediately after the line sync pulse ('rear porch') and the level of the sync pulse. This value (for example, 0.3 V_{pp}) is independent of the picture contents and is held constant (within limits, of course) by the recording circuitry.

Macrovision inserts seven new line sync pulses between the existing ones in ten picture lines in the blanking period (which is the invisible part of the picture). These extra pulses are followed by a jump to white level, creating a sudden difference level of about 1 V_{pp}. The VCR recording circuitry will respond by reducing this to 0.3 V_{pp} but in doing so also reduces the total CVBS level by about a third. As a result, the picture goes dark, synchronisation will start to fail, and sound and colour may become intermittent. To make things even worse, the interference is only stable for a short time before its amplitude is changed. The upshot is that the picture seems to err between too dark and too bright only. Additional interference is (partly) inserted into the visible section of the picture. The rear porch behind the real line sync pulses is switched to white and back to black again in the middle of the colour burst. In this way, the burst, reference and colour are corrupted and you will see severe picture distortion in the upper and lower picture areas. This distortion is not only visible on copies of Macrovision tapes, but also when playing back original tapes on older TV sets.

From VHS to S-VHS vice versa

The VCP-2002 Video Copy Processor described in this article will not only remove interference signals — it may also be used as a VHS to S-VHS or an S-VHS to VHS converter! Any combination is possible. The VCP-2002 has SCART as well as Hosiden/cinch inputs and outputs available for connecting up all sorts of video equipment. If, for instance, both inputs are simultaneously connected to a DVD player and a video recorder, the VCP-2002 will automatically select the input where an active signal appears. When both inputs are active at the same time, all outputs are muted in automatic mode, allowing a particular input to be selected manually. The outputs are always available simultaneously.

The VCP-2002 is a repeatable project that does not require anything in the way of 'spe-

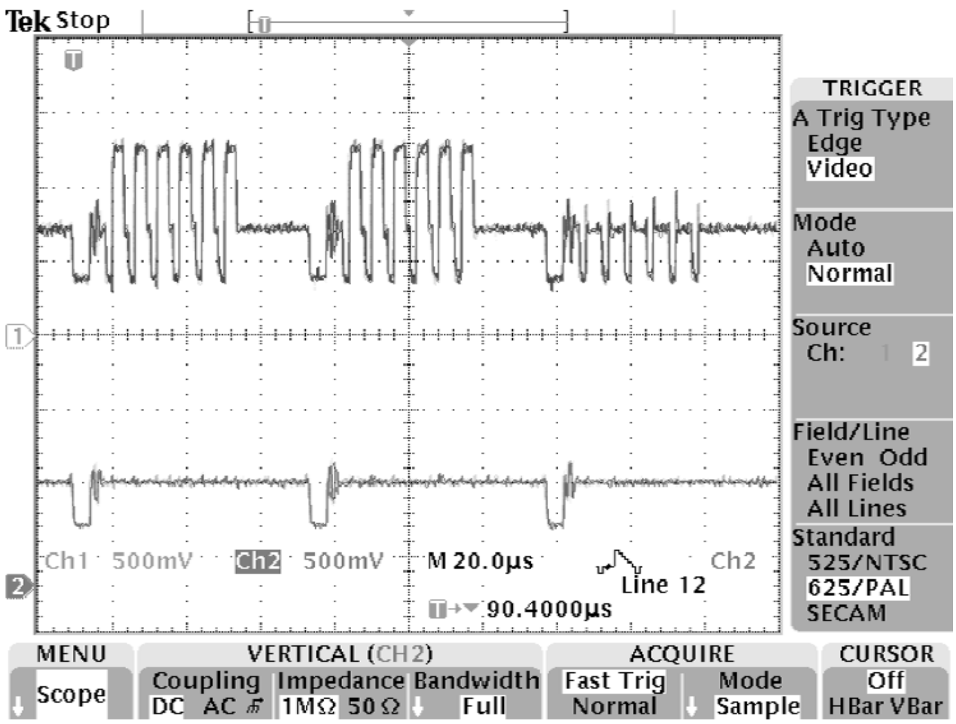


Figure 1. Oscillograms showing a normal video line and one 'infested' with Macrovision-generated copy protection interference.

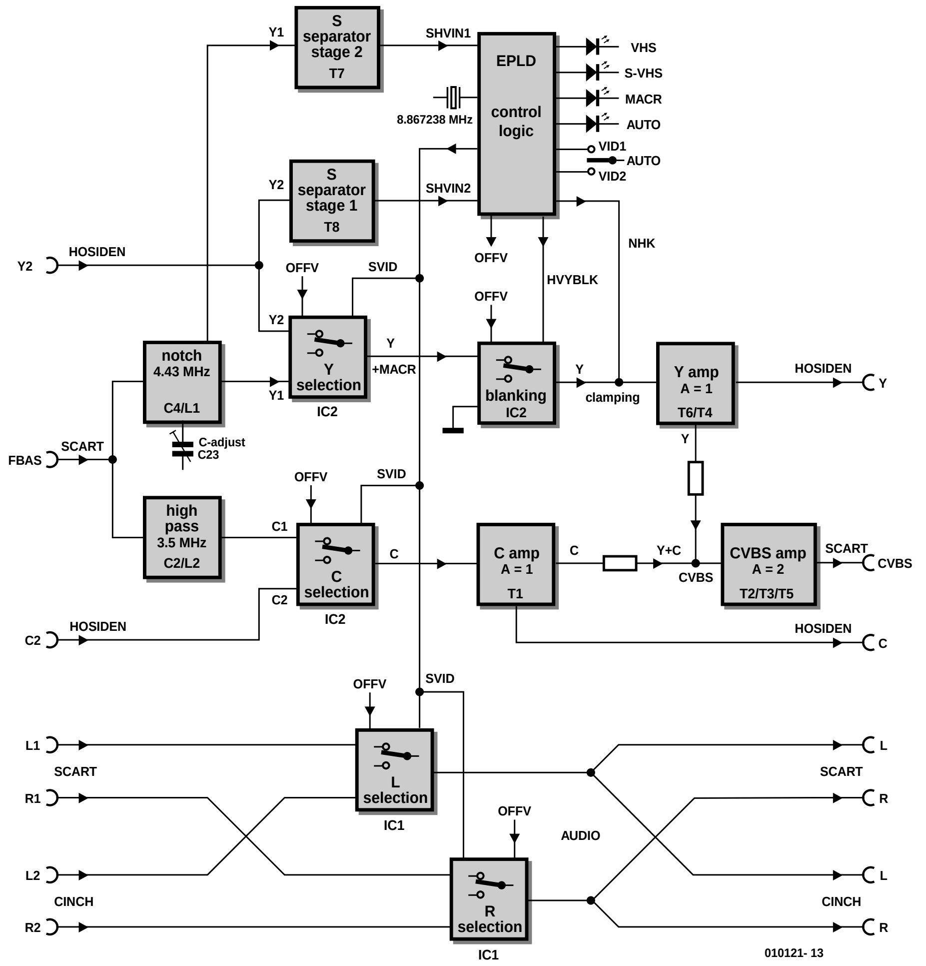


Figure 2. Block diagram of the VCP-2002.

cial adjustment tools'. Only one trimmer has to be tweaked for best picture reproduction.

The above requirements are way beyond those of a circuit thrown together on a rainy Sunday afternoon. A look at **Figure 2** will convince you that the VCP-2002 contains quite a few functional blocks. Salient part references make it easy to recognize these functions in the actual circuit diagram of the VCP-2002.

The video input signal always contains both the components **Y** (luminance) with a frequency range of 50 Hz to 5 MHz (including the line and raster syncs) and **C** (colour) comprising the 4.43-MHz colour burst and an associated bandwidth of –0.6 MHz to +1.2 MHz. VHS processes these components in one (CVBS) signal, while S-VHS handles them separately (differential Y/C).

From the CVBS signal, the VCP-2002 extracts the components Y1 and C1. Y1 is extracted using a notch at 4.43 MHz (C4-C23-L1-R10) and C1 using a high-pass at 4 MHz. At the same time, the Y1 signal is copied into the picture sync signal SHVIN1 by means of a synchronisation separator. The filter sections need to be fairly wideband, eliminating the need for high Q factors. That is why damping resistors (R10 and R3) are employed. A further advantage of relaxed Q requirements is that ready-made inductors such as miniature chokes may be used. No need to wind coils for this project!

The sync separator contains discrete parts. Sync pulses are the most negative-going components of the Y signal. They are clamped at 0.6 V below the supply level (+5 V) using C7-R16 and the base-emitter junction of T7, which is driven hard. At the collector (R19) we find the positive sync pulses with a level of 5 V_{pp}. These are used for the automatic switch-over and the positioning of the blanking signals. The line components are filtered by the EPLD, the raster components, by the two-section integrator R28-C18-R32-C17.

An S-VHS recorder will supply Y2 and C2 as separate components. The sync pulse are again extracted as described above (but using T8). The only difference is the lowpass R15-C11, which rejects unnecessary frequency components above 2 MHz or so. In this way, all input signal components are individually available for further processing. Audio and C signals are taken to the corresponding outputs via electronic switches. The input signal selection (between S-VHS and CVBS) is automatic or manual by means of S1.

A pre-programmed EPLD chip handles the automatic input selection (SVID) as well as the mute signal (OFFV). However, this is a minor job — the main task of the EPLD is to detect and suppress the Macrovision interference at the right instants.

Control logic in EPLD

All interfering signals have been superimposed on Y1 and Y2 — in six lines before and three lines after each raster sync pulse. A complete picture line lasts 64 µs, 52 µs being taken up by the visible line contents, which is, however, suppressed during the picture flyback period. In this period we find (but do not see) other information like Teletext, VPS and Video Test signals. The line blanking period lasts 12 µs and contains the line sync pulse as well as the colour burst.

All interfering signals are replaced by black levels. This can be done provided the instantaneous level for black is constant and independent of the picture content. This is achieved with the aid of a switched clamping circuit. Using a signal called NHK, the voltage level of the porch following the line sync pulse is pulled to ground for 3.5 µs by the EPLD chip (see **Figure 4**). This enables tantalum capacitor C8 (C6) to be charged to the instantaneous level of Y. Until the next blanking pulse, NHK switches to tri-state (high impedance). The charge built up in the capacitor remains constant, creating a kind of voltage source in series with the Y signal. This fixes the level for black at 0 V for the rest of the line. To enable the interference to be removed, Y only needs to be pulled to ground by means of the HVYBLK signal from the EPLD. The correct instant is determined by several functional blocks, comprising a line counter (clock = line, reset = picture); a picture element (pixel) counter clock = Q1 at PAL xtal 8.86 MHz), reset = line); and logic within the EPLD. To prevent the capacitors from being discharged between blanking pulses, the amplifier input to which they are connected should present as light a load as possible. Hence a field-effect transistor (FET) is used — here, a type BF245B. As opposed to a bipolar NPN transistor, the FET includes a diode in the

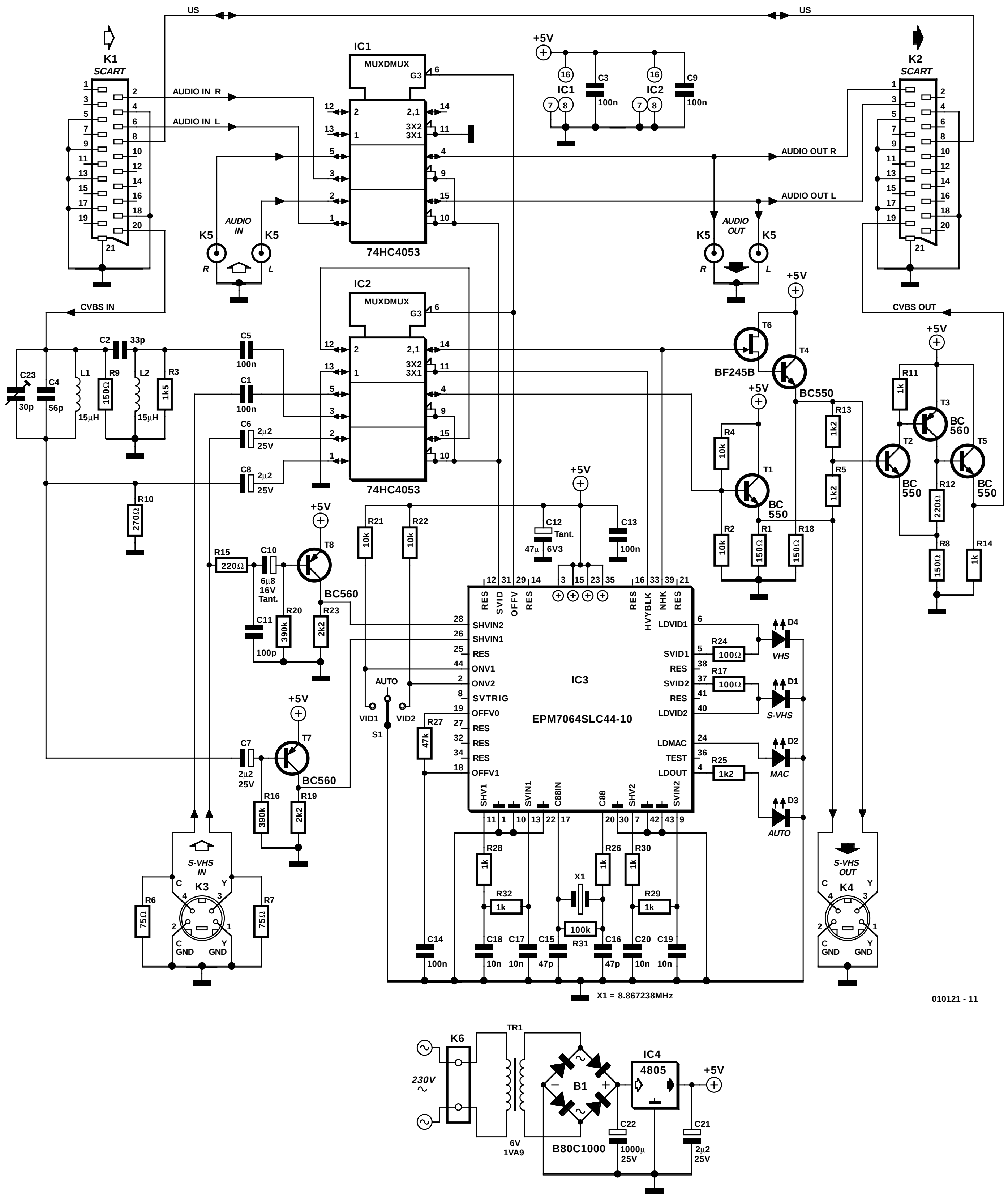


Figure 3. The circuit diagram of the VCP-2002 Video Copy Processor is heavily dominated by an EPLD chip.

blocking direction which makes it a very high-impedance device. Care should be taken to use the -B suffix device (BF245B) which has the right pinch-off voltage for this application. The pinch-off voltage is the positive level applied to the gate at which no current flows through the drain-source junction. The FET is connected in a source-follower configuration. The direct voltage at the source, and consequently that at the base and emitter of T4, corresponds to the pinch-off voltage. With the BF245B specified for 2 V in this respect, the output amplifier is properly biased within the available voltage range. This obviates the need for large electrolytic capacitors at the output.

With the interference signals suppressed, Y appears as an individual signal at the Hosiden socket, or combined with C (via R5-R13) to give CVBS at the SCART output (K2). This signal needs to be amplified by a factor of about 2 and that function is handled by T2, T4 and T5.

Four LEDs are incorporated into the design to indicate what the VCP-2002 is doing. Without signals but with the supply switched on you'll find that LEDs D1-D4 light dimly depending on the setting of S1. When input signals are applied, the LEDs light brightly. D2 in addition indicates the presence of Macrovision interference in the source signal.

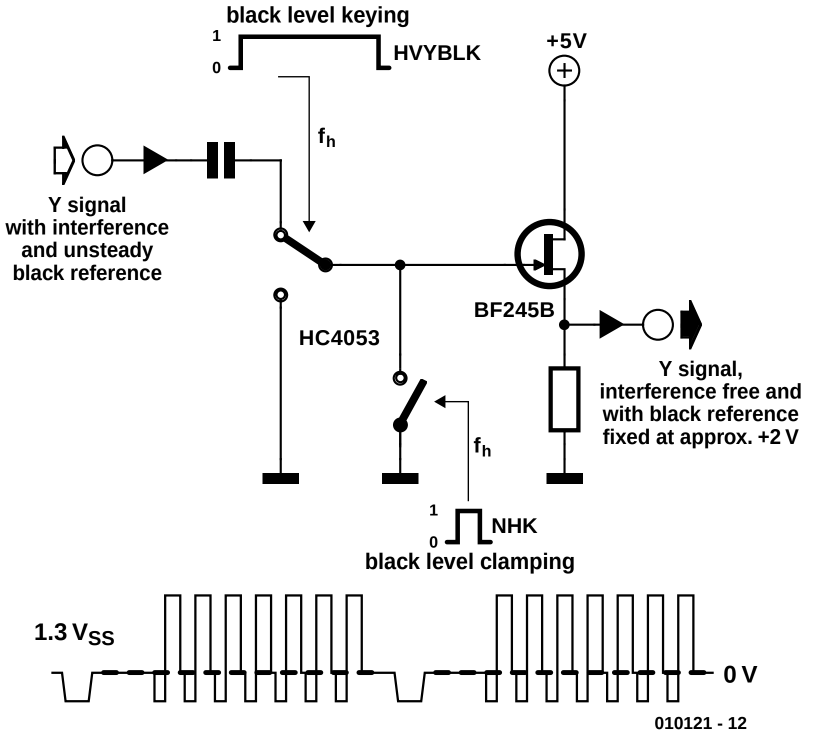


Figure 4. Principle of Macrovision detection and removal.

A miniature PCB

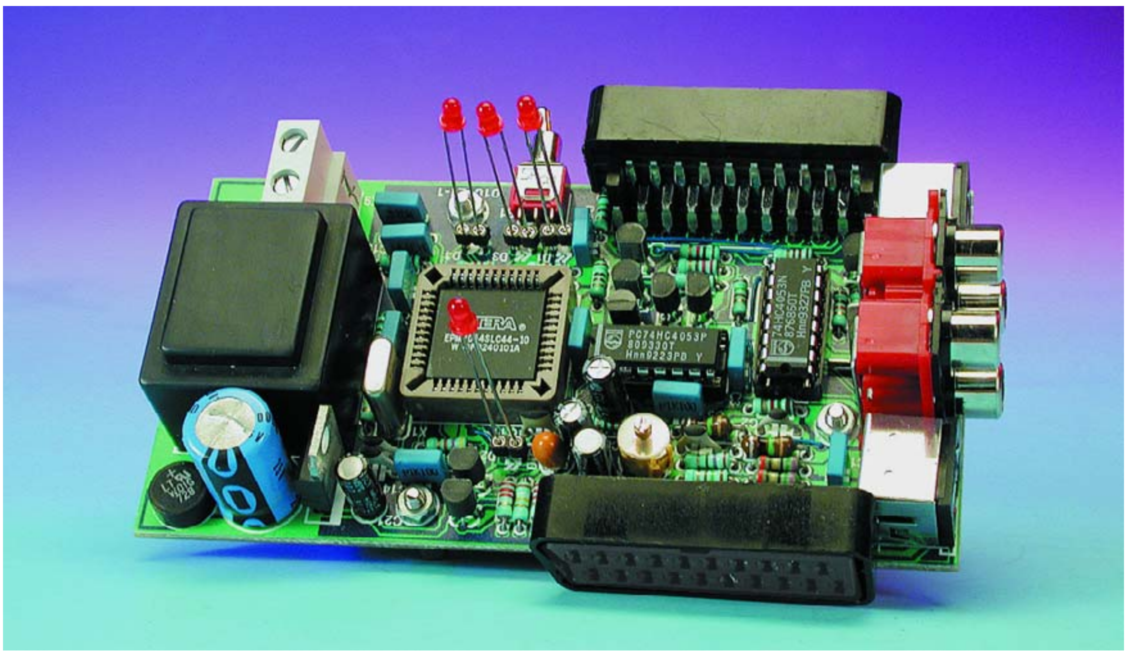
Despite the relative complexity of the VCP-2002 circuitry, the PCB shown in **Figure 5** is remarkably small and single-sided, too! The PCB accommodates all components including a mains power supply, and contains only one surface-mount device (SMD). With the resistors, their leads should be bent as close as possible to the resistor body. Alternatively, you could consider using miniature resistors (0.125 W) with a lead pitch of 7.5 mm.

Make sure you know the polarity of the tantalum capacitors before fitting them. The positive lead is (usually) indicated by a + sign, a line, a longer wire, or it is at the right-hand side when reading the type/value print on the device. C12 is the SMD mentioned above. It is soldered underneath the PLCC socket, at the copper track side of the board.

You should also not forget to fit two insulated wire links **underneath** IC1, two beside IC1, one below the PLCC and one beside L3.

The populated and tested PCB fits in a small ABS enclosure. The LED leads keep their original lengths to act as 'spacers' so the LED tops can protrude through the top panel of the enclosure. The LED leads may also be plugged into 2-way sockets. This will prevent damage to the PCB if you fit the LEDs the wrong way around.

Finally, to adjust the colour trap in



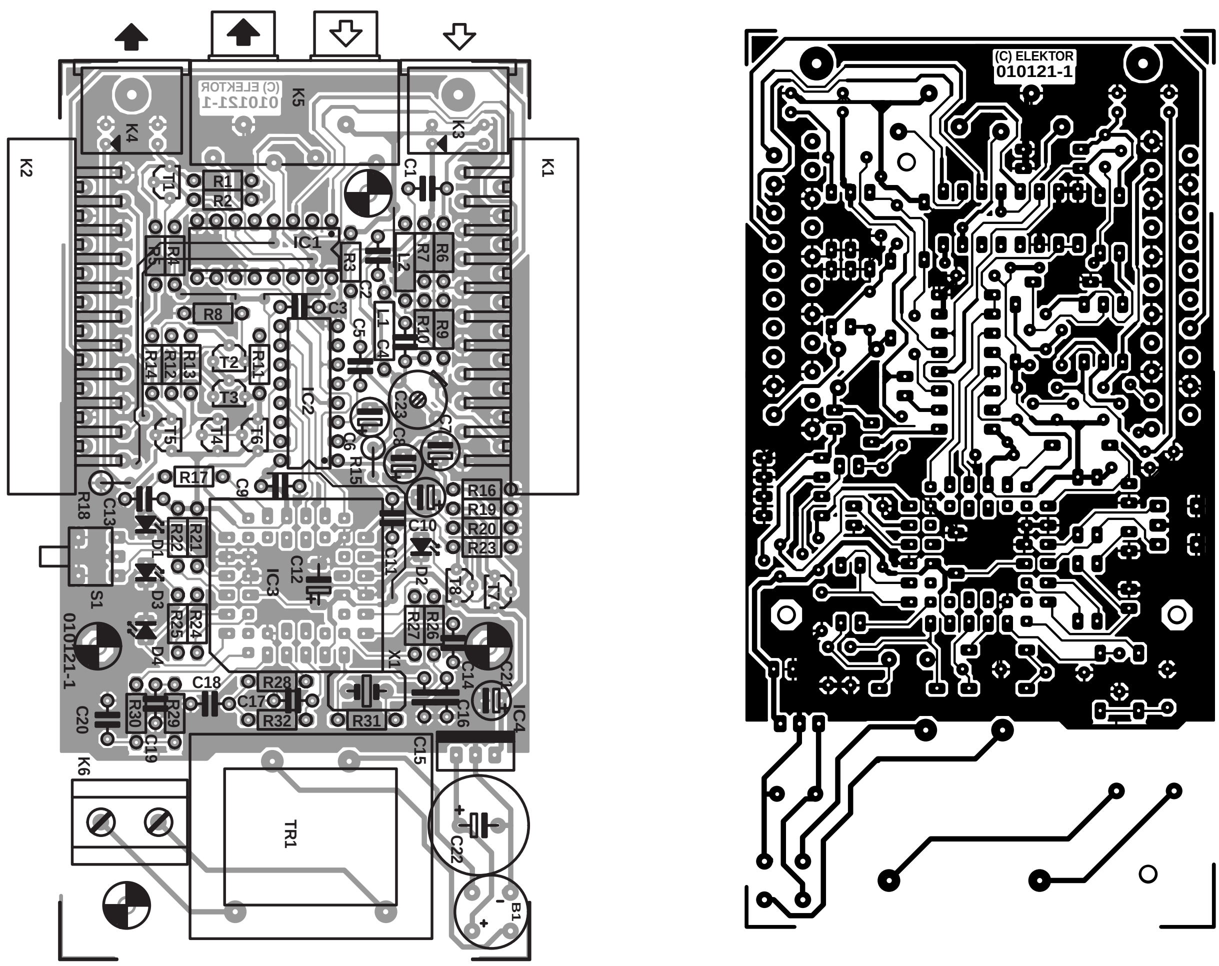


Figure 5. The small and extremely compact board designed for the VCP-2002 (board available ready-made).

the Y path of the CVBS connection, apply a video signal to K1 and check the picture quality on a TV set connected to the Hosiden socket. Watch relatively large areas of equal and steady colour, then adjust C23 for minimum interference. The Y channel should pass as little C information as possible.

(010121-1)

COMPONENTS LIST

Resistors:
R1,R8,R9,R18 = 150Ω
R2,R4,R21,R22 = 10kΩ
R3 = 1kΩ5
R5,R13,R25 = 1kΩ2
R6,R7 = 75Ω
R10 = 270Ω
R11,R14,R26,R28-R30,R32 = 1kΩ
R12,R15 = 220Ω
R16,R20 = 390kΩ
R17,R24 = 100Ω
R19,R23 = 2kΩ2
R27 = 47kΩ
R31 = 100kΩ

Capacitors:
C1,C3,C5,C9,C13,C14 = 100nF (5mm lead pitch)
C2 = 33pF
C4 = 56pF
C6,C7,C8,C21 = 2µF2 25V (radial)
C10 = 6µF8 16V tantalum
C11 = 100pF
C12 = 47µF, 6.3V tantalum (SMD, fit at track side)
C15,C16 = 47pF
C17-C20 = 10nF (5mm lead pitch)
C22 = 1000µF 25V (radial)
C23 = 30pF trimmer (miniature)

Inductors:
L1,L2 = 15µH miniature choke

Semiconductors:
B1 = B80C1000 (round case) (80V piv, 1.2A)
D1-D4 = LED, red, low current, 3mm
IC1,IC2 = 74HC4053
IC3 = EPM7064SLC44-10 (Altera), programmed, order code **010121-31**
IC4 = 4805 (low-drop)
T1,T2,T4,T5 = BC550
T3,T7,T8 = BC560
T6 = BF245**B** (do not use -A or -C suffix)

Miscellaneous:
K1,K2 = SCART socket, PCB mount, angled pins
K3,K4 = 4-way Mini DIN socket, angled pins (for SVHS)
K5-K8 = 4-way cinch (RCA) socket, chassis mount (Conrad Electronics # 736910)
K9 = 2-way PCB terminal block, lead pitch 7.5mm
S1 = toggle switch, 1 change-over, with centre-off position, miniature version for PCB mounting
Tr1 = mains transformer, PCB mount, 1 x 6V, 1.5VA - 1.9VA (Hahn # EI3032030)
X1 = 8.,867238MHz quartz crystal (PAL TV xtal)
ABS enclosure 124 x 71 x 41 mm (l x b x h) (Conrad Electronics # 520993 and 521035)
PCB, order code **010121-11***

PCB layout file available from Free Downloads section at www.elektor-electronics.co.uk

* See Readers Services pages in this issue and/or www.elektor-electronics.co.uk

Chip Tuning

Engine Management Tweaking

By Christian Tomanik ctomanik@gmx.de

Fed up with your car's sluggish performance? Want more power but can't stand the sight of furry dice? Many motorists are turning to the 'chip-tuning' alternative to secretly unleash their car's full potential.

Buy a car these days and you can be sure that the engine will be controlled by an Engine Management System (EMS) containing a microprocessor in an Electronic Control Unit (ECU). The processor reads values of engine speed, inlet airflow, water temperature etc., and decides when to produce the ignition spark. The processor also controls any fuel injectors or turbochargers fitted to the engine. The engines performance is defined in a look-up table stored in an EPROM, so for a certain engine speed at a certain manifold pressure the spark advance figure is given in the look-up table or 'map'. Chip tuning or 're-mapping' involves changing the values stored in this map.

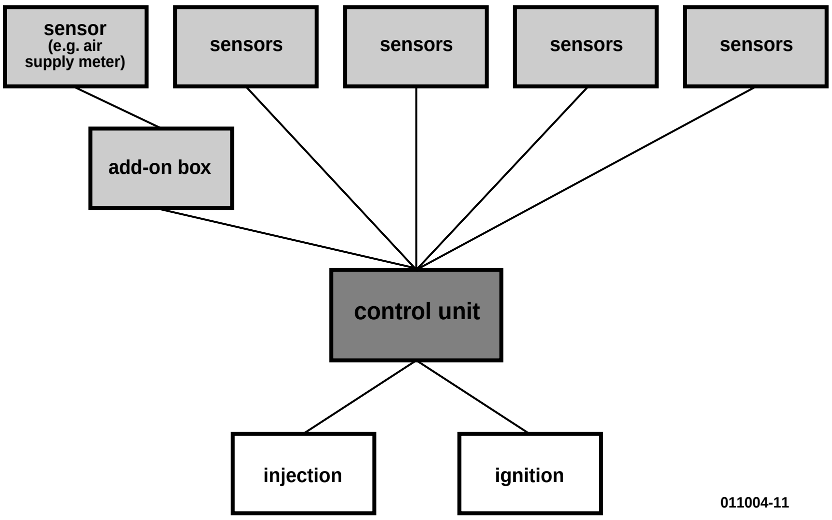


Figure 1. 'Add-on' box between the sensor and the Electronic Control Unit alters the readings from the airflow meter to increase fuel percentage.

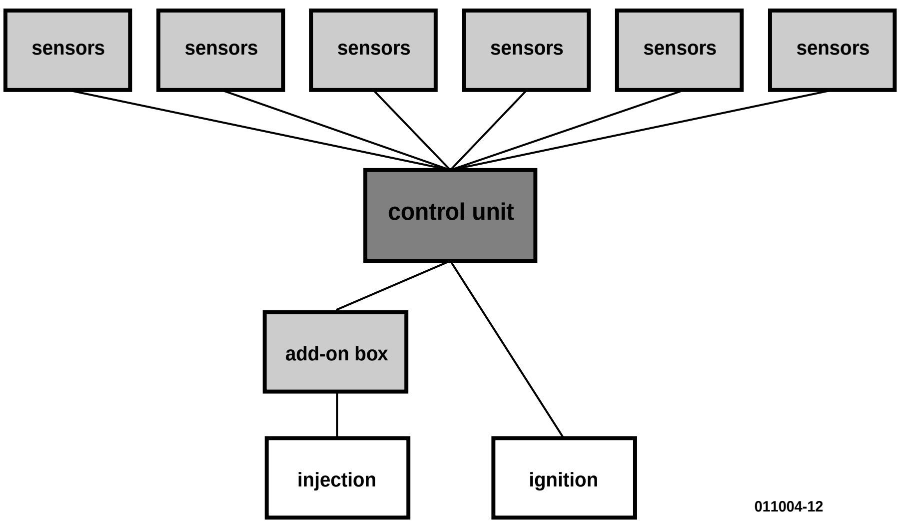


Figure 2. 'Add-on' box between the Electronic Control Unit and the fuel injector increases the injection period.

Most tuning companies suggest that chip-tuning can produce an improvement in power and torque of between 10 and 20 % for a basic engine but the increase can be more than 45 % for a turbocharged diesel engine.

Such staggering increases seem hardly credible — only a few years ago it would have been necessary to perform a top-end rebuild, perhaps slipping in a specialist camshaft and modifying the cylinder head but now the same effect can be achieved in a few hours with a simple software change to the motor's EMS. It also begs the question: why don't the manufacturers supply the engines already modified? In this article we hope to fill-in some of the background and look at the pros and cons of chip-tuning.

Modern manufacturing practices in the car industry mean that each new engine will be put through its paces on a test rig before being fitted to the vehicle. Here it undergoes fine-tuning to ensure that the car's performance conforms to its specification. Tolerances in the manufacturing process ensure that each engine produced will not have exactly the same performance as the next. The final phase of engine set-up will usually involve a certain amount of de-tuning to an average engine. This ensures that fewer engine units are rejected on the grounds of low performance and helps make the production process less wasteful. Chip tuning allows you to tap into this 'power reserve' by changing the engine characteristics.

Some manufacturers take the approach of developing a single high performance engine type which is fitted to their top-of-the-range models and then with a re-programming of the engine management software at the factory the same fire-breathing demon of an engine is turned into a lower revving, smoother and more fuel efficient pussy cat for use in lower-spec cars. These de-tuned engines offer greater potential power and torque gains when 'chipped'. Audi for example currently market at least four 2.7 litre models with a maximum quoted power output ranging from 230 to 380 bhp depending on model.

Turbo boosted engines (especially diesels) allow the boost pressure to be brought into the equation and give the greatest potential power gain. Some kits on the market allow boost pressure to be adjusted from a dashboard mounted control knob. It may sound a bit gimmicky but it will prolong engine life if boost pressure is set to minimum when the engine is cold (the time when most wear occurs) and only turned up once the engine is warmed up, just to give an extra kick when needed for overtaking.

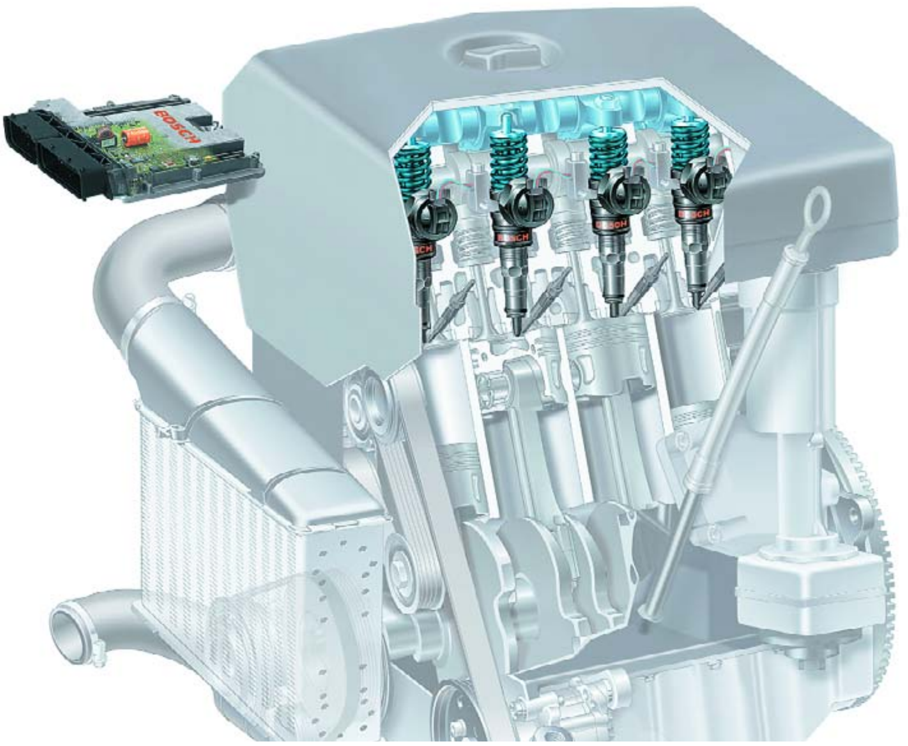


Figure 3. A favourite for chip-tuning is the modern turbo-charged, direct-injection diesel engine. Raising the boost pressure by 0.1 bar will give 10% more power after re-mapping. (Picture: Bosch)

Chip tuning and add-ons

You would expect the vehicle manufacturers to have complete documentation and all the tools necessary to analyse and modify the engine's characteristics, but this information is not generally available and few chip-tuning companies possess the necessary equipment to be able to read and re-map the ECU memory. A simpler form of chip tuning is available to get round this problem. An add-on tuning box is fitted in the wiring between the engine sensors and the ECU input. These boxes effectively have a scaling effect on the sensor values so that if, for example, the add-on box multiplies the values from the air input sensor (**Figure 1**) the processor will be fooled into thinking that more fuel is needed to keep the mixture correct. Alternatively, the box can be fitted between the ECU and the injector to increase the injection period (**Figure 2**). These add-on boxes are much simpler to fit and need little specialist knowledge to set up. In contrast, classical chip-tuning is much more complex and involves reading out the engine characteristics mapped in the EPROM and replacing them with modified values. This type of work is not easy for most mechanical workshops to perform because it involves knowledge of both hardware and software, so if you are considering a chip conversion it will be necessary to track down a specialist garage. Even then, there is no guarantee that the chip tuner knows what he is doing. Some less reputable operators favour the trial and error approach whereby the parameters are altered until a power increase is detected and then the tuning process is considered finished.

Experienced chip-tuners on the other hand use a professional Editor program to decode the ECU memory and also an engine simulation program to optimise and predict the effects of changes to the engine map. The process is labour intensive and requires a high level of skill.

Chip-tuning made easy?

Most people would expect the engine characteristics to be stored somewhere in an integrated controller/EPROM chip somewhere in the EMS of the car. In actual fact, they are almost invariably stored in a conventional EPROM plugged into a socket in the ECU.

The author had the opportunity to experience development of engine management electronics in both a conventional car production facility and also in the more rarefied atmosphere of competitive Motor Sport. This gave a good insight to the techniques of chip-tuning and the working practices of some tuning specialists. What this underlined is the importance of dealing with a reputable company. Very few tuning specialist have the resources to carry out the same sort of testing that vehicle manufacturers put their engines through. In the car industry, the engine control maps are not simply calculated from a computer simulation, they are optimised after an extensive cycle of test-bed development and test-driving on the track. By contrast, many chip tuners do not move far from their desk. Many calculate the offset values and simply re-program the map and test it on a rolling road to check for any change in the power output. The problem here is that it may produce an engine that has good performance only within a narrow range of engine speeds but runs unacceptably under other operating conditions. The missing ingredient is testing. Vehicle manufacturers invest a great deal of time and effort to ensure that the car runs just as well on the Saharan plains as it will on the foothills of the Himalayas. It is just not possible for a small tuning firm with limited resources to carry out the same range of testing for chip-tuned cars. Apart from this, a great deal of investment in personnel is necessary to build up experience and training before a tuner can reach any level of competence.

And on the downside…

In principle it is relatively easy to increase an engine's power output: squirt more fuel/air mixture into the cylinder and you get a bigger bang and hence more power and torque. One of the more obvious side effects of this modification is that fuel consumption will suffer. This form of tuning will not improve the engine efficiency or fuel consumption although many of the chip manufacturers claim better fuel economy for the same driving style.

Bigger bangs will obviously put more mechanical stress on the engine and drive train components and will lead to a reduction in their lifespan. Increasing the fuel/air mixture is also not such a simple procedure because if we keep the same injectors there will be a limit to the maximum fuel flow rate, so to produce a greater fuel volume it is necessary to increase the input cycle injection period. This extended injection period can cause problems in the engine. Especially in a diesel motor under load (turbocharged), excess fuel vapour can condense on the cylinder walls, leading to incomplete combustion and a build-up of soot in the cylinder head (not to mention the clouds of black smoke). This has the effect of increasing engine temperature and can cause the piston head to overheat. Temperatures in excess of 400 °C will cause the protective oil film to break down and produce excessive cylinder and piston wear. It is often necessary to fit a larger radiator or oil cooler to reduce the possibility of overheating.

If we now turn to the paperwork you should be aware that if the car is still under warranty, any tuning or engine modifications will render the warranty void. It will also be necessary to tell your insurance company. Any performance modifications to the car will mean that it no longer complies to its standard specification and you can be sure when it comes to the crunch your policy will be void unless they have been informed.

Add-on boxes

Add-on performance enhancing boxes are not true re-mapping devices because they do not alter the stored engine characteristics; they simply add an offset to the signals from the ECU and sensors but are never the less popular because of their ease of installation.

As we mentioned earlier, there are basically two types of add-on boxes. The original concept was that the box sits between the airflow sensor and the ECU and fools the ECU into injecting more fuel by scaling-up the measured airflow values. More recently manufacturers have been offering longer warranties on their cars and are anxious to prevent any possible damage caused by this type of tuning during the warranty lifetime. Some manufacturers are therefore introducing a reality check in the software of the ECU. This means that all the input sensor values are first checked to ensure they fall within accepted limits and any measurements outside these limits are ignored and a warning indicator lights on the dashboard. A diagnostic failure code will also be stored in the processor memory. The sensor signal is analogue, so the add-on boxes need not contain any form of processor, in fact just a couple of resistors and capacitors are all that is necessary to generate an offset.

The second type of box was developed to get round this restriction. This box is digital and sits between the ECU and injector. It works by intercepting and lengthening the fuel injection signal to the injectors. We have already touched on some of the possible disadvantages of this type of tuning.

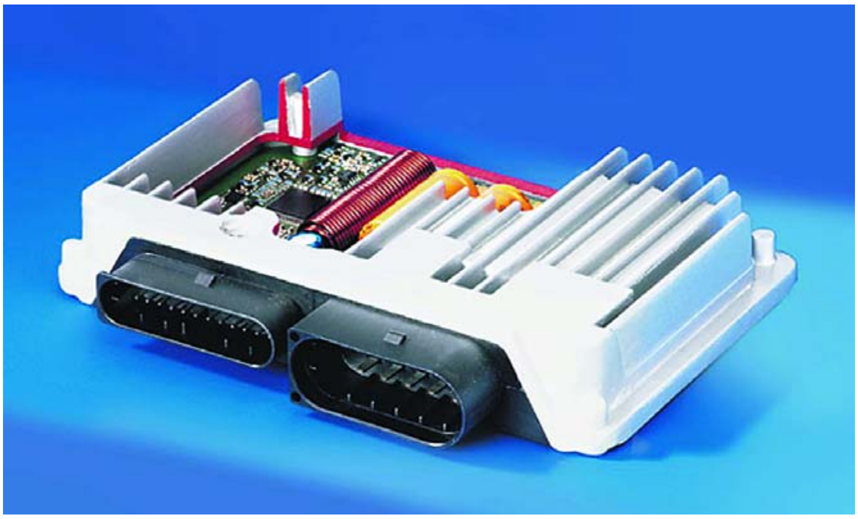


Figure 4. A Siemens variable valve controller fitted to BMW engines (picture: Siemens).

What you can expect

Generally speaking, if you take a standard normally aspirated (non turbo) production motor and fit an 'off-the-shelf' chip-tuning package you can expect a power (and torque) increase in the range of 10 to 15 %. These chip-tuning kits are designed specifically for one type of car/engine combination and can be fitted without any need for further adjustments. To extract more power the car needs to be individually adjusted and fine-tuned using a dynamometer and rolling road set-up where the engine mapping can be optimised. This method is more labour intensive but you can expect up to 35% improvement. Turbo diesels are the favourite and can be tweaked beyond 50%. Further power increases are unlikely without resorting to traditional tuning techniques.

Ex-works Chip tuning?

As we've already mentioned, chip tuning gives a standard vehicle the performance of a sports car but if you take a closer look under the bonnet of a production sports car you will notice that it is not just a different chip that distinguishes the car from the standard model. There will most probably be improvements to other systems in the car like the brakes and suspension, bigger radiator, different turbo and injectors. So buying a performance car from the manufacturer provides an integrated package.

In Germany where chip-tuning is more mainstream than here in the UK, there is one manufacturer of specialist diesel tuning equipment whose products are endorsed by the car company Ford. This means that the full range of products from the company 'WOLF concept' (including chips) can be fitted through the normal Ford dealership. The products are on the whole a little expensive but have the advantage that the vehicle's warranty will not be affected.

Don't try this at home

Any curious Electronics Engineer is probably thinking at this point that it would make an enjoyable evening's exercise to break out the EPROM programmer and re-map the family runabouts engine on the kitchen table but unfortunately it's not really something to be recommended unless you have access to a test bed and a spare engine (or two). If you are considering modifying your car, it is far better to spend your time tracking down a reputable specialist. Check out any guarantee conditions and talk to people who have already made the conversion. Get a quote from an insurance company before you invest in the tuning just to check if you really can afford it. Make sure that the tuning company can guarantee its promised power increase (dynamometer evidence) and check liability for any consequent engine damage. The size of the tuning company is less important than their experience, maybe they have a demo car that you can try out before you make any financial commitment. Ask if they will put you in touch with previous customers who have already made the conversion, a reputable firm will have nothing to hide.

As a final thought, chip-tuning is in its infancy in the UK, the second-hand market will have only a very small percentage of vehicles that have been 'chipped'. The manufacturers of these kits emphasise that a re-mapped engine will not be detected using normal servicing diagnostics and the engine can be easily converted back to standard tune when the time comes to sell the car on. The big question for any potential second hand car buyer in the future will be; would you want to buy a car that has been chipped? If not, how can you be sure that any car has not been chipped?

Internet sites

As ever, the Internet offers useful information and contacts for any potential chip-tuning fan. Van Aaken are a well respected company specialising in mechanical and electronic tuning aids. They also supply the EPROM emulators that allow tuning garages to connect a PC to the vehicles EMS. Based at the Transport Research Laboratory in Berkshire you can find them at *www.vanaaken.com*. Motorsport Developments (*www.motorsport-developments. co.uk*) are a company fitting EVO chips. These re-mapped EPROMS replace the standard chip. They have a dealership across the UK and tuning the engine involves extensive road testing. Tuning specialist BBR fit the Star*chip re-mapped memory and can be found at *www.bbrgti.demon.co.uk*. Superchips (*www.superchips.co.uk*) have been in business for some 15 years and offer re-mapping and a seven-day money back guarantee (less fitting costs). They have an extensive dealership network throughout the UK.

Remember, if you do decide to modify your engine you can also go ahead and fit the In Car Entertainment system and furry dice but neither are obligatory...

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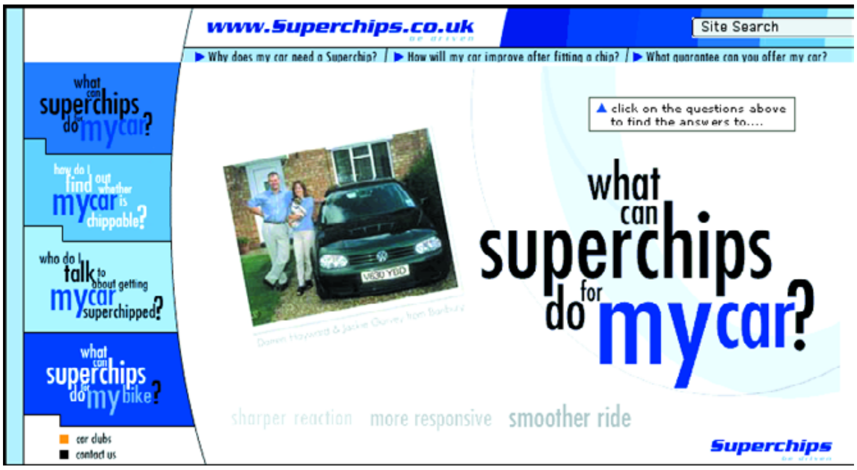
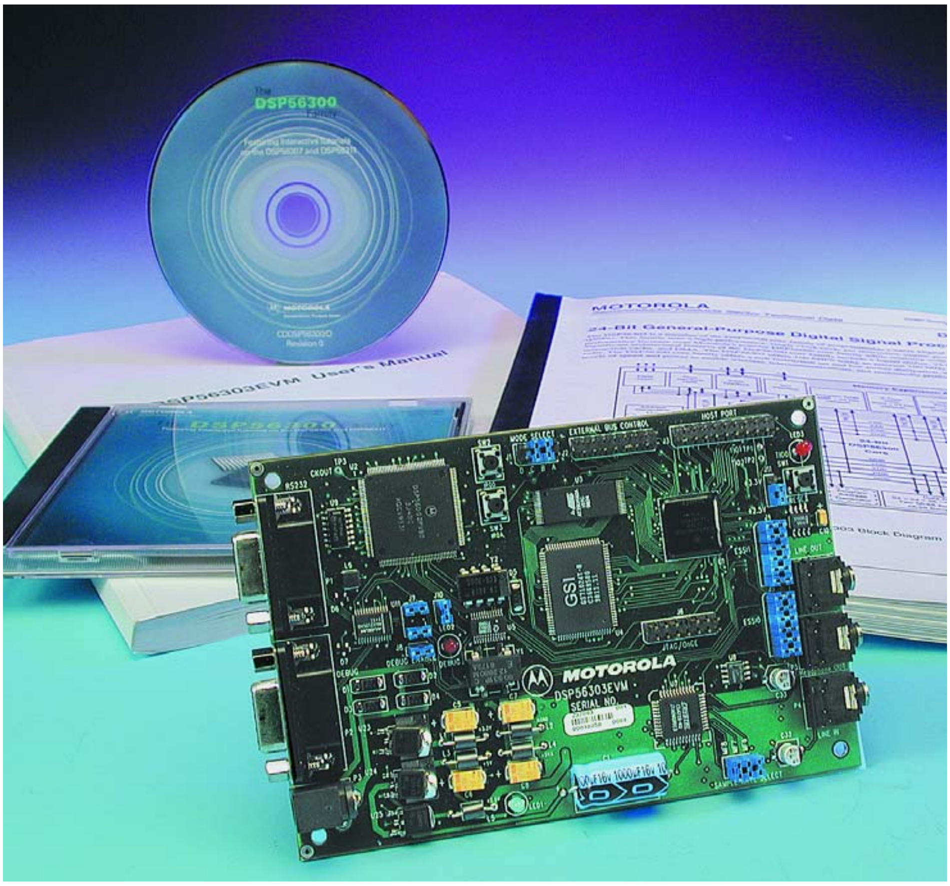


DSP Filter Primer

a practical example of Digital Signal Processing

By G. May, DL3ABQ

Many people are put off using Digital Signal Processors (DSPs) by the thought of grappling with complex mathematical algorithms. In some situations a DSP offers the most cost-effective solution and in practice you don't need to be Einstein to use them successfully. In this article we deal with a practical application and take a look at some of the background theory.



Back in March 1998 we published a series of articles titled "Introduction to digital signal processing" which gave a good insight into the theory behind signal processing and used software routines running on a PC and its sound card to process signals. This article takes a different slant. Here we take a proprietary DSP chip (just a special form of high-speed processor using an instruction set that allows mathematical computations to be performed efficiently) and together with the author's filter design software we build a practical filter circuit.

DSP Systems

The layout of a typical DSP system is shown in **Figure 1.** An analogue input signal (e.g. music) is first passed through a signal conditioning stage and then to a low-pass analogue anti-aliasing filter. Aliasing is a form of distortion that occurs when the input signal to an ADC has frequency components that are higher than half the sampling rate (or 'Nyquist frequency') of the ADC. When these components are sampled they appear in the output signal mirrored about the Nyquist frequency. For example, a pure 9 kHz tone applied to the input of an ADC sampling at 16 kHz (Nyquist frequency = 8 kHz) would result in a 7 kHz tone after sampling. The filter is therefore necessary to attenuate all frequencies above the Nyquist frequency to avoid aliasing effects.

The ADC produces a series of digital samples that represent the analogue input signal. The DSP evaluation kit also allows an external digital signal to be connected directly to the

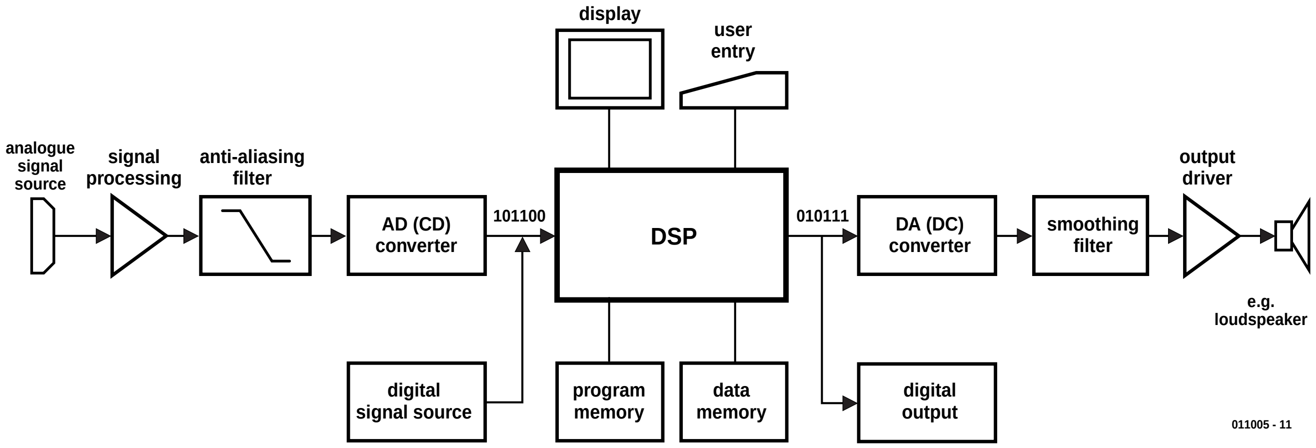


Figure 1. A typical DSP system.

input of the DSP. The DSP can now process this signal according to your software routines, store and display it. The processed digital signal is now output as a raw digital signal where it can be used by another DSP, for example, or converted back to analogue using a Digital to Analogue Converter (DAC). An analogue smoothing filter is next in the chain and this is necessary to smooth out voltage steps in the output signal from the DAC. The signal is now analogue and can be connected to the input of a conventional amplifier stage and loudspeaker.

A chip called a Codec (Coder/Decoder) is designed for use with a DSP and contains most of this peripheral circuitry including the ADC and DAC converters and sometimes the anti-aliasing and smoothing filters.

DSP Starter Kits

Most DSP chip manufacturers offer a starter kit for their product. These kits contain hardware including the DSP and peripheral circuitry mounted on a PCB together with a cable link to a PC. The software packages include an Assembler, Linker and Simulator and a code download program running on a PC. Many DSP application demonstrations are also included. These starter kits simplify the design process greatly and represent good value because their real cost is subsidised by the chip manufacturer in order to encourage wider uptake of their particular DSP. If you plan to incorporate a DSP chip in a project then it's worth remembering that the package outline of some of the more high-performance DSP's is not conducive to hand soldering (e.g. the TQFP outline) or indeed not possible (the BGA outline).

The most popular DSP manufacturers are Texas Instruments, Analog Devices and Motorola. The main features of each of their offerings are given in **Table 1**.

Which one's best?

With the range of starter kits available, you are faced with the decision of which DSP product to choose for your application. Here are some tips to keep in mind when making your choice:

– A Floating-Point type of DSP is much simpler to use. You don't need to worry about scaling, rounding errors or occurrences of underflows or overflows.
– Check that all the necessary software is included in the starter kit; some products

Table 1 Starter kit summary

	TI C3X Starter Kit	TI 6711 Starter Kit	AD 21061 EZLITE	Motorola DSP56303EVM
Manufacturer	Texas Instruments (www.ti.com)	Texas Instruments (www.ti.com)	Analog Devices (www.analog.com)	Motorola (www.motorola.com)
DSP	TMS320C6711	TMS320C31	ADSP-21061	DSP56303
MIPS/MFLOPS	25/50	1200/600	40/150	100/FLOPs (can only be emulated in software)
Rechengenauigkeit	32Bit Floating-Point	32Bit Floating-Point	32Bit Floating-Point	24Bit Fixed-Point
RAM	64kB int. RAM	16MB ext. SDRAM	1MB ext. SRAM	1,5MB ext. SRAM
ROM		128kB ext. Flash	1MB ext. EPROM	0,5MB ext. Flash
Codec	Mono, 20kHz, 14Bit	Stereo, 48kHz, 16Bit	Stereo, 48kHz, 16Bit	Stereo, 48kHz, 16Bit
Interfaces	Analogue, Parallel	Analogue, Parallel, JTAG, Daughter Card	Analogue, Serial, Expansion Connector	Analogue, Seriell, JTAG
Bundled Software	Assembler, Debugger	Assembler, Debugger, C-Compiler (Code size restricted to 256kB)	Assembler, Debugger, C-Compiler	Assembler, Debugger, C-Compiler
Bundled Hardware		Mains unit, Data cable	Mains unit (not included in the European version), data cable	
Price	$99	$295	$179	$249

Note: details subject to change.

require the purchase of additional software.
– Don't put too much faith in the quoted speed in MIPS or MFLOPS (see abbreviations table). Each DSP manufacturer uses a different method to calculate these figures and each will select certain routines that emphasise any efficiencies in their processors instruction set and architecture.

The memory capacity given in Table 1 is specified in bits. It is also important to note the instruction width in bits for each DSP.

Practical applications

One of the most common applications for digital signal processing is signal filtering. All of the starter kit packs include at least one filter implementation example. But before we attempt designing any we will take a brief look at the theory behind digital filters.

Digital Filters

Filters are found in many designs and can be used to extract a signal from a broad band of frequencies or remove unwanted frequency components from a signal. There are many different filter configurations, e.g. a low-pass filter will allow (or pass) all frequencies below a given cut-off frequency while attenuating (or stopping) all frequency components above this frequency. A Band pass filter would be useful in a system employing Frequency Division Multiplexing (FDM) where the receiving equipment needs to extract signal information from one carrier frequency while attenuating all adjacent carrier frequencies.

Figure 2 shows the frequency response characteristic of a low-pass filter. The horizontal axis represents the signal frequency passing through the filter while the vertical axis shows the filter attenuation. 0 dB indicates that the filter does not attenuate the signal at all. An important property of any filter is its cut-off frequency. This is the point at which the filter attenuation causes the output signal to fall to 3 dB less than the input signal. The characteristic shown here indicates that this low-pass filter has a cut-off frequency of approximately 9 KHz.

The design of a typical radio receiver will use many filters; some are narrow-band to pick out one radio station while attenuating all other stations. Other filters (band-stop) will be used to attenuate or suppress a narrow band of frequencies while allowing all other frequencies to pass. These filters are usually built with inductors, capacitors or ceramic resonators but they can also be implemented by a DSP with the filter characteristics programmed into software. In this case the DSP looks at the sampled values of the input signal and works out what the output sample should be. There are two main categories of digital filters:

– FIRs (Finite Impulse Response) Here the impulse response of the filter is convolved with the input signal to produce the output samples. The advantage of this type of filter is that the filter is guaranteed to be stable. This type allows phase-linear filters to be constructed and this characteristic is very important when the filter is used in data communication paths.

– IIRs (Infinite Impulse Response) also known as recursive filters. Each output sample is calculated by weighting the values of the input samples and adding them together. These filters have the advantage that far fewer values must be calculated so that these filters can be built with lower-spec DSP's using less memory. Care must be taken to ensure the filter is stable and designing an IIR filter with a phase-linear response is much more difficult.

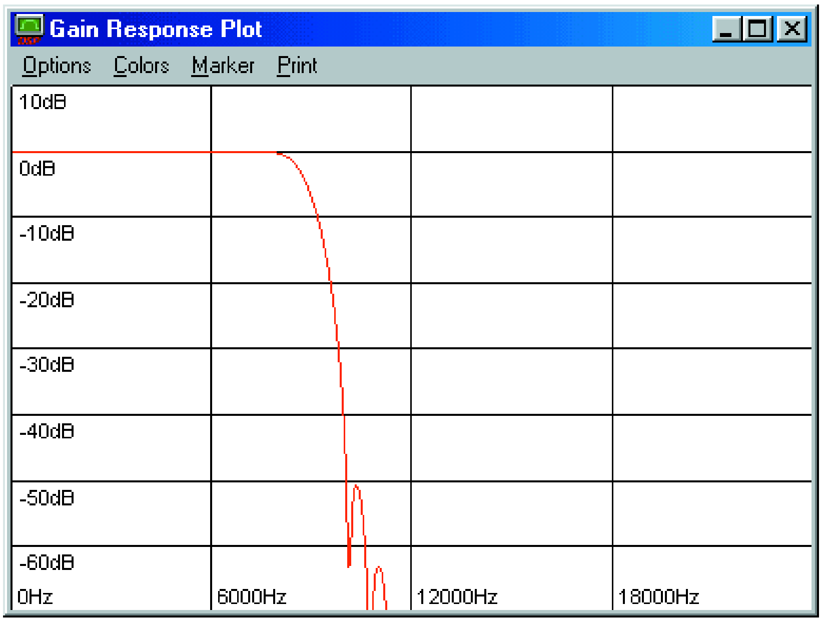


Figure 2. An FIR Filter frequency response.

Building a DSP filter

The steps necessary to build a digital filter can be summarised as:

– Decide on the filter characteristics that are important in the application. Use the filter characteristics in a filter design program to produce the filter coefficients.
– Enter the filter coefficients to produce the DSP filter program code.
– Convert the program to machine code.
– Download the machine code to the target system.

Abbreviations

Codec	Coder/Decoder
DSP	Digital Signal Processor.
FIR	Finite Impulse Response (non-recursive) Filter
IIR	Infinite Impulse Response (recursive) Filter
MFLOPS	Mega Floating Point Operations per Second.
MIPS	Mega Instructions per Second.

The first two steps can be accomplished using a filter design program. The author has produced the "DSP Filter Design" program that runs in Windows on a PC.

The program has a very simple user interface shown in **Figure 3.** To design a filter, the user needs to enter the filter characteristics in the top left window. The program will calculate the corresponding filter coefficients and display them in the top right hand window. The bottom left hand window shows the filter amplitude response while the phase characteristics (The phase shift between the input and output signal) are displayed in the bottom right hand window. The program will also generate the necessary program code directly from the filter coefficients for the majority of the most popular DSPs so that the user needs no special programming skills here.

Literature:

[1] Madiset, Williams (1998):
The Digital Signal Processing Handbook; CRC Press

[2] Taylor, The Athena Group Inc., Mellot:
Hands-on Digital Signal Processing; McGraw-Hill

[3] McClellan, Schafer, Yoder (1998): DSP First; Prentice Hall

[4] El-Sharkawy (1994):
Signal Processing, Image Processing and Graphics Applications with Motorola's DSP96002 Processor; Prentice Hall

[5] DSPs on the Internet, Elektor Electronics 11/1997

[6] Introduction to digital signal processing, Elektor Electronics series started 1/1998

[7] The Internet pages of Texas Instruments, Analog Devices and Motorola

The final two steps of compiling the machine code and downloading it are performed using the software supplied with the starter kit.

A demo version of the DSP Filter Design program can be downloaded from the Free Downloads page on the Elektor Electronics website at

www.elektor-electronics.co.uk.

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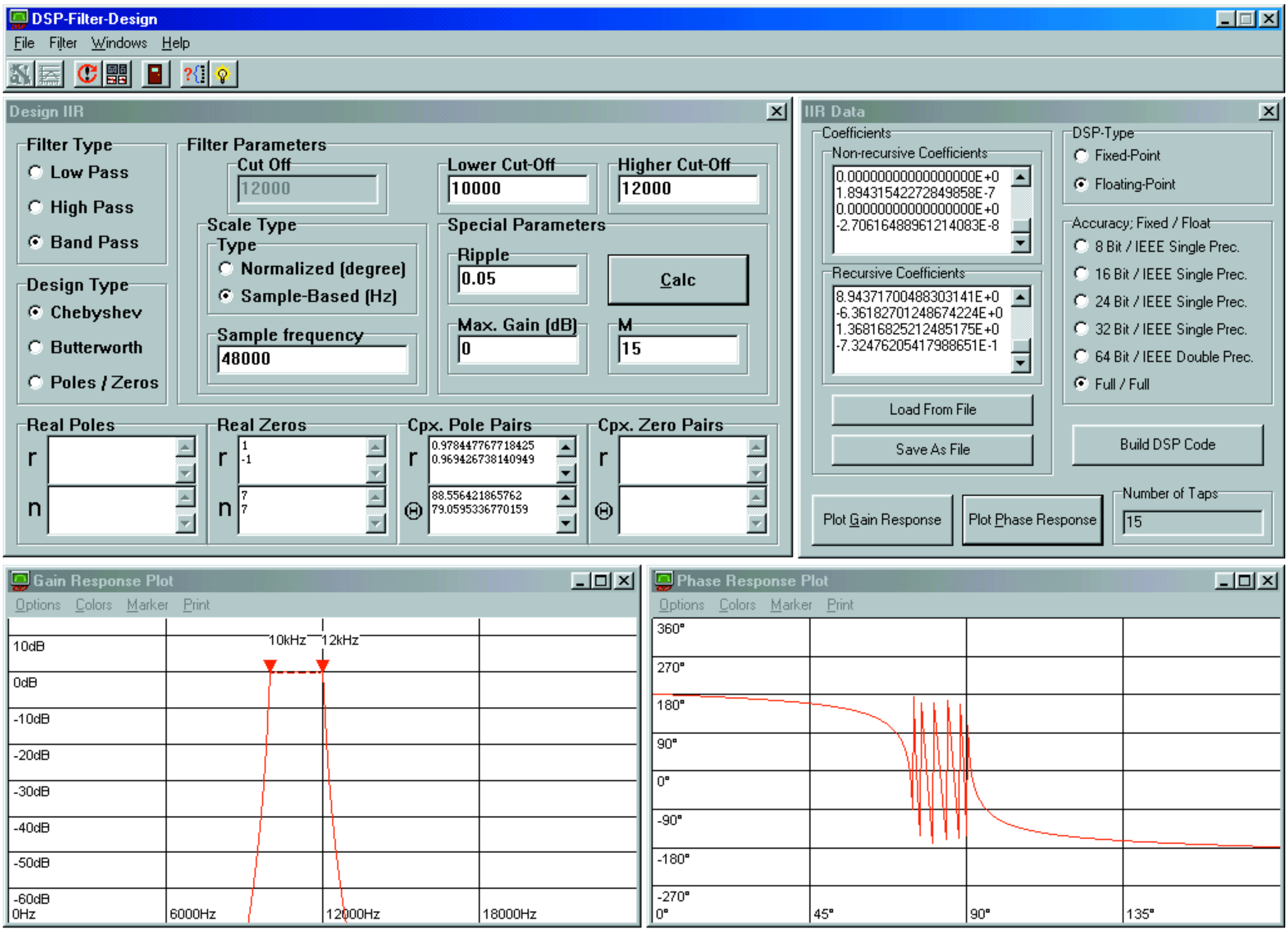


Figure 3. The "DSP Filter Design" program user interface.

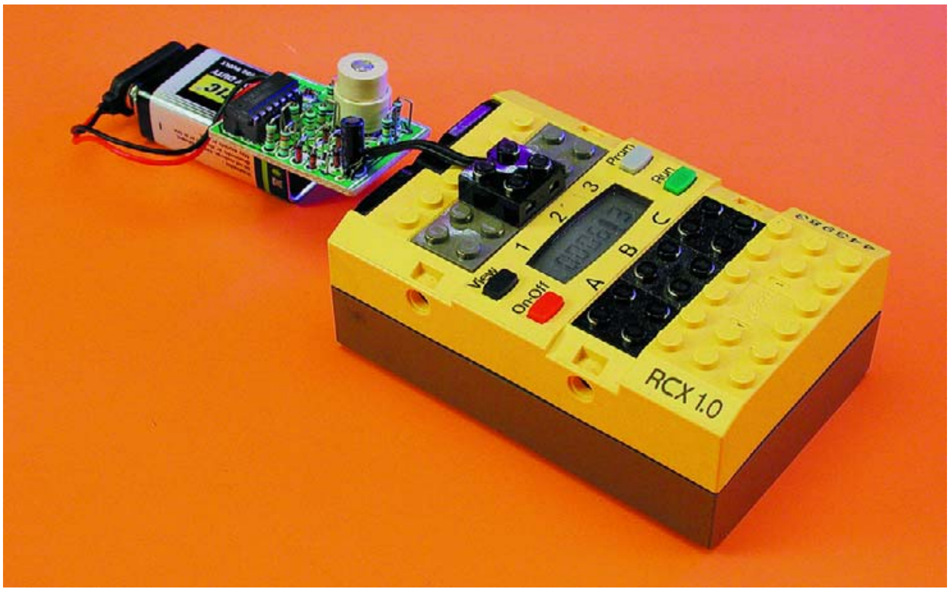
Compass Sensor for Lego RCX

never get lost again

Design by Z. Ottten, write-up by L. Lemmens

Mindstorms, the by now well-known series of robot construction materials from Lego, has already been the subject of several articles in *Elektor Electronics*. In 2000, we presented a series of five articles dealing with all the ins and outs of the RCX module, which is a microcontroller block with three sensor inputs and three outputs for controlling items such as motors and lamps. Following this came a light sensor, a proximity sensor and a sensor multiplexer, and recently a design for an I²C interface appeared in our magazine. This month it's time for a compass sensor, which enables us to give our robots a sense of direction.

If a robot has to travel through a space, this can be implemented in a variety of manners. The first option is to allow the robot to search out its own path. Proximity and contact sensors allow the robot to independently avoid obstacles in order to prevent damage and/or prevent the robot from ending up in a situation that it cannot get out of on its own. The Mindstorms package includes the sensors that we need for this approach.



A second option is to give the robot an objective: to allow it to independently find a way to reach a previously specified location. One example is following a line that defines the course to be taken, such as the one on the mat included in this Lego box.

The route can also be defined in the RCX module by energising the robot's motors for defined intervals and calculating the path that has been travelled during these intervals. The disadvantage of this method is that the speed of the motors depends on the voltage of the RCX battery. In the course of time, the motors will turn more slowly. This causes a gradual decrease in the accuracy of position determination. A compass is an excellent means to at least measure the direction in which the robot is moving, in order to allow the robot to determine its orientation and modify it as necessary. In this article, we describe a

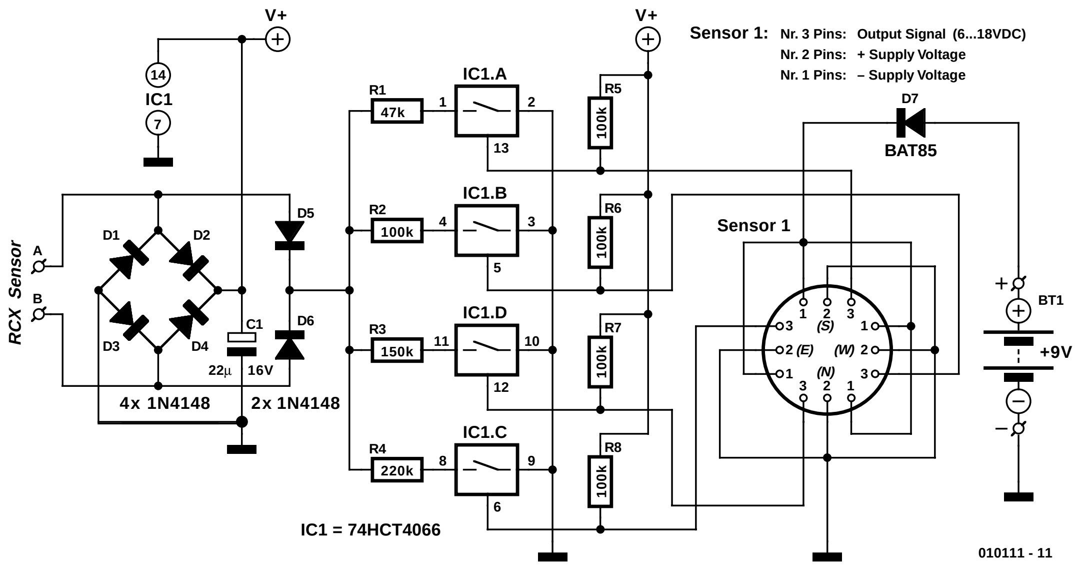


Figure 1. The circuit diagram consists of four analogue switches, a few passive components and a Pewatron type 6945 compass sensor.

simple manner to equip a Mindstorms robot with a compass.

Our design uses a Pewatron type 6945 digital compass sensor. This is not the first time that this sensor has been used in an *Elektor Electronics* project, since in September 1996 we presented an electronic compass that also uses the 6945. The sensor contains a miniature rotor, a magnet and a special Hall-effect IC, which are used together to determine its orientation with respect to the earth's magnetic field. The four digital outputs, of which two can be simultaneously active, provide a resolution of 45 degrees for determining the sensor orientation. It should be noted that the sensor must be aligned perpendicular to the surface of the earth and that external magnetic fields can significantly disturb the measurement. At a price of around €50 (approx. £30), the 6945 is not exactly cheap, so it should be handled as carefully as possible.

The circuit

The circuit diagram of our RCX compass is shown in **Figure 1**. The circuit that provides the interface between the sensor and the RCX module is powered from the RCX module. As you probably know, the RCX can power a sensor and make measurements on the same sensor using only two leads (see **Figure 1**). Most of the time, the supply voltage is present on the A and B terminals. During short intervals, the supply voltage is switched off and a measurement is made. During the measurement interval, the interruption in the supply voltage is bridged by electrolytic capacitor C1. Diode bridge D1–D4 ensures that terminals A and B can be swapped without causing any problems. During the measurement interval, diodes D5 and D6 play a comparable role in determining the resistance of the circuit connected to the RCX input.

The actual sensor, the 6945, draws a fairly large current (20 mA) and is thus powered by the 9-V battery Bt1, since the sensor terminals of the RCX module can supply a maximum current of only 10 mA and would otherwise be overloaded. Diode D7 protects the sensor against a reversed-polarity supply voltage.

The 6945 has four open-collector outputs, of which two are concurrently active (Low), depending on the orientation of the sensor. The pull-up resistors R5–R8 adapt the voltage levels at the sensor outputs to the supply voltage of the interface circuit, which as previously mentioned is powered by the RCX module.

The four outputs of the 6945 drive the four analogue switches of IC1, which in turn cause a certain combination of resistors R1–R4 to be connected in parallel to the RCX sensor input. **Table 1** shows the values measured by the RCX module, depending on the orientation of the sensor. Here we see eight compass directions as a function of the state of the compass sensor. Note that only a small portion of the measurement range of the RCX is used, but this is adequate for determining the eight states of the sensor with sufficient accuracy (the RCX can determine a measurement value between 0 and 1023 at its input).

It would have probably been possible to connect resistors R1–R4 directly to the outputs of the sensor, which would allow IC1 and R5–R8 to be eliminated. However, considering the price of the sensor and the fact that the sensor data sheet is very skimpy and gives hardly any information about the load capacity of the outputs, we decided we would rather be safe than sorry.

Construction

If you want to have a truly compact sensor, you best bet is to build the circuit on a piece

Table 1.
Recommended values for resistors R1–R4

Direction	sensor pin 3	6	9	12	RCX value
N	1	1	1	0	863
NO	0	1	1	0	920
O	0	1	1	1	810
ZO	0	0	1	1	856
Z	1	0	1	1	818
ZW	1	0	0	1	830
W	1	1	0	1	787
NW	1	1	0	0	887

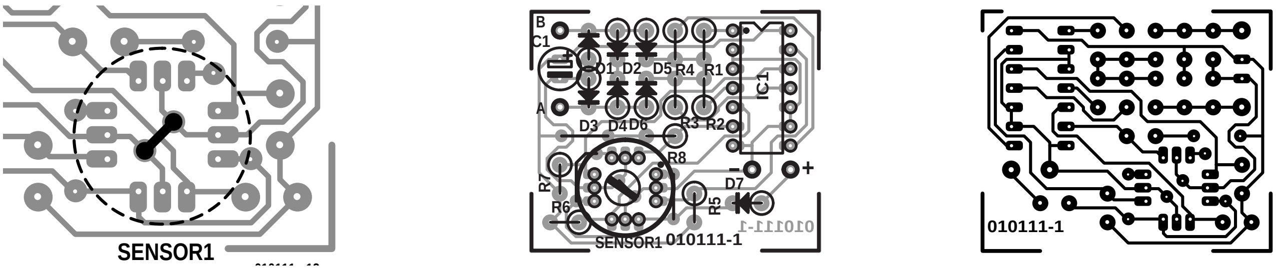


Figure 2. In order to keep the circuit board small, the resistors and diodes are fitted vertically. Note the wire bridges next to the sensor! (board not available ready-made).

of perforated prototyping board in order to keep the dimensions of the circuit as small as possible. However, if you would like a bit more convenience and are not all that worried about size, you can use the circuit board layout shown in **Figure 2**. Pay attention to the wire bridges next to the sensor, especially the short wire bridge under the middle of the sensor. The best approach is to use insulated wire for these bridges and fit them to the copper side of the board, since this allows the sensor to be fitted flush to the surface of the board.

For connecting the sensor to the RCX module, we use a standard Lego cable with terminal bricks, which we cut in two. The battery should be mounted as far away from the sensor as possible, since otherwise the metal case of the battery will affect the reading.

Listing 1. Sample program in Visual Basic.

```
Private Sub Command1_Click()
Const setpoint = 5
Const measure = 6
Const dif = 7

With RCX
.ComPortNo = 1  ' COM port 1
.LinkType = 0   ' 0 = IR
.PBrick = 1  ' 1 = RCX, 0 = CyberMaster
.InitComm          ' Initialise the COM port

.SelectPrgm SLOT3
.BeginOfTask MAIN
    .SetSensorType SENSOR_2, LIGHT_TYPE
    .SetSensorMode SENSOR_2, RAW_MODE, 0

    .SetVar setpoint, CON, 1 '1=N,2=NE,3=E,4=SE,5=S,6=SW,7=W,8=NW
    .SetVar measure, CON, 0
    .SetVar dif, CON, 0

    .StartTask 1
.EndOfTask 'Main

.BeginOfTask 1
.Loop CON, forever
'North
    .If SENVAL, SENSOR_2, EQ, CON, 863
    .SetVar measure, CON, 1
    .EndIf
    .If SENVAL, SENSOR_2, EQ, CON, 862
    .SetVar measure, CON, 1
    .EndIf
    .If SENVAL, SENSOR_2, EQ, CON, 861
    .SetVar measure, CON, 1
    .EndIf
'North-East
    .If SENVAL, SENSOR_2, EQ, CON, 920
    .SetVar measure, CON, 2
    .EndIf
    .If SENVAL, SENSOR_2, EQ, CON, 919
    .SetVar measure, CON, 2
    .EndIf
    .If SENVAL, SENSOR_2, EQ, CON, 918
    .SetVar measure, CON, 2
    .EndIf
```

Software

The sample program (see **Listing 1**) for using our compass sensor is written in Visual Basic. When the software provided on the Mindstorms CD is installed, the SPIRIT.OCX software library is also installed. This library allows the RCX to be programmed and controlled using this high-level language. The listing can also be used with other languages, such as Delphi or C++, without major modifications.

Starting with a blank form in Visual Basic, we place a Spirit control and a Command button on it. A double click on this button opens a window into which we can copy Listing 1. When we execute the Visual Basic program, the application is sent to the RCX block via the Tower.

The sample program is reasonably simple. In the program, a direction is specified in the line *SETVAR setpoint, CON, 1*. The latter number can be assigned a value from 1 to 8 in order to set the desired direction. Following this, the value of the sensor is read using an infinite loop. If the orientation matches the previously set direction, a tone is emitted

COMPONENTS LIST

Resistors:
R1 = 47kΩ
R2,R5-R8 = 100kΩ
R3 = 150kΩ
R4 = 220kΩ

Capacitors:
C1 = 22µF 16V radial

Semiconductors:
D1-D6 = 1N4148
D7 = BAT85
IC1 = 74HCT4066

Miscellaneous:
Sensor1 = Pewatron 6945 (Pewatron AG, www.pewatron.com)
BT1 = 9V PP3 battery with clip-on lead
Lego connection cable (cut in two):
length 26.6cm, Lego order code 5311
length 9cm, Lego order code 5041

by the RCX buzzer.

The four lines following the statement *With RCX* are related to the communications between the PC and the RCX module. They have no direct significance for the actual application. Following this, program slot 3 in the RCX module is selected. After this, sensor 2 is configured in the task *Main* so we can work with the compass sensor, the desired direction is selected (*setpoint*) and the registers for storing the measured value (*measure*) and the difference between the measured and target values (*dif*) are set to zero. The final statement in *Main* starts task 1, which contains the infinite loop.

When a measurement is made, three values are checked for each direction in order to compensate for any variations that may be present. If the measured value actually falls within the range of a particular compass direction, the *measure* register receives a value equal to the sequence number of this direction.

After these 24 comparisons with the measured value have been made, a check is made to see whether the measured direction matches the set direction. If this is the case (*dif=0*), routine *Playsystemsound* is executed.

(010111-1)

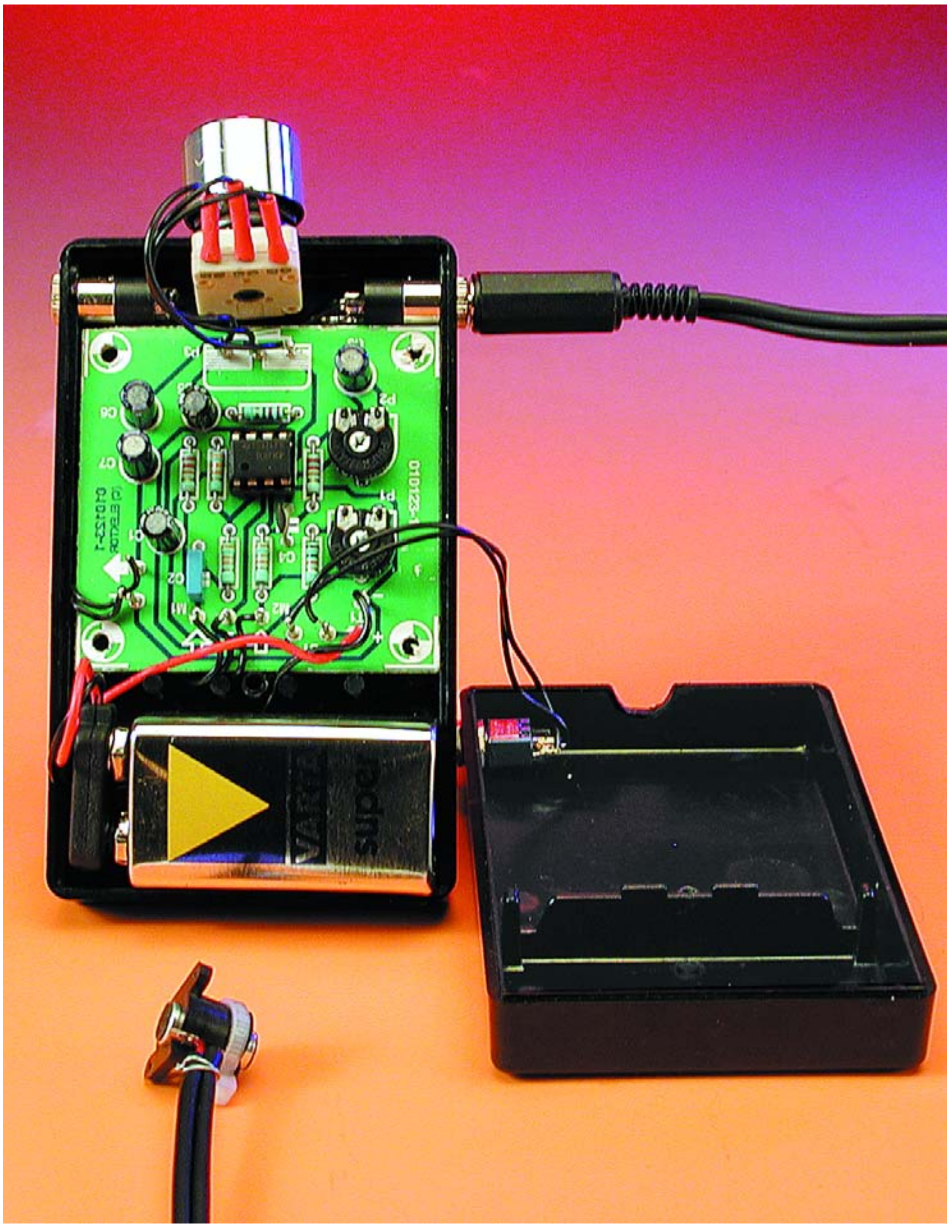
```
'East
    .If SENVAL, SENSOR_2, EQ, CON, 810
    .SetVar measure, CON, 3
    .EndIf
    .If SENVAL, SENSOR_2, EQ, CON, 809
    .SetVar measure, CON, 3
    .EndIf
    .If SENVAL, SENSOR_2, EQ, CON, 808
    .SetVar measure, CON, 3
    .EndIf
'South-East
    .If SENVAL, SENSOR_2, EQ, CON, 856
    .SetVar measure, CON, 4
    .EndIf
    .If SENVAL, SENSOR_2, EQ, CON, 855
    .SetVar measure, CON, 4
    .EndIf
    .If SENVAL, SENSOR_2, EQ, CON, 854
    .SetVar measure, CON, 4
    .EndIf
'South
    .If SENVAL, SENSOR_2, EQ, CON, 818
    .SetVar measure, CON, 5
    .EndIf
    .If SENVAL, SENSOR_2, EQ, CON, 817
    .SetVar measure, CON, 5
    .EndIf
    .If SENVAL, SENSOR_2, EQ, CON, 816
    .SetVar measure, CON, 5
    .EndIf
'South-West
    .If SENVAL, SENSOR_2, EQ, CON, 830
    .SetVar measure, CON, 6
    .EndIf
    .If SENVAL, SENSOR_2, EQ, CON, 829
    .SetVar measure, CON, 6
    .EndIf
    .If SENVAL, SENSOR_2, EQ, CON, 828
    .SetVar measure, CON, 6
    .EndIf
'West
    .If SENVAL, SENSOR_2, EQ, CON, 787
    .SetVar measure, CON, 7
    .EndIf
    .If SENVAL, SENSOR_2, EQ, CON, 786
    .SetVar measure, CON, 7
    .EndIf
    .If SENVAL, SENSOR_2, EQ, CON, 785
    .SetVar measure, CON, 7
    .EndIf
'North-West
    .If SENVAL, SENSOR_2, EQ, CON, 887
    .SetVar measure, CON, 8
    .EndIf
    .If SENVAL, SENSOR_2, EQ, CON, 886
    .SetVar measure, CON, 8
    .EndIf
    .If SENVAL, SENSOR_2, EQ, CON, 885
    .SetVar measure, CON, 8
    .EndIf

'The sensor value is stored in register 'measure'
'the target value in 'setpoint'
.SetVar dif, VAR, measure 'dif=measure-setpoint
.SubVar dif, VAR, setpoint
.If VAR, dif, EQ, CON, 0
    .PlaySystemSound 0
.EndIf
.EndLoop
.EndOfTask
End With

End Sub
```

Directional Microphone

for 'softer' instruments

design by K. Rohwer

Compared to the other instruments in a band the mouth organ is often not loud enough. Turn up the amplifier, and you risk feedback. What is needed is a microphone with a highly directional response.



Microphones that are equally sensitive in all directions have a spherical, or omnidirectional, response pattern. Such microphones respond to the changes in pressure that represent a sound. Pressure is a directionless quantity, and so a microphone with an omnidirectional response pattern can also be used as a pressure transducer (**Figure 1a**).

As an alternative to this there is the pressure gradient transducer. Pressure gradient, the derivative of pressure with respect to distance, is a directed quantity: it reaches a maximum in the direction of the sound source, is zero perpendicular to that direction, and in the opposite direction is again at a maximum, but with opposite sign. A microphone which is a pure pressure gradient transducer has a response that depends on the angle to the sound source that resembles a figure-of-eight (**Figure 1b**).

Most microphones have a characteristic that lies between these two extremes. By mixing the omnidirectional response and the figure-of-eight response in various proportions, a response maximum in one particular direction can be achieved, the so-called cardioid characteristic (**Figure 1c**). If a small maximum in the direction opposite to the sound source is acceptable, the maximum in the direction of the sound source can be made more pronounced, resulting in a so-called supercardioid pattern (**Figure 1d**).

Two spheres =one supercardioid

How do we build a pressure gradient transducer? Take two pressure transducers, arrange them a (small) distance apart — or else it won't work — and form the difference between the two output signals. Any microphone capsule will do for a pressure transducer, as long as the sound cannot reach the back of the membrane: for example, an electret microphone capsule will work.

The distance between the two capsules must not be too great. If half the wavelength of the sound is equal to the distance between the two membranes, the two pressure signals will be exactly 180° out of phase. At the differential amplifier the signals cancel one another out rather than reinforcing one another. At higher frequencies, therefore, the microphone becomes less directional.

The author has developed a microphone along these lines, with a suitable preamplifier that allows the signals from the two capsules to be mixed in variable proportion: the directional characteristic is adjustable.

Figure 2 shows the construction of the microphone. The microphone capsules are fitted into a rubber tube (which came with the capsules), cut lengthways with a craft knife. A small plastic disc, cut out using an office hole punch, serves to isolate the electrical connections from one another. The front surfaces of the microphone capsules are at a distance of about 15 mm from one another, which gives a cutoff frequency of 11 kHz for the directional characteristic. Above 11 kHz the microphone is also sensitive to feedback, and so it makes sense to turn down the treble control on the amplifier.

The arrangement can be built into a miniature enclosure as shown in **Figure 3**. The enclosure has been specially modified: both sides have been perforated to allow the sound to get to the two microphone capsules easily. The cable entry on the left-hand side is fixed to the enclosure via a length of aluminium angle extrusion, and the microphone capsule unit therefore sits freely between two layers of foam rubber. A third thin strip of foam rubber is fitted in front of the microphone to protect it in the direction of its maximum sensitivity.

The author uses his microphone with a

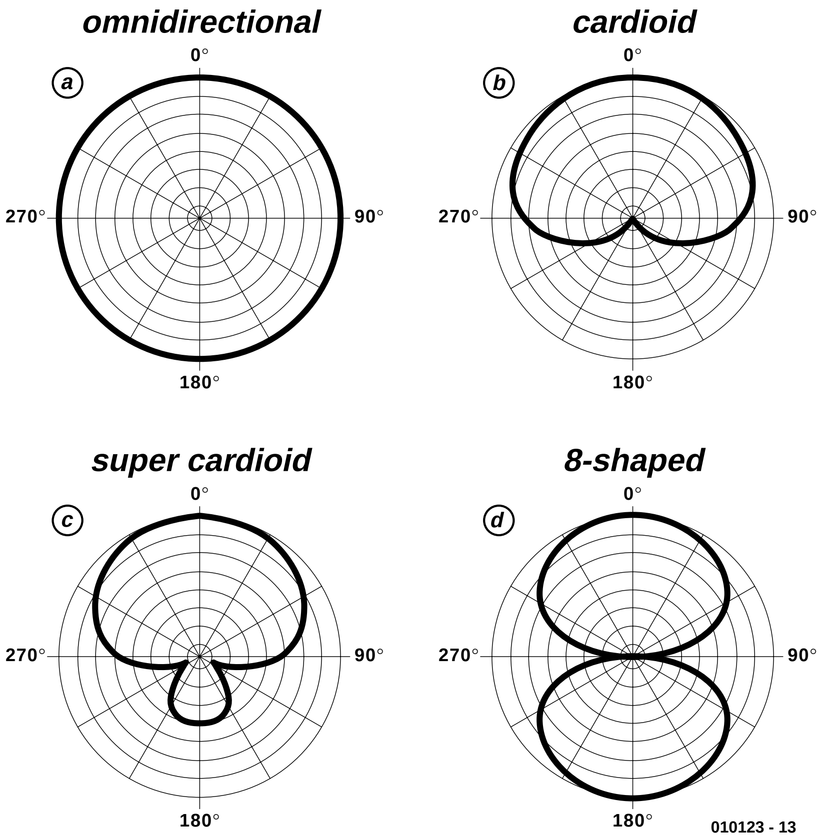


Figure 1. From omnidirectional to supercardioid: a range of microphone characteristics.

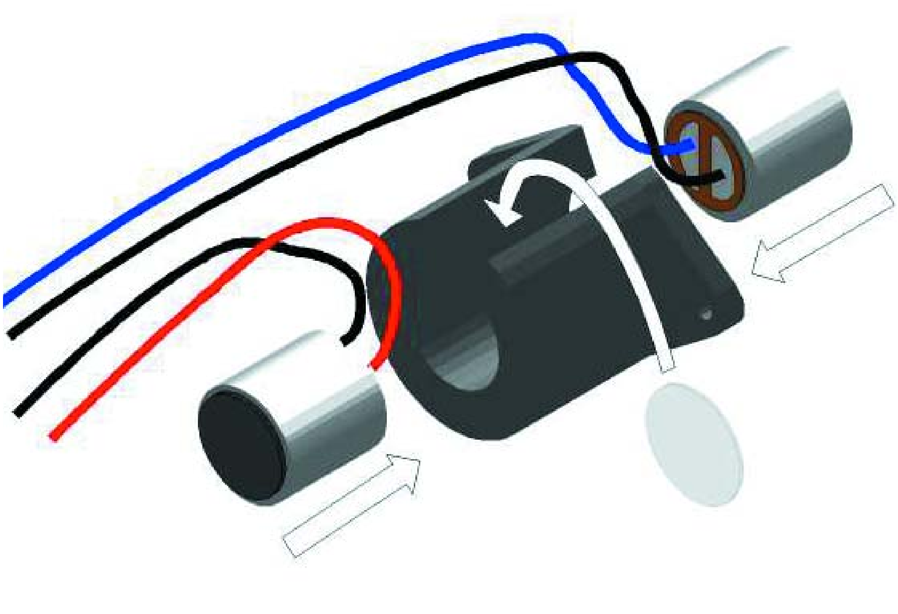


Figure 2. The two microphone capsules, fitted opposite one another in a rubber tube.

Figure 3. The author's microphone enclosure with freely-hanging microphone.

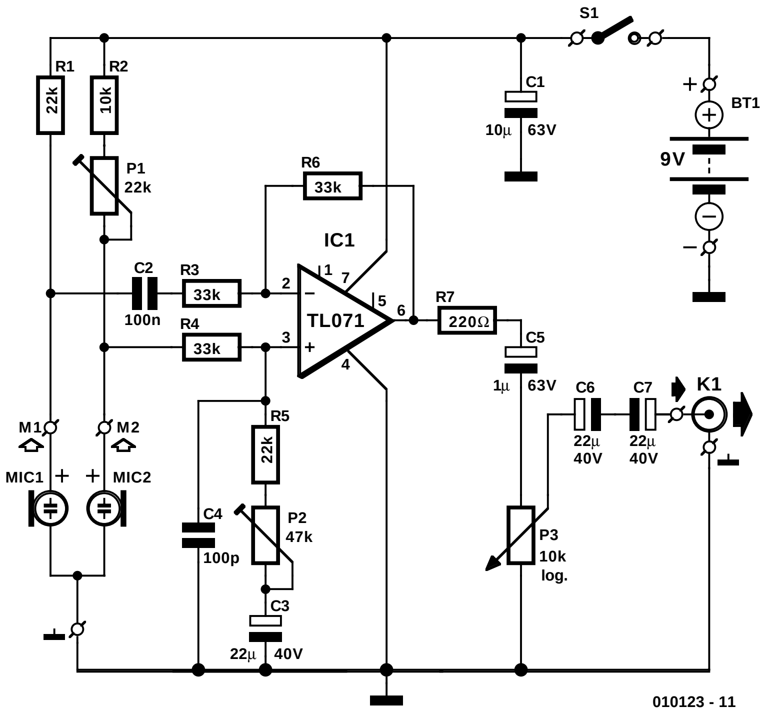


Figure 4. Circuit of the differential microphone amplifier.

COMPONENTS LIST

Resistors:
R1,R5 = 22kΩ
R2 = 10kΩ
R3,R4,R6 = 33kΩ
R7 = 220Ω
P1 = 22kΩ preset
P2 = 47kΩ preset
P3 = 10kΩ potentiometer, logarithmic law, miniature mono type

Capacitors:
C1 = 10µF 63V radial
C2 = 100nF
C3,C6,C7 = 22µF 40V radial
C4 = 100pF
C5 = 1µF 63V radial

Semiconductors:
IC1 = TL071CP*

Miscellaneous:
Bt1 = 9V PP3 battery with clip-on connector
MIC1,MIC2 = miniature condenser (electret) microphone capsule with rubber holder (e.g. Monacor/Monarch type MCE2000)
3.5mm stereo jack socket or 5-way DIN-socket, chassis mount*
K1 = 6.3mm mono jack socket with integral isolated switch, for chassis mounting
Enclosure with battery compartment (and belt clip), size approx. 102 × 61 × 26 mm

PCB layout available from Free Downloads section at www.elektor-electronics.co.uk)

* see text

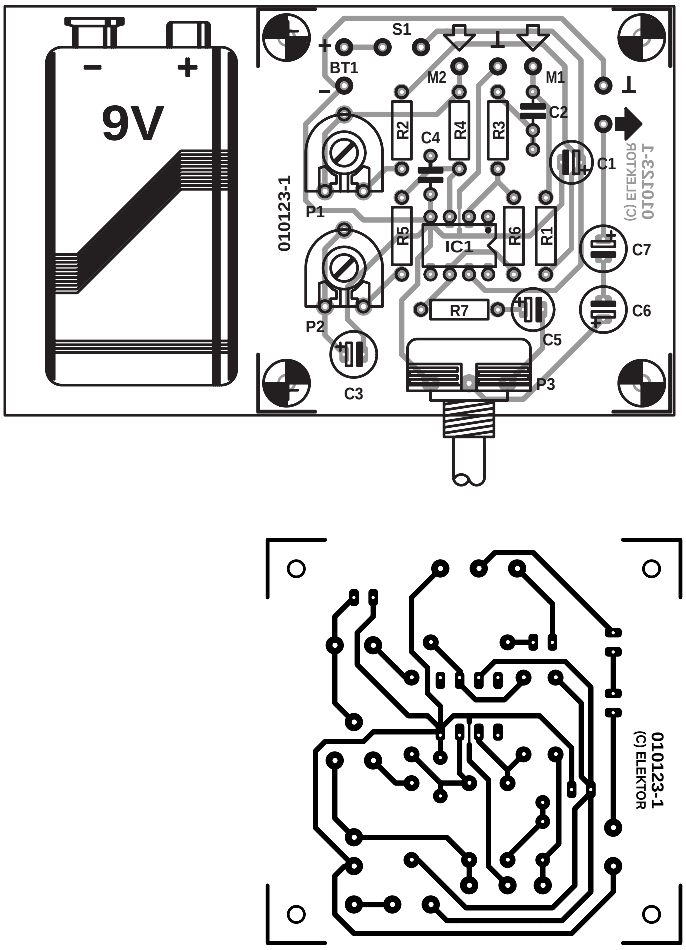


Figure 5. The circuit board is fitted into an enclosure with belt clip and battery compartment.

mouth organ and has therefore made a couple of special modifications. The two countersunk screws in the aluminium angle section hold a Velcro band to the outside of the case, making it easier to hold the microphone comfortably with two fingers. It also became clear that some of the holes had to be closed up in order for the microphone not to pick up the sound of the player breathing through the nose. A foam headphone pad was therefore fitted to the front of the enclosure to dampen the sound of the air flowing over the sharp edges of the holes.

The microphone capsules are connected, observing polarity, to the central cores of a two-core screened microphone cable, the other two connections both being connected to the screen. A 3.5 mm stereo jack is fitted to the other end of the cable.

For improved ruggedness, for use in a studio for example, 3.5 mm jack plugs are not good enough. It is better to use a five-pin DIN connector, wiring pairs of contacts in parallel (ground being taken both to pin 2 and to the screen of the plug).

The microphone preamplifier

The circuit of the differential preamplifier is shown in **Figure 4**. Electret microphone capsules have an internal FET impedance converter and behave as a current source of about 250 µA. This value can vary wildly, even between examples of the same type. The current is modulated by the sound signal, and again the sensitivity can vary widely from device to device. The microphones' series resistors, R1 and the R2/P1 combination, are chosen so that about half the supply voltage is dropped across them. This may not be exactly the case for MIC1, and so C2 is provided to remove the DC component from the input to the amplifier circuit. The series resistance for MIC2, on the other hand, sets the DC offset of the whole circuit, and is adjusted using P1 so that the output voltage of the op-amp is equal to half the supply voltage. The signals from the two microphones are taken to the op-amp, which is connected as a differential amplifier. The author used a type TL071 here, but we also recommend pin-compatible alternatives such as the OPA181GP, TS921IN or OPA350PA rail-to-rail op-amps in order to squeeze the last milliamp-hour of capacity out of the battery.

The signal from MIC2 can be attenuated or boosted using P2 in order to compensate for variation between the microphone capsules. This also affects the directionality of the unit. If R5 plus P2 could be made a short circuit, MIC2 would have no effect and just the omnidirectional characteristic of MIC1 would remain. On the other hand, if MIC1 and MIC2 were identical, and R5+P2 were made equal to 33 kΩ, the figure-of-eight characteristic would result. In between we can obtain the cardioid and supercardioid characteristics. C4 prevents the circuit from going into oscillation.

At the output we have volume control P3 followed by (imitation) bipolar electrolytic C6, which isolates the circuit from any phantom power supply that might be present. A 6.3 mm jack socket, as commonly used in PA applications, serves as the output connector. The jack should have an integrated, isolated switch which can be used to activate the unit only when the jack is plugged in. This saves a power switch, which is all to easy to forget — leaving the battery flat when the unit is next used.

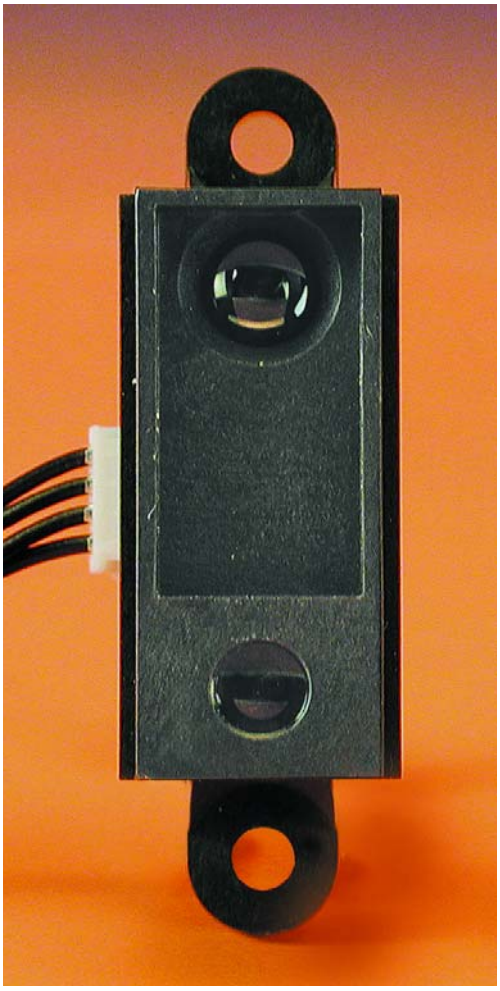
The whole unit is readily assembled on the circuit board, whose layout is shown in **Figure 5**, and easily fitted into a suitable enclosure with battery compartment. P2 should be fitted with a spindle which can pass through a suitably drilled hole in the enclosure. This allows easy adjustment of the directional characteristic. Take care to ensure that the construction is sturdy and that the sockets are wired tidily: you do not want to suffer a loose connection in the middle of a performance!

(010123-1)

Distance Measurement using Infrared

with a new proximity sensor module

In the form of the GP2D02, Sharp have produced a sensor which, with the aid of a microcontroller, can provide reasonably accurate measurements of distances in the range 10 to 80 cm.



Originally developed for use as a proximity switch in sanitary ware, this sensor is also ideal for detecting objects in robotics applications or as a parking aid. The distance is reported as an 8 bit value output serially on a single pin, which can be subsequently processed in software. The sensor includes an infrared transmitter diode along with a suitable receiver and processing electronics. The component is so designed that ambient light and the colour and reflective properties of the object being detected have practically no effect on the measurement results.

Operation of the GP2D02

The heart of the sensor, whose internal circuit is shown in **Figure 1**, is a PSD (position sensitive device), set behind an optical lens. The measurement method uses the triangulation principle. Using precision optics, the transmitter is made to emit a highly focussed beam of light which is reflected from the target object. The angle of the reflected beam varies with the distance between sensor and target object (**Figure 2**).

At the receiver the light passes through a further lens, which focuses the beam to a point on a photosensitive device. As the distance between target object and sensor changes, so the focussed point of light moves along the photodetector. The position-dependent output signal from the photodetector is amplified, digitised, processed and then sent out over an interface, whereupon it can be processed further.

Reading the distance information

The sensor is read by taking the input signal V_{in} low for at least 70 ms. During this period the infrared transmitter diode emits 16 bursts. The internal logic calculates the average of the 16 digital distance measurements taken, thereby reducing the effect of measurement errors.

V_{in} is then clocked eight times and on each falling edge one bit of the digital value representing the distance to the target object can be read from pin V_{out} by the microcontroller. The sensor must always be supplied with an external clock, and therefore cannot work without some external 'intelligence'. **Figure 3** shows the timing diagram for operation of the device. The relationship between the actual distance and the digital value is not linear. The resolution is around 1 cm, and falls to around 10 cm with increasing distance. The non-linearity is a consequence of the optical method used and of the relatively small distance between the transmitter and the receiver in the sensor enclosure. **Figure 4** shows an example of the relationship between the measured value (shown converted into decimal) and the actual distance. The

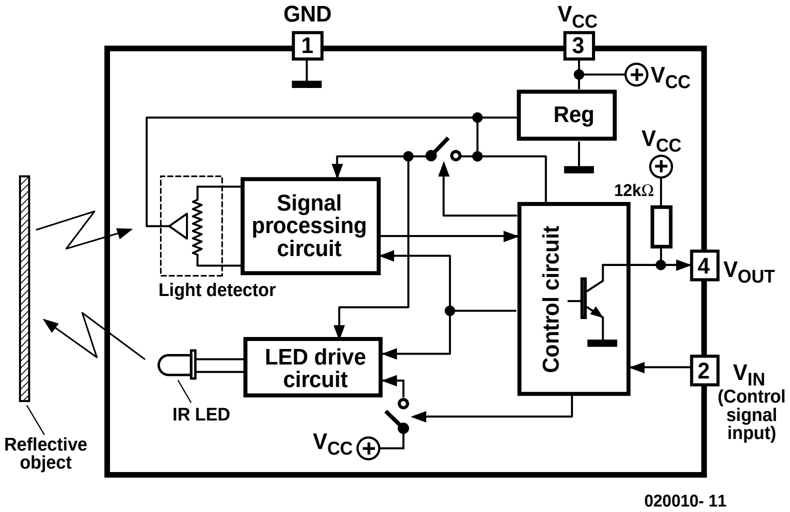


Figure 1. Internal circuit block diagram of the Sharp GP2D02.

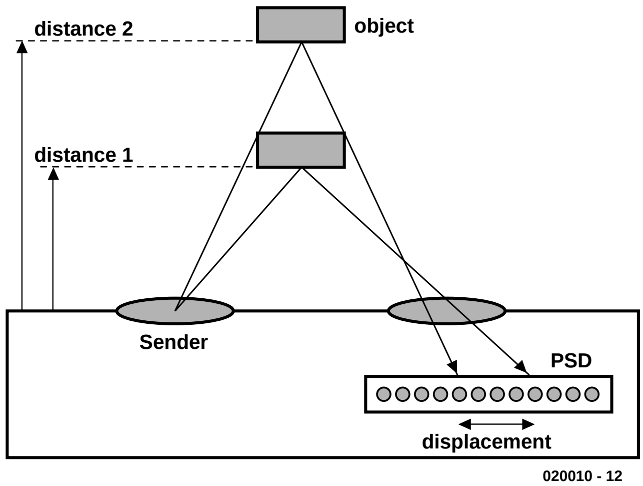


Figure 2. Measuring distance by triangulation.

interesting parts of the curve are the regions below about 10 cm and above around 80 cm. In the former case the output value starts to rise with decreasing distance, which can confuse subsequent processing circuitry. Above about 80 cm the curve becomes so flat that it is practically impossible to obtain a distance measurement. It is important to note that this curve only holds good for an object with particular reflective characteristics: different materials exhibit different relationships between output value and distance.

The sensor is available in various alternative versions. Here we will just mention the GP2D05, which has the facility for setting a threshold distance to the target object using a potentiometer. This device operates without an external clock and requires only a trigger signal to initiate a measurement cycle. This simplifies the electronics, but does not allow us to measure actual distances.

Although the GP2D02 is specified for a supply voltage of 4.4 V to 7 V, the voltage at the V_{in} pin should not be allowed to exceed 3 V. As can be seen from the circuit diagram in **Figure 5**, this can be prevented by fitting a protection diode. In the example circuit the distance to the target object is measured continuously. The measured value from the GP2D02 is shown as a three-digit decimal value on the display. In addition, pin P1.2 of microcontroller IC1 is connected to a buzzer (with integrated oscillator) which is turned on when an object comes closer than a distance set via P2.0 and P2.1. The settings are as follows:

P2.0	P2.1	Threshold value
H	H	> 219
L	H	> 209
H	L	> 199
L	L	> 179

Pins P2.0 and P2.1 are fitted with internal pull-up resistors, so for a threshold value of 219 no external connections are necessary. Of course, the switched output could be used to operate a relay (with flyback diode!) in order to switch other equipment. P1.2 is an active-low open drain output.

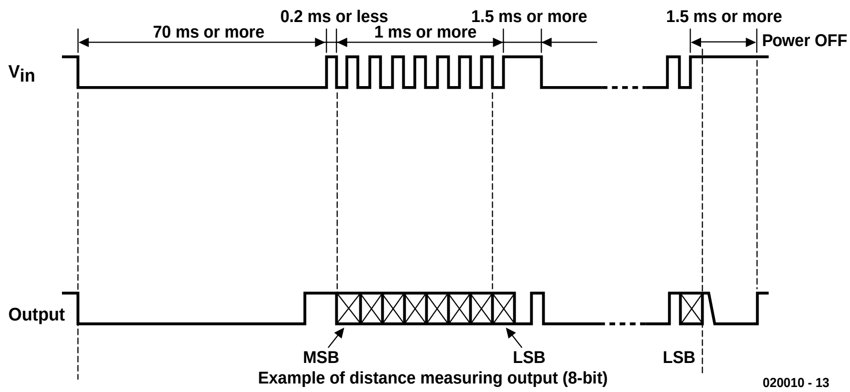


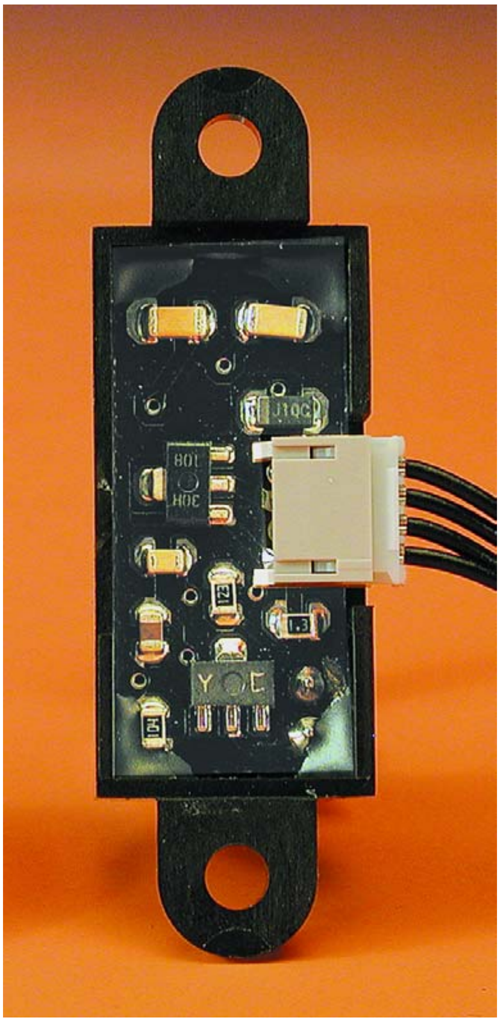
Figure 3. Timing diagram for reading the digital distance value.

Display module

IC3 is a highly-integrated LED display system, which, in a very small area and without additional circuitry, allows 128 different characters to be shown. In a space just 20 mm by 10 mm by 5 mm, this tiny display includes a ROM, a multiplexer and a driver for the individual LEDs of the 5-by-7 matrix that forms each character. In total four characters can be displayed and separately addressed. The relevant character position is addressed using pins A0 and A1, while the data, presented on pins D0 to D6, are written in by taking the WR pin low. A low level on the CLR pin clears the internal RAM (which is not used in this application). The display can be dimmed by applying a squarewave signal to the BL pin. It is also possible to cascade display modules, connecting all the pins (except WR) in parallel.

Programmed microcontroller

Thanks to its architecture and simple programming, the Philips microcontroller is ideal for this system. It includes 2 k of ROM, 128 bytes of RAM, two 16 bit timer/counters



along with other internal circuitry which is not used in this application. It is worth noting that the (RC) clock oscillator and the reset circuit are fully integrated into the device and no external circuitry is required.

The software, which may be obtained on floppy disk or as Free Download from our website, is relatively straightforward. After initialising the microcontroller, including clearing the 128 bytes of RAM, the main program begins. Three subroutines are executed in a continuous loop.

In the *Sensor* routine, the distance sensor

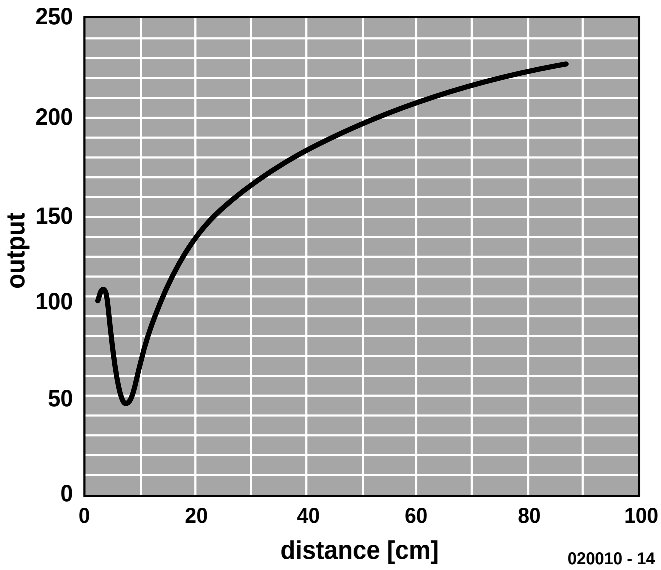


Figure 4. Relationship between digital output value and distance.

is read as specified in its data sheet. After a specified delay the V_{in} pin is clocked eight times and a bit is read on each falling edge, MSB first. The GP2D02 enters a quiescent mode when V_{in} remains high for more than 1.5 ms.

In the *Threshold* ('Grenzwert') subroutine, the measured value is tested to determine whether it exceeds the value set on pins P2.0 and P2.1. If so, output P1.2 is taken to ground, turning on the connected buzzer.

In the *Display* ('Anzeige') subroutine, the measured value is first converted from hexadecimal form into decimal. The SLR2016 intelligent display module is driven by presenting the appropriate data on D0 to D6 and corresponding addresses on A0 and A1. On the falling edge of WR the data are transferred and then displayed.

(020010-1)

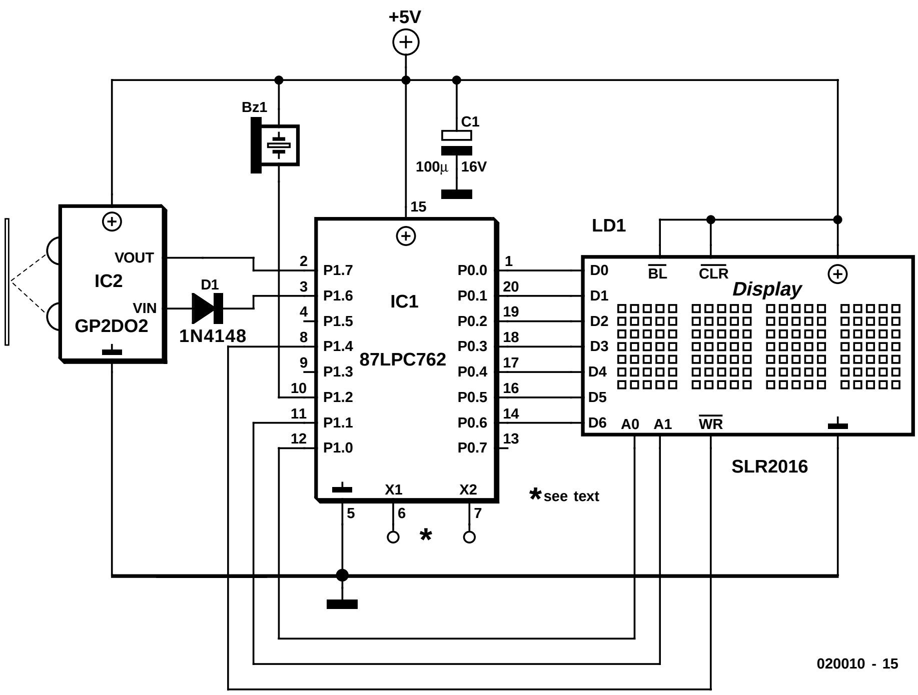
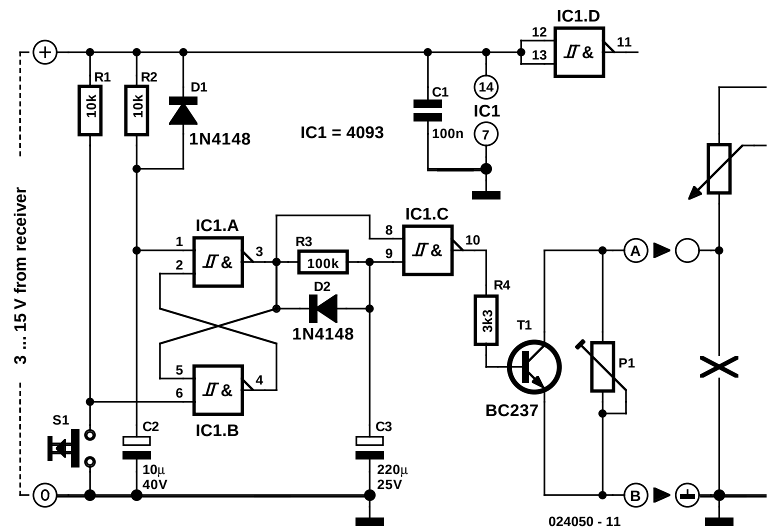


Figure 5. Circuit diagram of the distance measuring system.

Waking to Music

001

U. Reiser



Inexpensive clock radios do not have separate slumber and wake-up volume settings. However, with a small, easily fitted circuit you can go to sleep with quiet music and still be sure of waking up on time. This is achieved without the cumbersome fitting of a second potentiometer. Instead, the circuit boosts the volume level somewhat relative to the set level. For this purpose, the ground lead of the volume pot is opened and a trimpot is inserted. The setting of this trimpot is a matter of personal taste and also depends on the setting of the 'real' volume pot. As a general rule, you should use around one quarter of the setting of the original pot. A transistor is connected in parallel with P1 in order to short-circuit the trimpot and thus guarantee that the volume can be fully reduced to zero. Both of the additional components should be soldered directly to the original pot for better hum suppression.

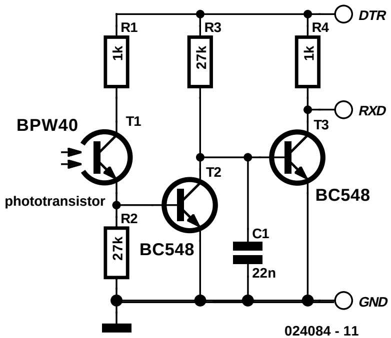
An R-S flip-flop made from AND gates and delay networks is used to control the transistor. The flip-flop (IC1a and IC1b) is set with a High level at the output of IC1a by a Low pulse from RC network R2/C2. The signal from IC1a reaches the two inputs of IC1c after being delayed by R3 and C3. If both of these inputs are High, IC1c cuts off the transistor. Due to the time delay, the wake-up music begins at the reduced volume level and then changes to the set level when the time delay expires. The volume level can be reduced manually by pressing reset switch S1, which pulls the input of gate IC1b Low and causes the R-S flip-flop to toggle to the opposite state. The bistable stage remains in this state until the supply voltage is switched off.

(024050-1)

Simple IrDA Receiver

002

B. Kainka



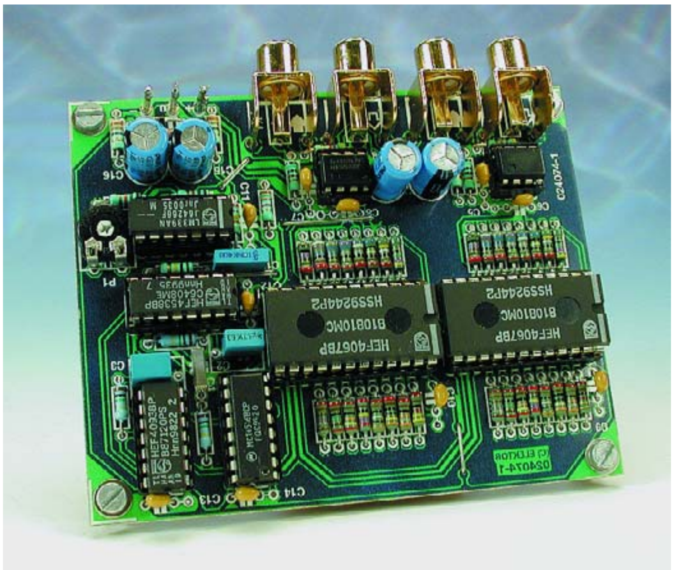
Palm Pilots are just one of the devices that use the IrDA standard for infrared data communication. Most Desktop PCs are not equipped with an IrDA interface so this simple circuit conveniently converts the PC serial port into an IrDA receiver. The circuit needs to detect and stretch the received infrared pulses so that the output signal conforms to the standard serial data format. The circuit is designed to operate at 9600 Baud and uses just two NPN transistors and a phototransistor. The 22-nF capacitor performs the job of lengthening the signals. It is important to ensure that the DTR signal is set to a high state on the serial port settings because this signal is used as the power supply to the circuit. The sample program listed below is written in HotPaw Basic for the Palm and tests the interface by sending a short greeting followed by a sequence of integers.

```
#irdatx.bas
open “com1:”,9600, ir as #5
 print#5,”Hello”
for n= 1 to 100
 print#5,n
 a= fn wait(1)
next n
close #5
end
```

(024084-1)

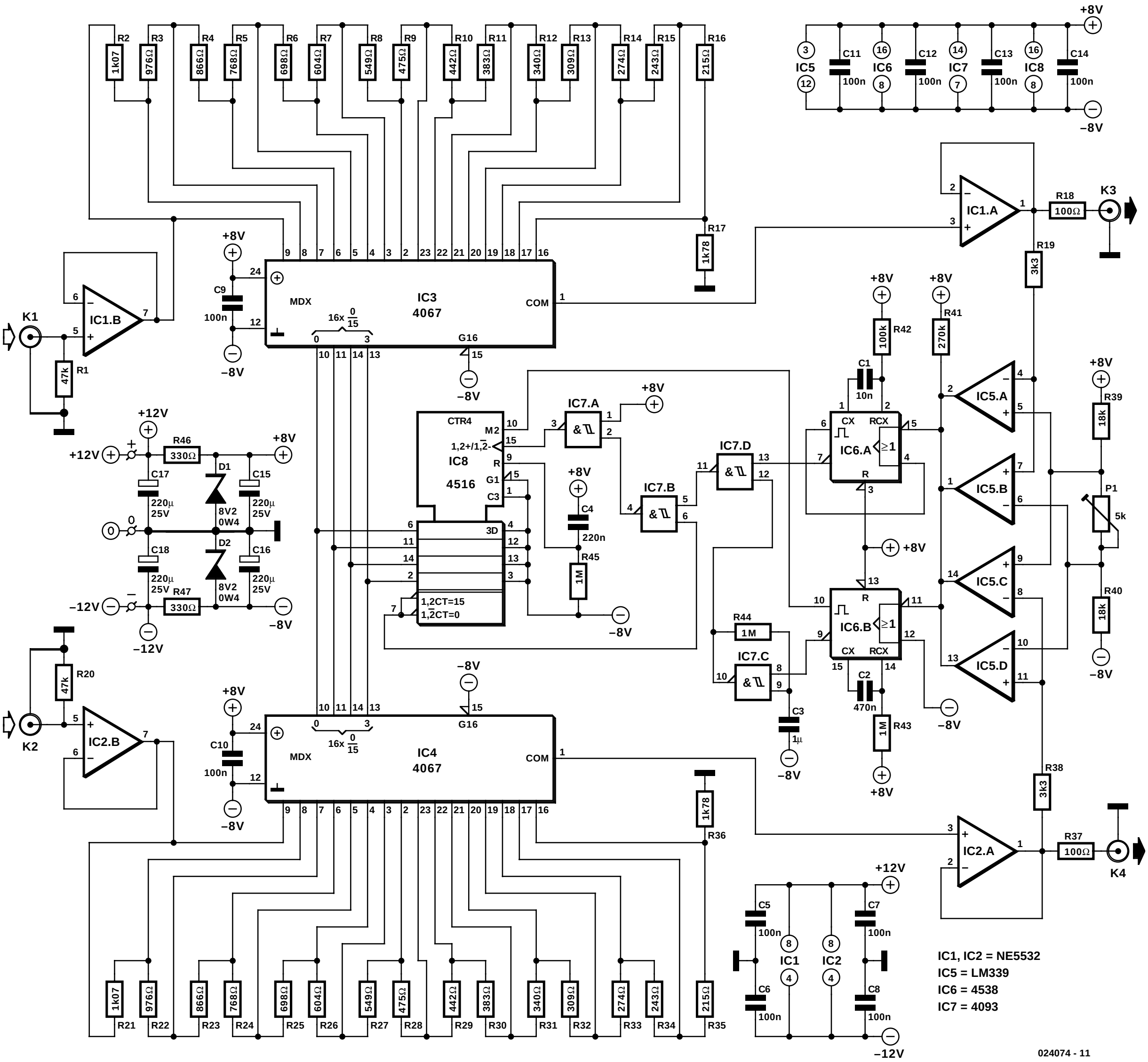
Audio Limiter (for DVD)

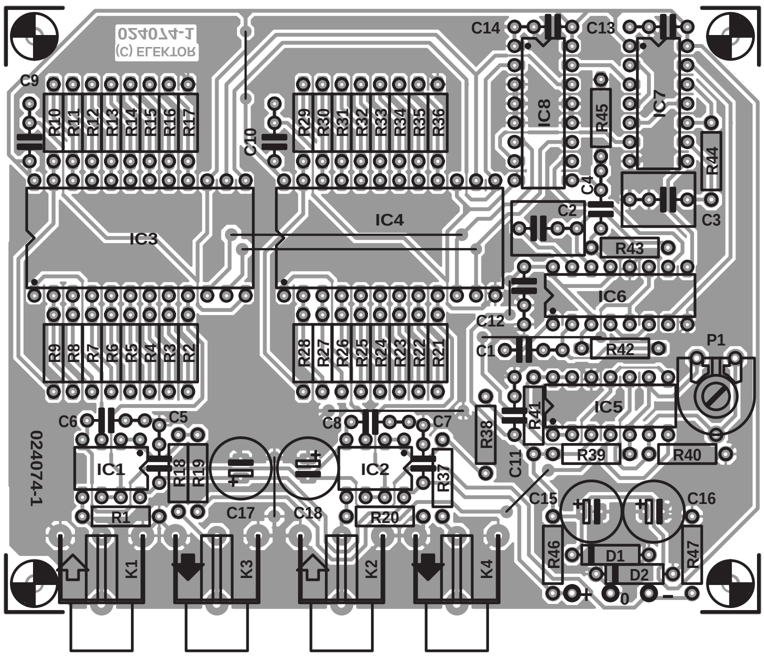
003

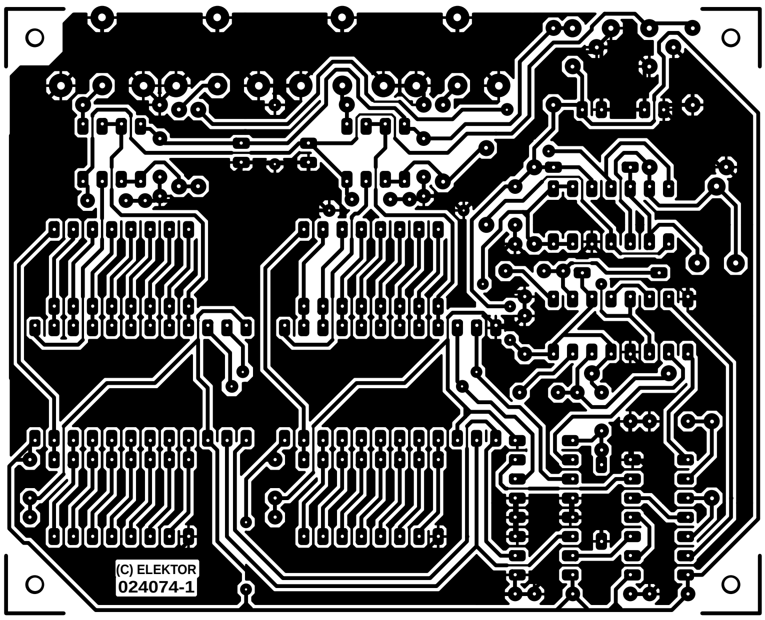


Those of you who are lucky enough to possess a DVD player and have watched a movie with lots of special effects, will certainly have noticed that the dynamic range of the audio can be very extreme. So much so that during normal use it is desirable to take steps in the form of an automatic volume limiter. If the DVD player is connected to an audio installation, then it is, in principle, not difficult to install such a limiter between the player and the audio system. It is important, of course, that the limiter does not introduce any distortion.

The audio limiter presented here limits the volume practically immediately and then slowly returns it to the normal level. The principle of operation is the same as that of a classical volume control. For each channel, the limiter comprises a voltage divider (R2 to R17 and R21 to R36) and a 16-channel analogue multiplexer/de-multiplexer type 4067 (IC3 and IC4). The voltage dividers are buffered at both the input and the output by a dual







COMPONENTS LIST

Resistors:
R1,R20 = 47kΩ
R2,R21 = 1kΩ07
R3,R22 = 976Ω
R4,R23 = 866Ω
R5,R24 = 768Ω
R6,R25 = 698Ω
R7,R26 = 604Ω
R8,R27 = 549Ω
R9,R28 = 475Ω
R10,R29 = 442Ω
R11,R30 = 383Ω
R12,R31 = 340Ω
R13,R32 = 309Ω
R14,R33 = 274Ω
R15,R34 = 243Ω
R16,R35 = 215Ω
R17,R36 = 1kΩ78
R18,R37 = 100Ω
R19,R38 = 3kΩ3
R39,R40 = 18kΩ
R41 = 270kΩ
R42 = 100kΩ
R43,R44,R45 = 1MΩ
R46,R47 = 330Ω
P1 = 5kΩ preset

Capacitors:
C1 = 10nF
C2 = 470nF
C3 = 1µF MKT, lead pitch 5 or 7.5 mm
C4 = 220nF
C5-C14 = 100nF ceramic
C15-C18 = 220µF 25V radial

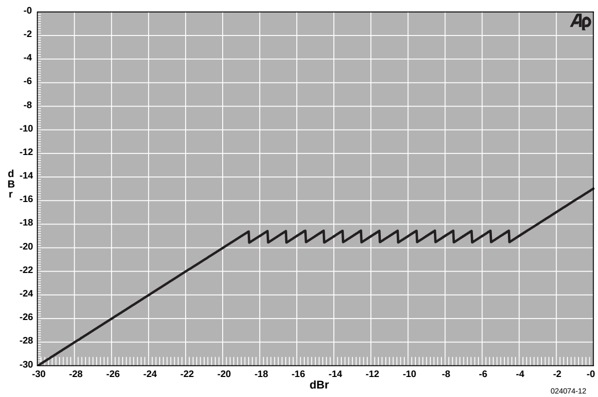
Semiconductors:
D1,D2 = zener diode 8V2 0.4W
IC1,IC2 = NE5532
IC3,IC4 = 4067
IC5 = LM339
IC6 = 4538
IC7 = 4093
IC8 = 4516

Miscellaneous:
K1-K4 = cinch (RCA) socket, PCB mount, e.g., Monarch T-709G
PCB, order code **024074-1**

opamp. The limiter has a control range of 15 dB in steps of 1 dB, which, to the ear, results in a smooth control. When the limiter is not active, the amplification is equal to 1.

The detection of the level occurs at the output with the aid of a window comparator for each channel (IC5a to d). To keep things simple, only the absolute peak value is measured and compared with an adjustable reference (R39/P1/R40). The reference voltage can be adjusted from 0 to 1 V. At the maximum output signal of 2 V_{rms} this means an attenuation of 9 dB of the maximum output voltage. The range of the limiter is intentionally limited to 15 dB, to make sure that not the entire dynamic range is removed. The best setting is that when during normal volume (a conversation for example) the limiter does not yet act, but starts to reduce the level at a few dB more than that.

The outputs of the comparators provide the trigger pluses for the two monostable multivibrators (IC6) that provide the correct drive for the binary up/down counter (IC8). We make use of the falling edges of these pulses because they are steeper. IC6a turns every trigger pulse into a 1-ms pulse for the counters. This way it is avoided that at the highest frequencies the limiter will immediately limit to the maximum amount, but that a short amount of time is required (the attack time). This design has the feature that at frequencies below 1 kHz the lim-



> **A few test results for completeness' sake (0dBr = 2 Vrms):**
>
> | | |
> |---|---|
> | Current consumption: | ± 26 mA |
> | THD+N | |
> | at 1 kHz : | 0.0012% (0 dBr, Gain = –15 dB) |
> | at 20 kHz: | 0.0058% (0 dBr, Gain = –15 dB) |
> | at 20 Hz to 20 kHz: | 0.0054% (10 dBr, Gain = 0 dB) |

iter needs 15 times the period of the applied signal to limit to the maximum extent. With a pure 20 Hz signal this will take 0.75 seconds, but in practice the signal is much more complex and therefore will occur much faster.

IC6b is triggered at the same time as IC6a and has, in combination with a number of NAND-gates, a dual function. On the one hand, the counter pulses from IC6a are passed on via IC7c and IC7d whenever IC6b is active. $\overline{Q}$ of IC6b will be 'low', the output of IC7c 'high' and IC7d is enabled. In the other hand, the Q output of IC6b defines the counter direction. With every clock pulse the multiplexers will connect a lower tap of the voltage divider to the output. IC7b and IC7a make sure that the carry out (also called terminal count) prevents the counter from wrapping around (both when counting up and down).

Once the input signal has reduced back down and after the period of IC6b, the oscillator IC7c will be released and the counter will count down. The time it takes for the level to start to increase is defined by IC6b and amounts to approximately half a second (t = R43 ? C2). The rate at which the increase happens is determined by the frequency of oscillator IC7c.

R45/C4 form a power-up reset for the counter. The power supply voltage for the opamps is higher than the voltage for the digital circuitry because the opamps have a knee-voltage of a few volts at the output and have a common mode range at the input that is smaller than their power supply voltage. This way, maximum use can be made of the available voltage range of the multiplexers. The power supply for the digital section of the circuit is provided by two zener diodes D1/D2 and decoupled by R46/C15 and R46/C16, and in this way is nicely symmetric around the analogue ground level. Considering the logic levels for the trigger pulses, the comparators are also connected to the digital power supply. The total current consumption of the entire circuit is around 26 mA.

In conclusion, the adjacent graph clearly shows how the limiter influences the audio signal, because the output signal is shown as a function of the input signal with a large number of measurement points. When the input signal is slowly increasing, the output signal will follow the input until the set level is reached. Once the input signal exceeds the reference value the output amplitude is immediately reduced by 1 dB; the output then continues to follow the input until the output level reaches the reference again, etc. This will occur up to 15 times in total, at which point the output will follow the input, but attenuated by 15 dB.

(024074-1)

4-Bit Decimal Display

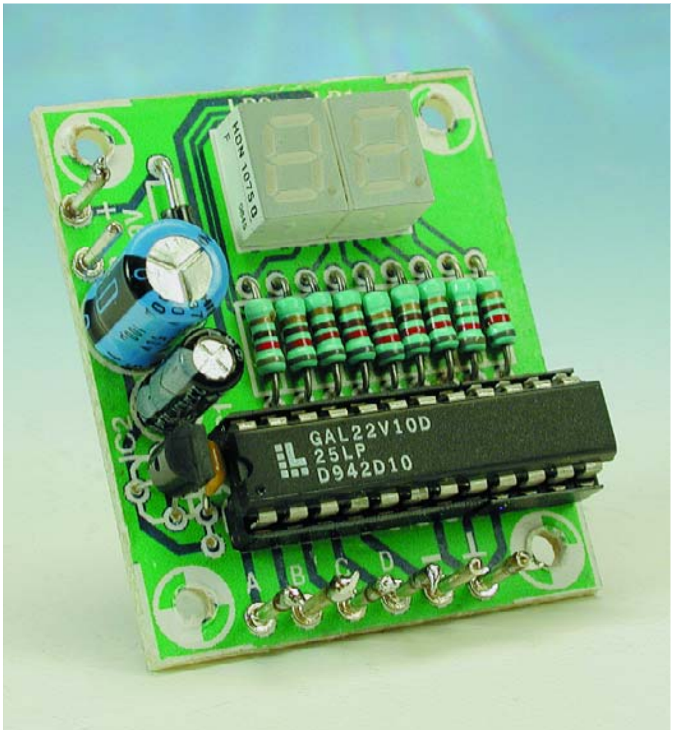
004

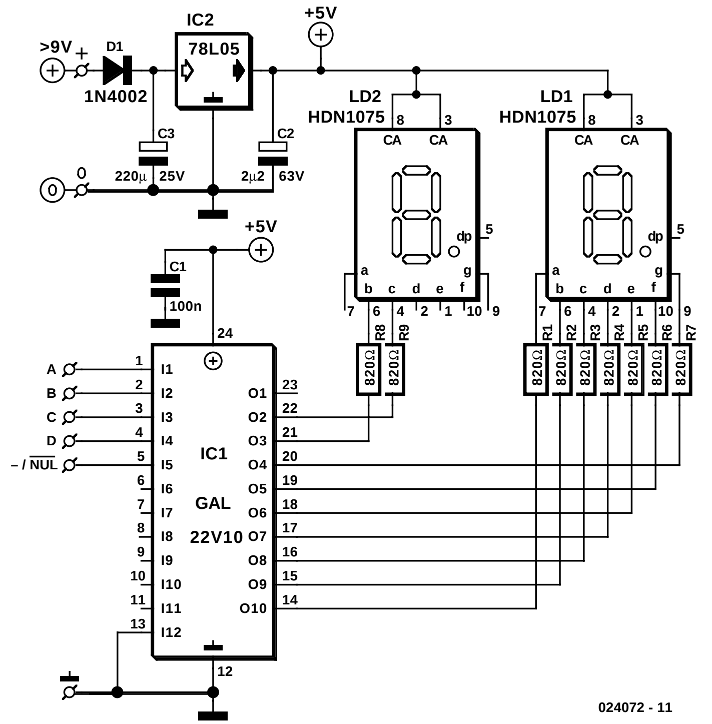
Display driver ICs are available in several standard implementations. This circuit makes use of a GAL 22V10 to drive two 7-segment displays without multiplexing. A 4-bit binary code at inputs A/B/C/D is converted to a decimal number. An example of an application is the 'Audio Limiter (for DVD)', but take note of the voltage levels! The multiplexers used there are driven by a 4-bit binary counter. This circuit will give a better indication of the behaviour and settings of the limiter.

The segments that have to light up with the various input bit combinations are shown on the accompanying table.

This table is used as the basis for writing the equations that result in the program for the GAL. Use can be made of either the max- or the min- terms depending on whichever results in the smallest number. The outputs are active low, because they are able to sink more current than they can source. The resistors R1 through R9 are selected such that the resulting current through each of the segments is about 3mA. The displays, therefore, must be common-anode. The displays used here are very small, the characters are only 7 mm tall (the displays are 10 mm high in total). The printed circuit board is actually intended to be more of an example, because in the final application it may be more desirable to fit LD1 and LD2 on a separate PCB.

As can be seen from the table, there is an extra feature when the inputs are 0000, in which case only a single dash is displayed (segment g of LD1). When the corresponding input is left open a dash is visible (low level will give a '0'). With the aforementioned limiter this means that no attenuation is taking place.





The circuit is provided with its own 5-V regulator (78L05, take note of the dissipation!) so finding a suitable power supply should not be problem. The current consumption is a minimum of about 60 mA (indication '–') to about 85 mA maximum (indication '10'). The jedec file you'll need to program the GAL may be obtained as a Free Download from the *Elektor Electronics* website. The PCB shown here is unfortunately not available ready-made.

(024072-1)

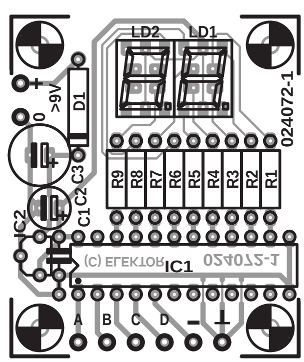
	b2	c2	a1	b1	c1	d1	e1	f1	g1	D	C	B	A
–	0	0	0	0	0	0	0	0	1	0	0	0	0
0	0	0	1	1	1	1	1	1	0	0	0	0	0
1	0	0	0	1	1	0	0	0	0	0	0	0	1
2	0	0	1	1	0	1	1	0	1	0	0	1	0
3	0	0	1	1	1	1	0	0	1	0	0	1	1
4	0	0	0	1	1	0	0	1	1	0	1	0	0
5	0	0	1	0	1	1	0	1	1	0	1	0	1
6	0	0	1	0	1	1	1	1	1	0	1	1	0
7	0	0	1	1	1	0	0	0	0	0	1	1	1
8	0	0	1	1	1	1	1	1	1	1	0	0	0
9	0	0	1	1	1	1	0	1	1	1	0	0	1
10	1	1	1	1	1	1	1	1	0	1	0	1	0
11	1	1	0	1	1	0	0	0	0	1	0	1	1
12	1	1	1	1	0	1	1	0	1	1	1	0	0
13	1	1	1	1	1	1	0	0	1	1	1	0	1
14	1	1	0	1	1	0	0	1	1	1	1	1	0
15	1	1	1	0	1	1	0	1	1	1	1	1	1

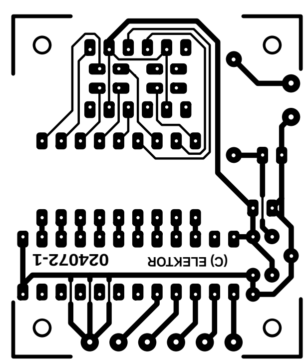
COMPONENTS LIST

Resistors:
R1-R9 = 820Ω

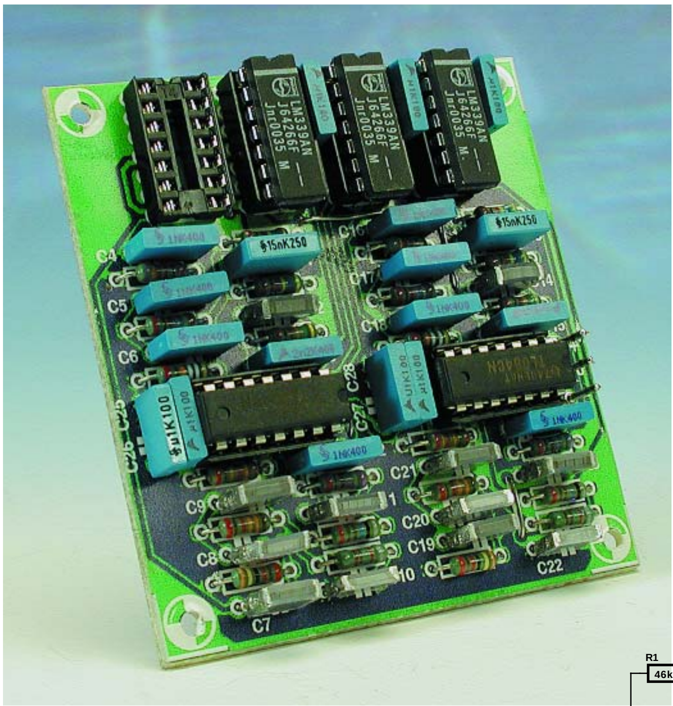
Capacitors:
C1 = 100nF ceramic
C2 = 2µF2 63V radial
C3 = 220µF 25V radial

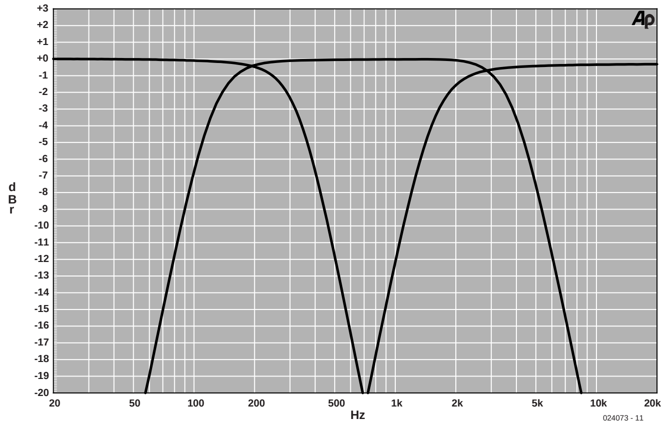
Semiconductors:
D1 = 1N4002
IC1 = GAL22V10
IC2 = 78L05
LD1,LD2 = HDN1075





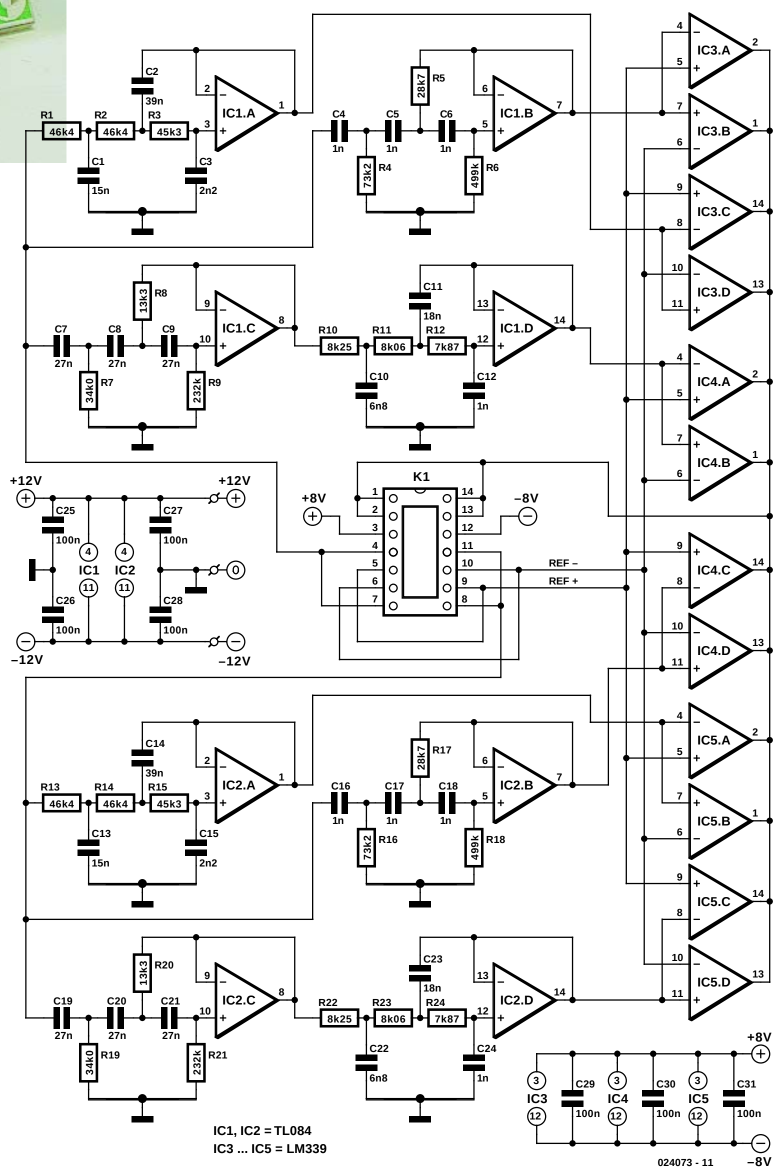
Filter for Audio Limiter (for DVD) 005





In the 'Audio Limiter (for DVD)' circuit, the peak values of the audio signals are used to reduce the dynamic range of the sound. A possible disadvantage is that the entire audio spectrum is used to determine the level, so that peak levels in the low or high frequencies may lead to suppression of, for example, voices in the mid frequency range. If we divide the spectrum into three ranges and for each range a separate window comparator defines the signal level then the signals in one range will have a smaller influence on the other two ranges. It is the intention of this filter, therefore, that the notorious 'breathing' of the limiter is reduced.

The filters proposed here are standard 3rd order types with crossover frequencies of 200 Hz and 2.5 kHz. IC1a/IC2a form the low-pass filters for the low range, IC1b/IC2b are the high-pass filters for the high range, and IC1c/IC2c and IC1d/IC2d the high- and low-pass respectively for the mid range. The crossover frequencies are not simply the corner frequencies of the filters, but these frequencies have been calculated such that the curves cross when the attenuation is 0.25 dB. This way the detected amplitude remains approximately equal across the entire audio spectrum. The real corner frequency with a 3rd order Butterworth is a ratio of 1.6 further away than the –0.25 dB point. The curve shows what this looks like in practice.

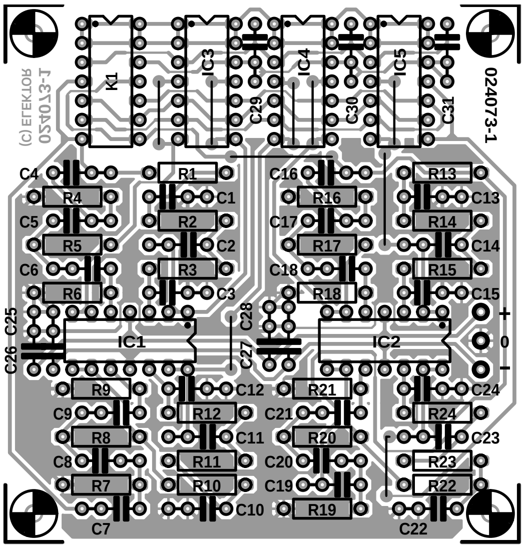


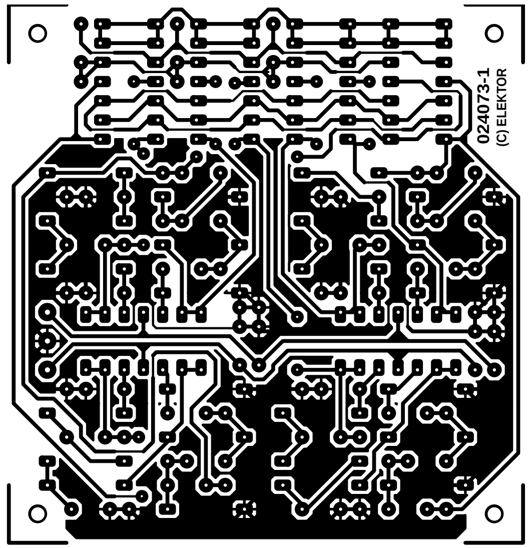
At the crossover from the mid to high range, the high-pass filter has a little more damping and it appears therefore that the crossover point has shifted slightly. This is of no real consequence in practice.

The connection to the audio limiter is made with a 14-pin DIL-connector to the socket for the comparator of this limiter. This filter utilises the same DIL connector (K1) so that the connection can be made with a short length of ribbon cable. The power supply for the comparators is also connected through this ribbon cable to supply power for the filter. The power for the opamps, however, has to be taken from the power supply with three separate wires. The increase in current consumption of the limiter is about 15 mA.

In addition, a couple of small changes have to be made to the limiter: R19 and R38 (both 3k3) have to be replaced with 47-Ω resistors. Otherwise the input impedance of the filters will affect the level of the input voltage. The PCB shown here is unfortunately not available ready-made.

(024073-1)





COMPONENTS LIST

Resistors:

R1,R2,R13,R14 = 46kΩ4
R3,R15 = 45kΩ3
R4,R16 = 73kΩ2
R5,R17 = 28kΩ7
R6,R18 = 499kΩ
R7,R19 = 34kΩ0
R8,R20 = 13kΩ3
R9,R21 = 232kΩ
R10,R22 = 8kΩ25
R11,R23 = 8kΩ06
R12,R24 = 7kΩ87

Capacitors:

C1,C13 = 15nF
C2,C14 = 39nF
C3,C15 = 2nF2
C4...C6,C12,C16,C17,C18,C24 = 1nF
C7,C8,C9,C19,C20,C21 = 27nF
C10,C22 = 6nF8
C11,C23 = 18nF
C25-C31 = 100nF

Semiconductors:

IC1,IC2 = TL084
IC3,IC4,IC5 = LM339

Miscellaneous:

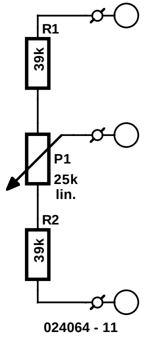
K1 = 14-way DIL connector (2 off)
14-way flatcable

Joystick Replacement

006

The joysticks used in games and modelling contain two potentiometers with a resistance of about 100 kΩ, which turn through 60 to 90 degrees. In fact only one third to one quarter of the total resistance is used in these potentiometers. The diagram shown here should be used when making your own joystick with ordinary potentiometers that turn through 270 degrees. The values for R1 and R2 are given as guidelines only and their optimal value should be found through trial and error. It will be easier if you temporarily replace R1 and R2 with a combination of a fixed resistor and a preset, since it can be a time consuming job to determine the correct values; this is because each of the resistors affects the other.

(024064-1)



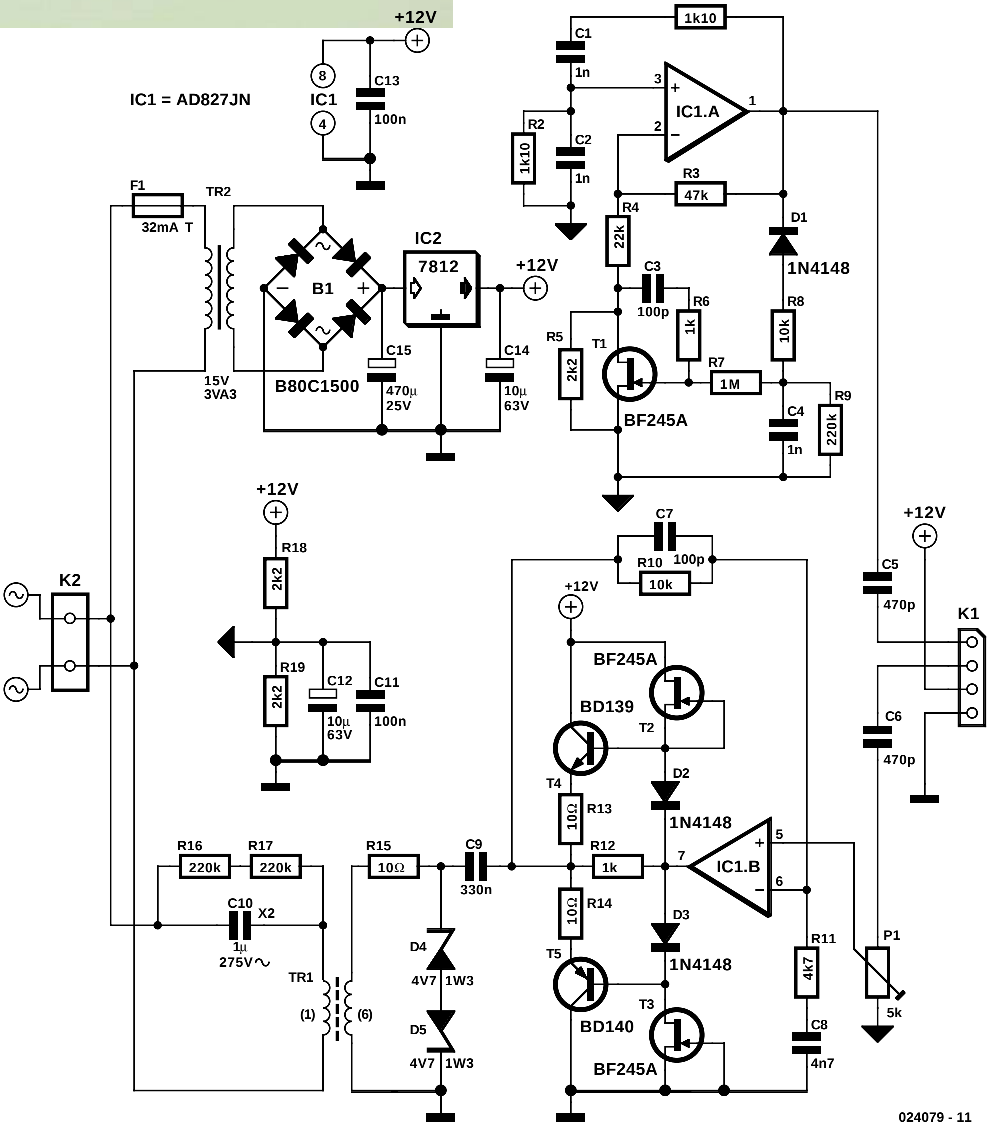
Mains Remote Transmitter

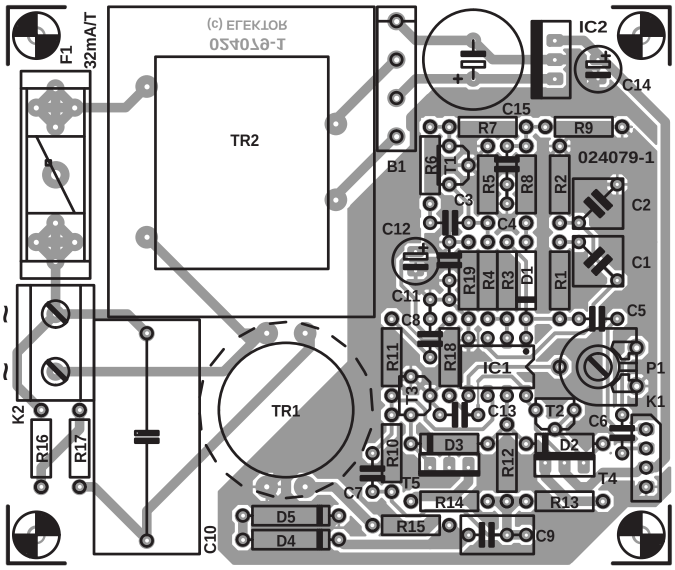
007

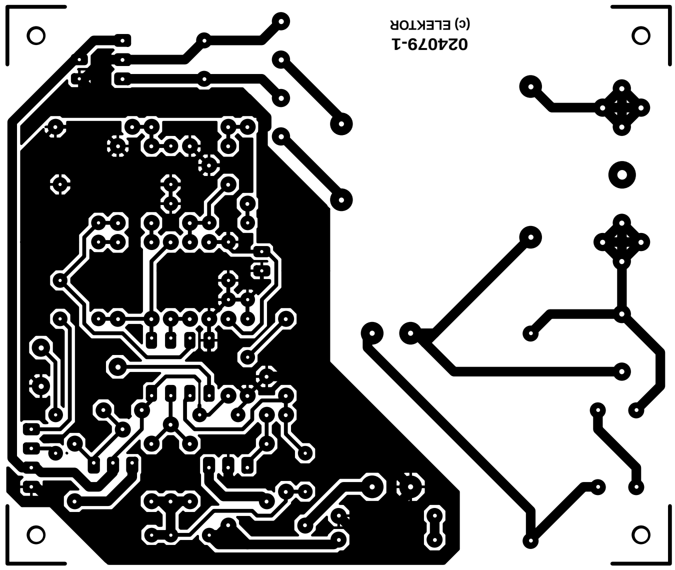
This circuit can be used to superimpose a 143-kHz carrier on the mains voltage, which allows various applications to be realised. One example is the 'Mains Remote Switch'. Besides the power supply, the circuit consists of only a sine-wave oscillator, a buffer stage and an output transformer for isolation from the mains network.

The oscillator, which is built around IC1a, is a standard Wien-bridge design whose frequency is determined by R1, C1, R2 and C2. The combination of R3, R4 and the amplitude stabilisation circuitry built around T1 provides a gain of 3. FET T1 is used here as a controllable resistance, with R6, R7 and C3 providing a certain amount of linearisation of the channel resistance. The output voltage is rectified by D1 to obtain a negative voltage (with respect to virtual ground), which is smoothed by C4/R9 and applied to the gate of T1 via R7. If the amplitude increases, the channel resistance increases due to the larger negative gate voltage, so the gain of IC1a decreases. The characteristic of the FET thus determines the output voltage of the oscillator. With the type BF254A used here, the peak-to-peak value of output voltage is approximately equal to half the supply voltage, but it must be noted that the FET characteristics are subject to a considerable degree of device-to-device variation.

A type AD827 was selected for the opamp since it is fast enough to have a minimal effect on the oscillation conditions. The frequency is set to 143 kHz, since this value happens to fall nearly in the middle of the band from 140 kHz to 148.5 kHz (Cenelec standard 50065-1) when E24 values are used for the frequency-determining







COMPONENTS LIST

Resistors:
R1,R2 = 1k10 1%
R3 = 47kΩ
R4 = 22kΩ
R5,R18,R19 = 2kΩ2
R6,R7 = 1MΩ
R8,R10 = 10kΩ
R9,R16,R17 = 220kΩ
R11 = 4kΩ7
R12 = 1kΩ
R13,R14,R15 = 10Ω
P1 = 5kΩ preset

Capacitors:
C1,C2 = 1nF 1%
C3,C7 = 100pF
C4 = 1nF
C5,C6 = 470pF
C8 = 4nF7
C9 = 330nF
C10 = 1µF 275VAC, Class X2, lead pitch 27.5 mm
C11 = 100nF
C12,C14 = 10µF 63V radial
C13 = 100nF ceramic, lead pitch 5 mm
C15 = 470µF 25V radial

Semiconductors:
D1,D2,D3 = 1N4148
D4,D5 = zener diode 4V7 1.3W
T1,T2,T3 = BF245A
T4 = BD139
T5 = BD140
IC1 = AD827JN Analog Devices (Farnell)
IC2 = 7812

Miscellaneous:
K1 = 4-way pinheader
K2 = 2-way PCB terminal block, lead pitch 7.5mm
B1 = B80C1500 (rectangular case) (80V piv, 1.5A)
F1 = fuse, 32mAT (slow), with PCB mount holder and cap
Tr1 = 6:1 N30 core 16x6.3 mm EPCOS B64290L45X830 (Farnell)
Tr2 = mains transformer 15V >3VA, dims. 35x41mm, e.g., Hahn BV EI 382 1193 (15 V/4.5 VA) or Block VB 3,2/ 1/ 15 (15 V/3.2 VA, short-circuit resistant)

components. For general use, the maximum allowed voltage in this band is 116 dBµV.

The oscillator output is passed to the buffer stage IC1b via header K1. K1 provides the extra feature of allowing this signal to be modulated or coded using an external circuit. Depending on the circuit used for this purpose, it may be necessary to bypass C5. A potentiometer is placed at the input of IC1b to compensate for the tolerance variations of the oscillator. This allows the circuit to be adjusted to meet the requirements of the standard.

Two small power transistors (the ‘old faithfuls’ BD139 and BD140) are wired as complementary emitter followers in the output stage of the buffer. The quiescent current through the output stage depends on the voltage drop across D2/D3 and the value of the emitter resistors R13 and R14. Here the quiescent current is only a few milliampères. The maximum signal excursion is determined by the current sources T2 and T3 and the current gain of the output transistors. R12 provides better behaviour in the zero-crossing region.

In order to ensure a certain amount of isolation from the mains network, an output transformer (Tr1) is used. From the point of view of safety, though, it’s a good idea to regard the entire circuit as being connected to the mains potential and to bear this in mind when fitting it into an enclosure and using it in other applications. The primary winding of the transformer is driven via C9. The turns ratio of the transformer is chosen such that the maximum allowable value is achieved, but nothing significantly greater than this. Since the impedance of the mains network is a few tens of ohms, at 143 kHz a rather large capacitor (C10) is needed to isolate the 230-V mains voltage from the 143-kHz carrier signal. An X2 type must necessarily be used for this capacitor. R16 and R17 are placed in parallel with C10 to immediately discharge the voltage on K2 in the unlikely event that fuse F1 blows. R15, D4 and D5 protect the output of the amplifier stage against noise pulses and switch-on phenomena (i.e., against possible current spikes passing through C10).

Now for a couple of practical points. You will have to wind transformer Tr1 yourself, but this is not particularly difficult. The primary consists of 6 turns and the secondary is 1 turn. The core is an EPCOS type with a diameter of 16 mm, made from N30 material. Both windings are made using 1-mm diameter wire with synthetic insulation (total diameter 2.5 mm). The primary winding is split into two equal halves such that the secondary fits exactly between them. The leads of the transformer thus emerge on opposite sides. In order to increase the maximum insulation resistance, the original bare wire can be replaced by vanished wire.

The power supply follows the standard recipe of transformer, bridge rectifier and electrolytic capacitor, followed by a voltage regulator (IC2). Since the circuit operates from asymmetric supply voltages, voltage divider R18/R19 and decoupling capacitors C11/C12 are necessary to reference IC1 to half of the supply voltage. The supply voltage is also fed to connector K1 so that the stabilised +12 V is also available for possible expansion circuits.

(024079-1)

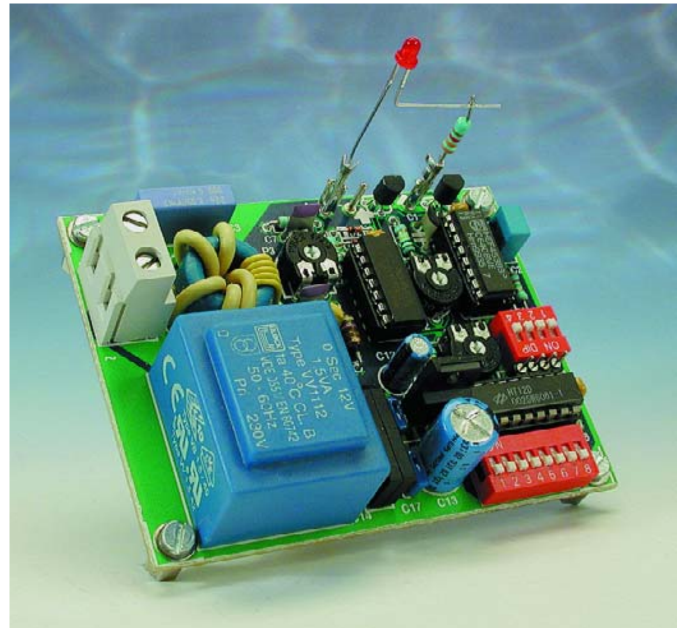
Mains Remote Control: Decoder 008

This receiver/decoder forms part of a simple mains network remote control system, which also includes the 'Mains Remote Transmitter' and the 'Mains Remote Encoder'.

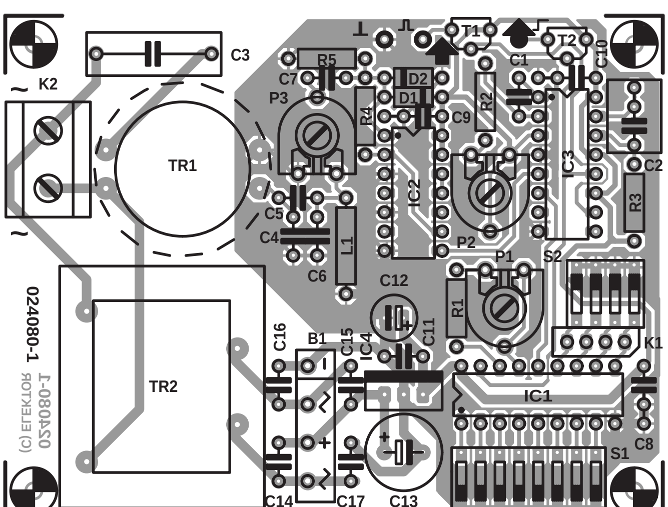
The decoder is built around IC1, which is a Holtek type HT12D or HT12F. For the receiver we use the same circuit as in the 'mains remote switch', namely a passive circuit tuned to approximately 143 kHz, since we assume that the transmitter is powerful enough to provide an adequate signal.

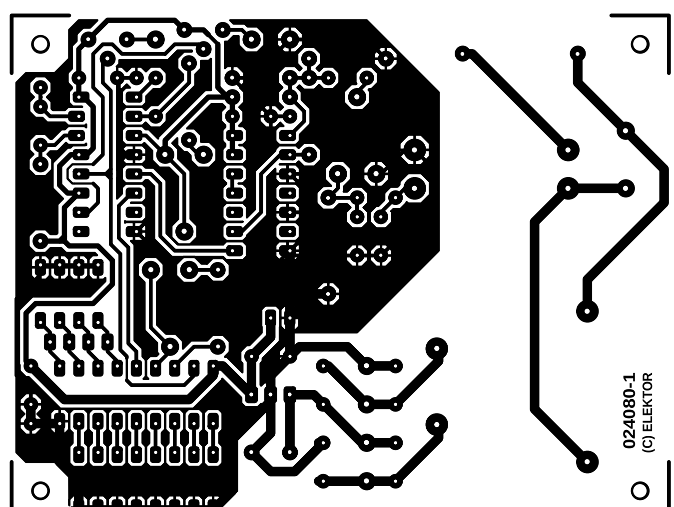
Two 4069U inverters (IC2) are used to convert the received signal to TTL levels. D1 and D2 provide extra protection against noise pulses and the like. The sensitivity can be adjusted using P3, but you should bear in mind that overdriving IC2 can cause corruption of the data. The trick with IC2 is that a small offset applied to the first buffer causes the second buffer to be displaced from the middle (which can be checked using a multimeter), so that the following monostable multivibrator (IC3, a 4538) receives a usable burst as a trigger signal. IC3a is retriggerable, which means that if a trigger pulse arrives within the set time, the output pulse is extended. However, if the set pulse width is too long, the output pulses are extended so much that the decoder will not recognise them as valid data.

IC3a thus recovers the originally sent code. P2 is added to the circuit to allow the pulse length to be adjusted as accurately as possible, but an oscilloscope is required for this. In practice, the adjustment is not all that critical and P2 can be simply set to its midrange position.



The output of IC3a is fed to the decoder (IC1), which compares the recovered code with the settings of S1 and S2. If the received code matches these settings, output VT goes High

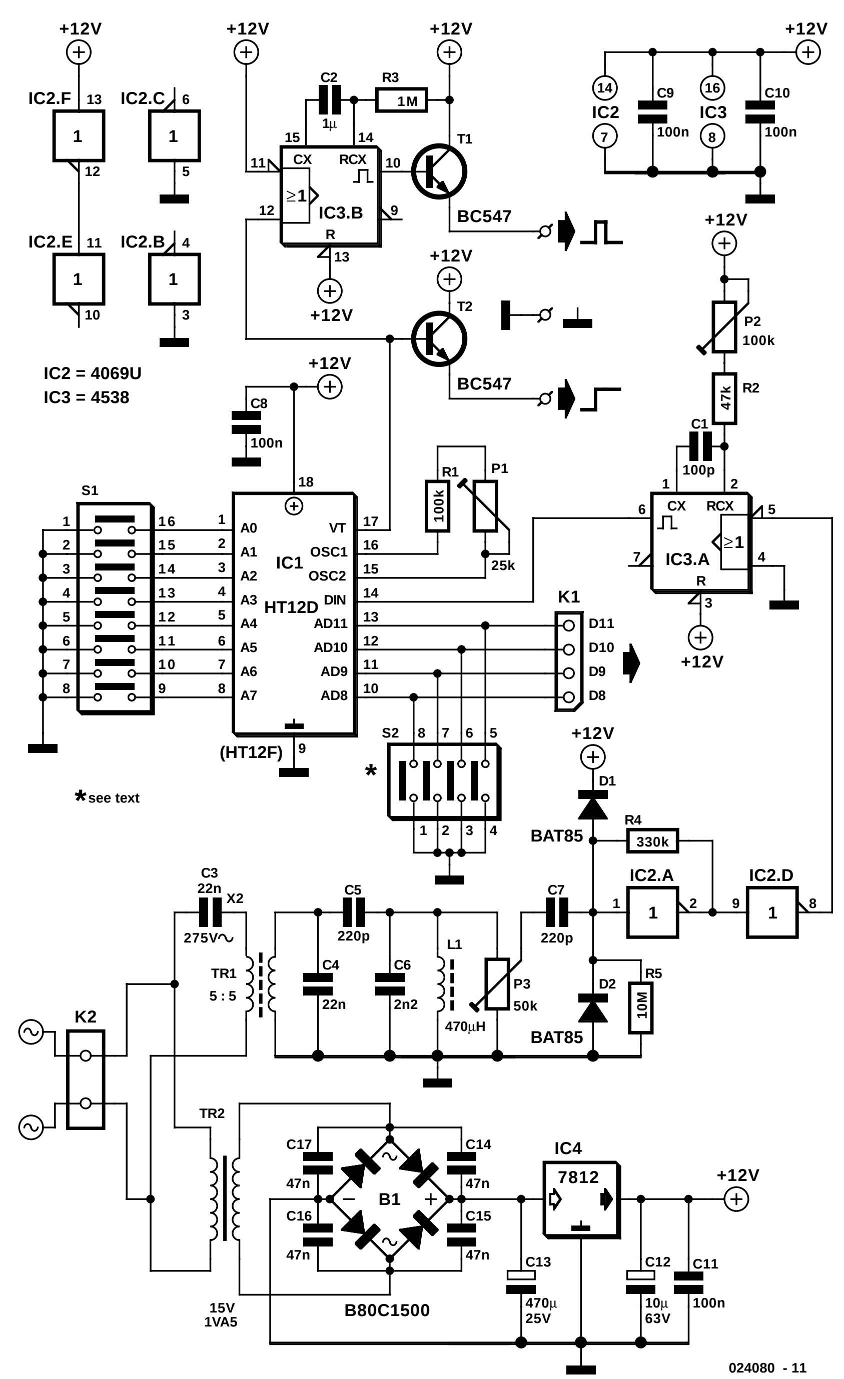




and some sort of application can be energised via buffer T2. If you have in mind connecting an active buzzer to the buffer output, you must thoroughly decouple it using a 10-mH coil in series and a 100-μF/16-V electrolytic capacitor in parallel, since these buzzers can be a source of stubborn interference. The second monostable (IC3b) is used to generate a supplementary pulse with a duration of roughly one second. The pulse length can be modified (by changing R2 and/or C2) to meet the needs of a particular application that requires a certain minimum duration. T1 acts as a simple buffer for this output.

As already noted, in principle two different types of decoder can be used: HT12D or HT12F. The HT12D has four data-bit outputs (AD8–AD11), with the data being made available on the SIL header K1. In this case it is better not to fit S2. If an HT12F is used for the decoder, K1 has no function, but a 12-bit address can be set.

Naturally, the oscillator of the decoder should be tuned to match the encoder used with the transmitter. For the HT12D/F, the oscillator frequency is 50 times that of the encoder. That means that here the oscillator must be set to around 112 kHz.



COMPONENTS LIST

Resistors:
R1 = 100kΩ
R2 = 47kΩ
R3 = 1MΩ
R4 = 330kΩ
R5 = 10MΩ
P1 = 25kΩ preset
P2 = 100kΩ preset
P3 = 50kΩ preset

Capacitors:
C1 = 100pF
C2 = 1μF MKT, lead pitch 5mm or 7.5mm
C3 = 22nF 275VAC, Class X2
C4 = 22 n ceramic, lead pitch 5mm
C5,C7 = 220pF
C6 = 2nF2 ceramic, lead pitch 5mm
C8,C9,C10 = 100nF
C11 = 100nF ceramic, lead pitch 5mm
C12 = 10μF 63V radial
C13 = 470μF 25V radial
C14-C17 = 47nF ceramic, lead pitch 5mm

Inductor:
L1 = 470μH miniature choke

Semiconductors:
D1,D2 = BAT85
T1,T2 = BC547
IC1 = HT12D/F (Holtek) (Farnell) *
IC2 = 4069U
IC3 = 4538
IC4 = 7812

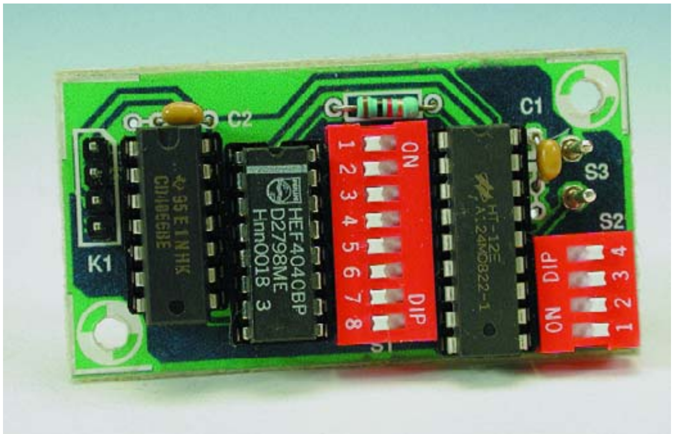
Miscellaneous:
K1 = 4-way pinheader
K2 = 2-way PCB terminal block, lead pitch 7.5mm
S1 = 8-wayDIP-switch
S2 = 4-way DIP-switch *
B1 = B80C1500 (rectangular) (80V piv, 1.5A)
TR1 = N30 ring core 16x6.3 mm EPCOS B64290L45X830 (Farnell) *
TR2 = mains transformer 15V/1.5VA, short circuit resistant, e.g., Block type VB 1,5/1/ 15

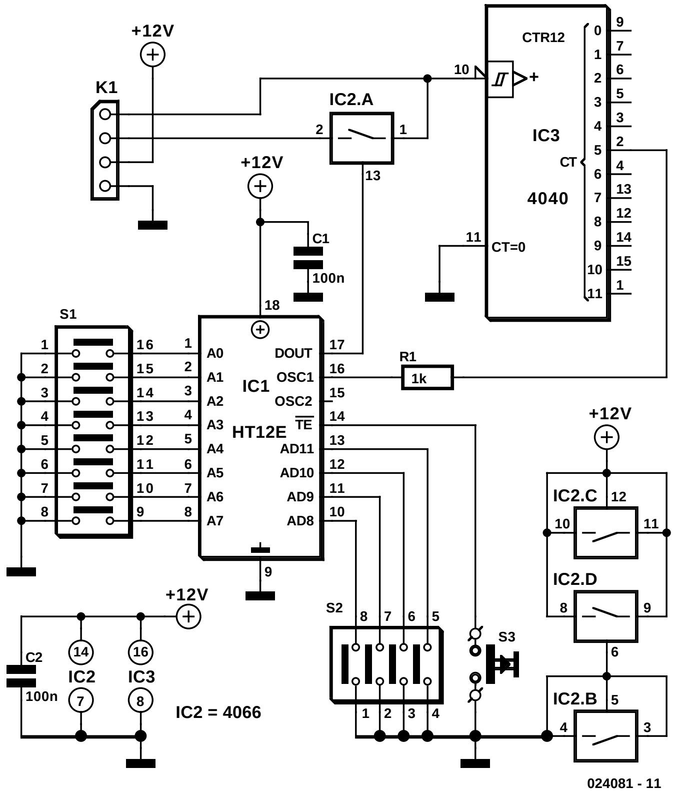
* see text

According to the related curve on the data sheet, thisrequires an external resistance of approximately 115 kΩ to be connected between the OSC1 and OSC2 pins. This can be precisely set using P1, and the potentiometer also allows for adjustments to compensate for various tolerances.
The power supply for the circuit is designed according to the usual standard configuration, with the transformer (Tr2) being intentionally somewhat overdimensioned to provide extra capacity for powering small applications (buzzer, LED etc.). Building the circuit is a simple task if the illustrated printed circuit board is used. Since the power supply (including the transformer) is fitted on the circuit board, the amount of wiring required is minimal.

(024080-1)

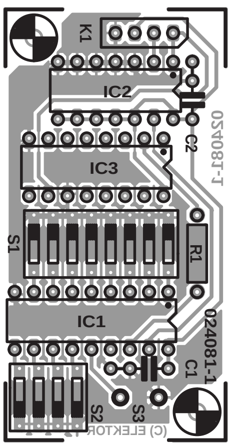
Mains Remote Control: Encoder 009

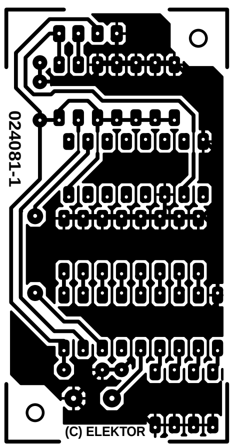




This application can actually be considered to be a slight adaptation of the standard use of the Holtek HT12E encoder (www.holtek.com). We have used this IC several times already, so it does not need any further explanation. The small circuit described here is intended to be used as an extension to the 'Mains Remote Transmitter', but it also clearly illustrates how the IC can be used in a non-standard manner.

The HT12E is normally used with its internal oscillator by connecting a resistance between the OSC1 and OSC2 pins. Here we instead use the carrier frequency of the transmitter. For this purpose, connector K1 of the transmitter is connected to K1 of the encoder. The 143-kHz signal generated by the oscillator in the transmitter is divided by 64 by the counter (IC3), producing an oscillator frequency of approximately 2.2 kHz for IC1. **Note**: for this application, coupling capacitor C5 in the





COMPONENTS LIST

Resistors:
R1 = 1kΩ

Capacitors:
C1,C2 = 100nF

Semiconductors:
IC1 = HT12E Holtek
(Farnell)
IC2 = 4066
IC3 = 4040

Miscellaneous:
K1 = 4-way SIL-header
S1 = 8- way DIP-switch
S2 = 4- way DIP-switch
S3 = pushbutton, 1 make contact

transmitter must be bypassed in order to ensure that IC3 receives a sine-wave signal centred at half the supply voltage as a clock signal. The Philips type 4040 used here has a Schmitt-trigger clock input, which allows the sine wave to be used as a source of 'clean' clock pulses.

The HT12E has an output that is not internally modulated (DOUT, pin 17). The carrier wave from the transmitter is modulated by using a type 4066 analogue switch to switch the carrier on and off. The nice thing about this is that the switching is synchronous, since the data output of the encoder is derived from the carrier wave. Instead of using an IR LED modulated at 36 kHz, here we modulate a 143-kHz signal and transmit the remote control signal via the mains network.

The encoder is enabled using S3. S1 and S2 determine the address of the transmitted code, with the setting of S2 serving as the transmitted data in the receiver if an HT12D decoder is used. R1 provides a certain amount of decoupling for the capacitor of the HT12E oscillator circuit. The remaining switches of the 4066 are not used. The maximum current consumption with S3 pressed is around 0.6 mA.

The illustrated circuit board layout is approximately the size and shape of a matchbox and guarantees problem-free construction of the encoder.

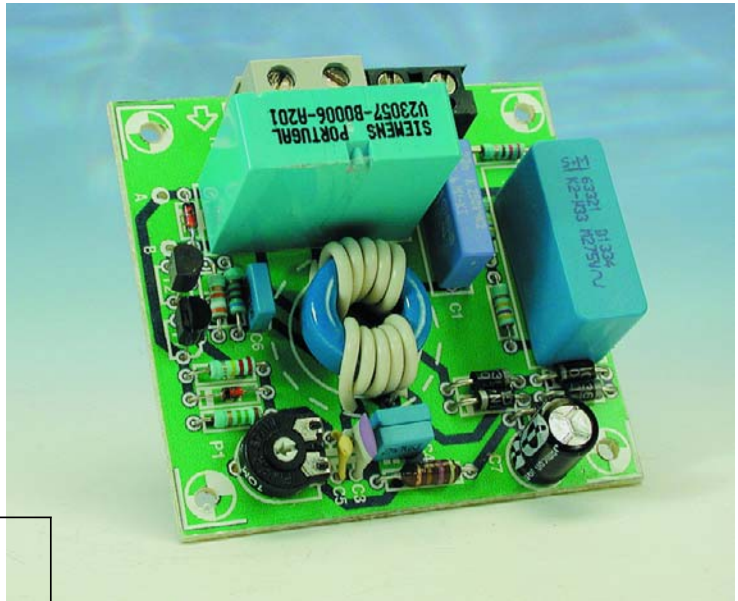
(024081-1)

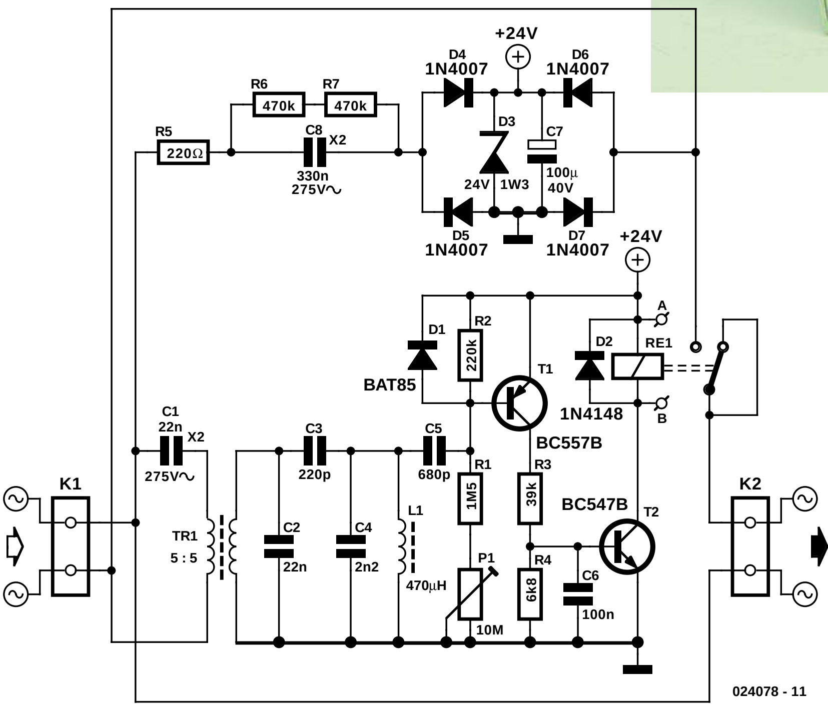
Mains Remote Switch

010

This compact design forms a remotely operated switch that receives its control signal via the mains voltage. The switch is operated using the 'mains remote transmitter' described elsewhere in this issue. With this transmitter, a switch should be connected between pins 1 and 2 of K1. Depending on the application, this must be either a press contact or a make contact.

The idea of the 'mains remote switch' is that a relay is energised in order to connect the mains voltage on K1 through to K2. The 'receiver' (a somewhat exaggerated term for such a simple design) is formed by Tr1 and the tuned circuit L1/C4. The network C1/Tr1/C2 serves as a coupled circuit tuned to the frequency of 143 kHz generated by the transmitter. The selectivity is determined by L1/C4 and is primarily dependent on the standard suppression coil L1.





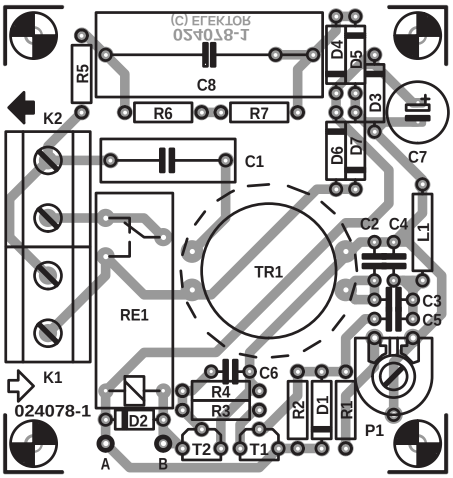
Gain for operating the relay is provided by T1. The amplified signal is smoothed by C6 and provides the voltage necessary to cause T2 to conduct and energise the relay. The voltage divider formed by P1, R1 and R2 provides a bias voltage for T1 in order to increase the sensitivity of the receiver. This also allows the relay to be energised without a received signal. D1 ensures that C5 does not become charged and prevents T1 from conducting even more. The operation of the circuit is based on the fact that the incoming signal is sufficiently strong to overcome the hysteresis of the relay. Once the signal is no longer present, the relay must naturally again release.

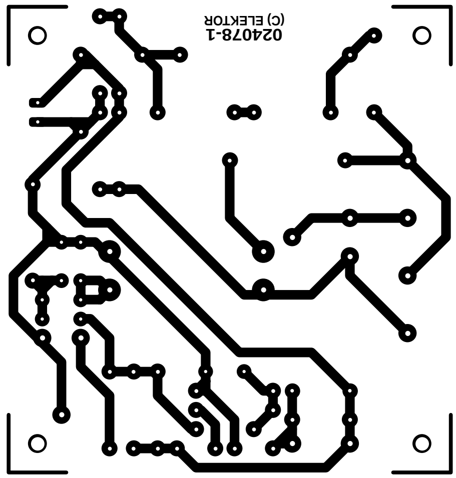
To be honest, it must be noted that the simple design of this circuit has the disadvantage that its sensitivity may be somewhat inadequate, depending on household circumstances. One possible solution is to reduce the frequency of the transmitter to the region between 95 and 125 kHz. The values of C1, C2 and C4 will then have to be modified to match, so this is something for readers who like to experiment.

Do not forget that just as with the transmitter, the entire circuit (once it has been switched on, of course) is connected to the mains potential. Power for the transistor stage and the relay is taken directly from the mains voltage using a capacitive voltage divider; R5 is only necessary to limit the current through the diodes to a safe value on switch-on. Rectification is provided by diodes D4–D7 and filtering by C7. The impedance of C8 is low enough to provide sufficient current. The no-load voltage (when T2 is not conducting and the relay is not activated) is limited by zener diode D3. R6 and R7 discharge C8 immediately after the circuit is disconnected from the mains, in order to prevent any dangerous voltage from remaining on the input terminals.

Connections A and B are provided for test purposes and also allow something other than the relay to be energised (but keep in mind that the circuit is electrically connected to the mains network!). The pinout of the relay is standard, so a type other than that shown in the components list can also be used, as long as you make sure that the operating voltage is 24 V and the operating current does not exceed 28 mA.

(024078-1)





COMPONENTS LIST

Resistors:
R1 = 1MΩ5
R2 = 220kΩ
R3 = 39kΩ
R4 = 6kΩ8
R5 = 220Ω
R6,R7 = 470kΩ
P1 = 10MΩ preset

Capacitors:
C1 = 22nF 275VAC Class X2, lead pitch 15mm
C2 = 22nF, lead pitch 5 mm
C3 = 220pF
C4 = 2nF2, lead pitch 5mm
C5 = 680pF
C6 = 100nF, lead pitch 5 mm
C7 = 100µF 40V radial
C8 = 330nF 275VAC, Class X2, lead pitch 22.5mm or 27.5mm

Inductors:
L1 = 470µH

Semiconductors:
D1 = BAT85
D2 = 1N4148
D3 = zener diode 24V 1.3W
D4-D7 = 1N4007
T1 = BC557B
T2 = BC547B

Miscellaneous:
K1,K2 = 2-way PCB header, lead pitch 7.5 mm
Tr1 = 5:5 turns 1mm dia. isolated wire on N30 ring core 16x6.3 mm, B64290L45X830 EPCOS (Farnell # 311-0266)
Re1 = PCB relay, 1 c/o contact, 8A 24V 1200Ω, e.g., Schrack # V23057-B0006-A201

Modem Line Protection

011

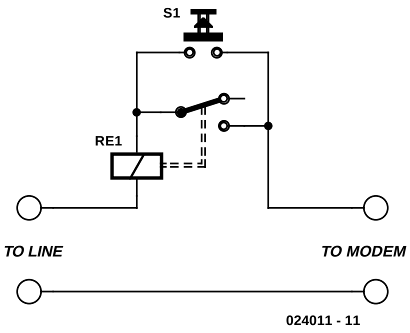
V. Steensgaard

We haven't yet experienced this first-hand, but it seems to occur more often that when some websites are visited they switch the Internet connection over to a premium rate dial-up number. This is of course very irritating, because it is hardly noticeable and it certainly costs a lot more.

This circuit is simplicity itself, but still offers effective protection against these practises. The circuit consists of nothing more than a push-to-make switch and a reed relay with a home-wound coil and is connected to the phone line in series with the modem. During the dial-up the switch has to be pushed down for a while (a bit earlier really, or otherwise the modem won't be able to detect the dial tone). Once the modem is 'off hook', the pushbutton can be released. The current in the phone line then keeps the reed relay energised, maintaining the connection.

When an attempt is made to switch to a different number, the line has to be dropped first, no matter how quickly this happens. The reed relay opens at any interruption of the current, stopping these tricks in their tracks. No matter how clever the programmer is at the other side, this simple hardware protection cannot possibly be circumvented!

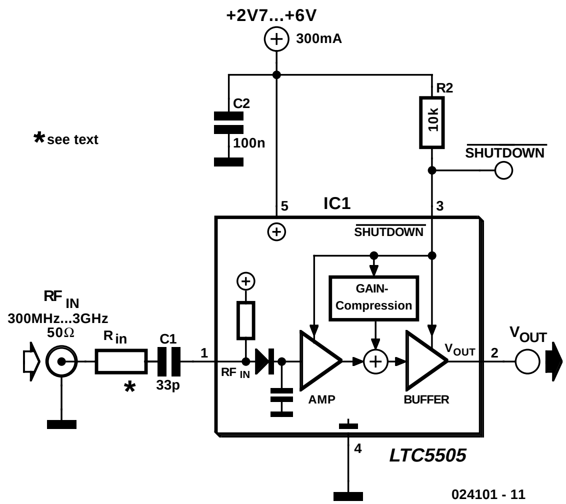
A guideline for the construction of the reed relay coil is to wind about 200 turns of 0.1 mm or 0.2 mm copper enamelled wire round the relay. The complete reed relay can also be bought

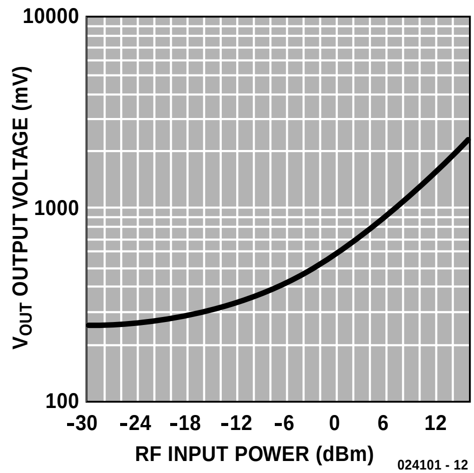


from the author via his website: http://home.worldonline.dk/~wildsto/sdb/. For completeness we should mention that a patent has been applied for.

(024011-1)

300-3000 MHz RF Detector 012





An RF detector with a dynamic range of 40 dB can now be obtained from Linear Technology (http://www.linear-tech.com/pdf/5505i.pdf). The type LTC5505, housed in an SOT23 SMD package, can handle input frequencies between 300 MHz and 3 GHz for input signal levels between –32 dBm and +18 dBm (0 dBm = 1 mW into 50 Ω). There are two versions having different input level ranges, as shown in the table.

Version	Input power range	R_{in}
LTC5505-1	–28 dBm to +18 dBm	20 Ω
LTC5505-2	–32 dBm to +12 dBm	0 Ω

The LTC5505-1 is intended to be used for the upper range of signal levels. A series input resistor (R_{in}) in combination with the internal input resistance attenuates the input signal. For both versions, the input impedance is approximately 50 Ω.

The LTC5505 contains a Schottky diode used as a detector, which is temperature compensated by additional circuitry. The IC requires an operating current of only 0.5 mA at a supply voltage between +2.7 V and +6 V. An active-Low shutdown input can be used to disable the detector. In the disabled state, the IC draws less than 2 μA.

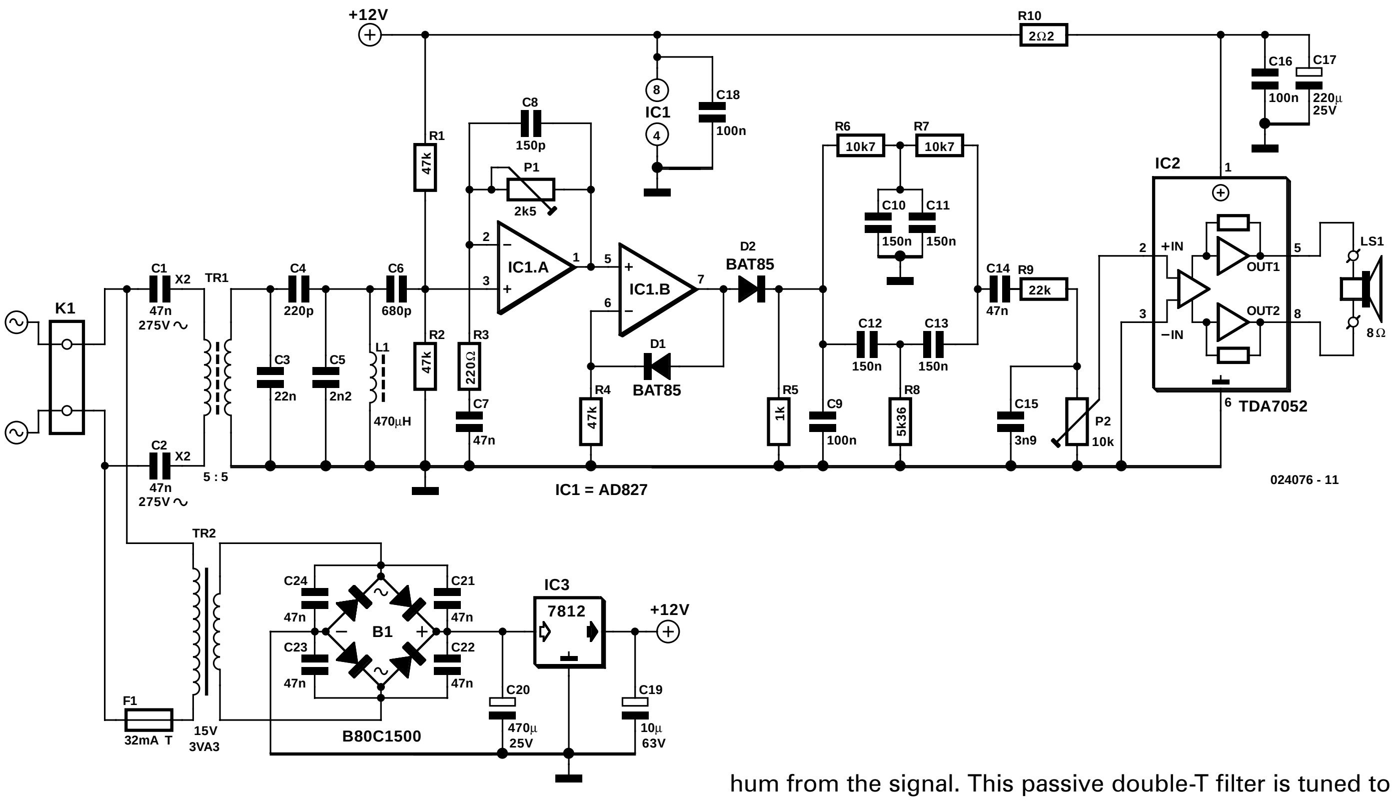
The output voltage of the detector ranges from +280 mV to more than +2 V, depending on the input signal level, with a load current capacity of around 1 mA. A gain compression block reduces the output level at high RF levels in order to keep the output signal within the stated range, whose upper limit is set by the minimum supply voltage of +2.7 V.

Using this detector IC, simple diode detectors can be replaced by a component of the same size having significantly better characteristics.

(024101-1)

AM Demodulator for Intercom

013



This circuit should be considered as more of an experimental circuit for AM demodulation, than as a practical application. In the associated 'AM modulator' we have raised the problems caused by mains hum, getting in the way of interference-free operation of the AM mains intercom. When the transmitter and receiver are not coupled through the mains then the quality is perfectly adequate.

The 'receiver' used here is the same as found in the 'mains remote switch' and 'mains remote control decoder' (C1 to C5/TR1/L1). The capacitor that connects the small toroidal transformer to the mains has been split into two, making the circuit a little bit safer. But please note: **the complete circuit should be considered as being at mains potential.** So don't solder to the circuit or take any measurements while it is switched on.

The input signal is first amplified to the right level by a fast opamp (AD827). P1 can be adjusted to give this stage a maximum gain of 20 dB. The actual demodulator is about as simple as you can get, since it consists of nothing more than a diode, a capacitor and a resistor (D2/R5/C9). Due to the RC time constant and the diode the voltage across the capacitor follows the envelope of the AM carrier wave. The circuit of IC1b, D1 and R4 make the characteristics of diode D2 somewhat more linear. This effect is fairly small, so if simplicity of the circuit is important you could leave this part out.

The filter following this stage attempts to remove the worst hum from the signal. This passive double-T filter is tuned to 100 Hz because that was the strongest component of the interference. But in practice all harmonics of 50 Hz are present, too many to suppress with a simple filter. Because the sensitivity of the power amplifier used here is fairly high, the double-T filter is followed by a potential divider and first order high-pass and low-pass filters (C14/R9/P2/C15). This keeps as much of the speech signal as possible, while removing more of the interference.

The power amplifier is a TDA7052 (IC2) that is meant for 6 V operation but which can also be used at 12 V. Take care that it doesn't consume too much power (if you can get hold of one, a 16 Ω speaker is better). The amplification is approximately 40 dB. IC2 has been well decoupled from the supply by R10/C16/C17. Constructors have to make sure that both the ground and the +12 V for the power amplifier are taken directly from the regulator and are not used to supply any other part of the circuit. Preset P2 is used to set the volume level, although a 'real' (logarithmic) potentiometer could also be used.

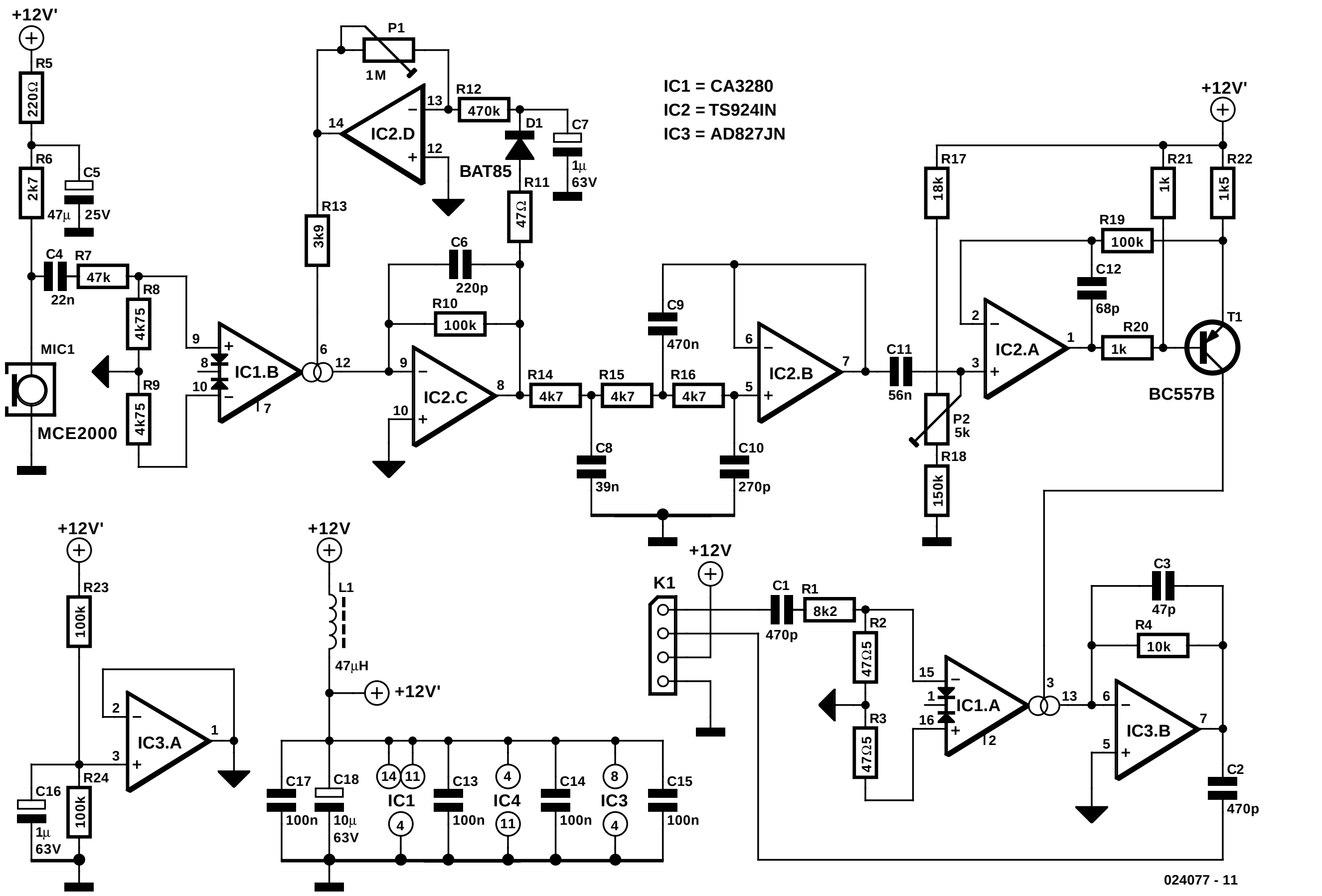
The power supply uses a standard circuit: a bridge rectifier (B1) with suppression capacitors (C21 to 24), a smoothing capacitor (C20), a 7812 regulator (IC3) and a final decoupling capacitor.

The quality of the circuit could be improved by replacing the passive filter with higher order active filters (possibly switched-capacitor types), but this is clearly only something to try for dedicated experimenters.

(024076-1)

AM modulator for Intercom

014



This circuit was originally designed as a simple mains intercom for use in the home. To complete the circuit you could use the 'Mains Remote Transmitter'. We have to admit that our tests with the mains intercom where somewhat disappointing due to a persistent audible mains hum from the speaker. That doesn't diminish the usefulness of the AM modulator as such, and it is still a very useful circuit for experimentation with this type of modulation. The accompanying receiver is called 'AM demodulator for Intercom'.

The circuit consists of a microphone amplifier with a presettable automatic gain control (IC1b/IC2c/IC2d), a speech filter (IC2b) and the actual modulator (IC1a/IC3b/IC2a/T1). Because an asymmetrical power supply is used, the circuitry round IC3a has been added to provide a virtual ground which has its potential at half the supply voltage.

The hart of the circuit is formed by a dual OTA (IC1, a dual operational transconductance amplifier), of which one is used for the microphone amplifier and the other for the AM modulator. It would be too much to give a detailed description of the operation of an OTA; it suffices to give a brief explanation of the various parts.

The potential divider (R7/R8) at the input of IC1b protects it against excessive inputs. The current output is converted into a voltage by buffer stage IC2c. The level of transconductance of IC1b is controlled by the bias input (pin 6). The current fed to this pin (I_{ABC}, amplifier bias current) is limited by R13 to a maximum of 1.5 mA. The peak level of the output from IC2c is rectified by D1 and C7 and fed back as a control current to the OTA via inverting buffer IC2c. When the output voltage of IC2c increases, so will the voltage across C7, resulting in a smaller bias current and a reduction in the amplification of the microphone signal. This effect is most pronounced with preset P1 at its maximum setting. As P1 is turned down the amplification level of the microphone amplifier becomes more constant. With P1 fully closed the amplification is a constant 38 dB. With P1 at its maximum, the gain is automatically varied by up to 30 dB. P1 can therefore be used to set up the microphone amplifier as required.

R6 is used to bias the electret microphone (in this case a MCE2000 from Monacor). R5 and C5 decouple the supply to the microphone. Since the bandwidth of the microphone is much greater than that available in the 'Mains Remote Transmitter', a 3rd order Chebyshev filter (IC2b) has been added directly after the microphone amplifier, which has 3 dB ripple and a bandwidth of just 3.15 kHz.

The signal is then fed to current source T1/IC2a. The circuit round T1 functions as a current source that can be modulated: a 'constant' current that varies linearly according to the processed microphone signal. This current is then used as bias current for OTA IC1a, causing the signal that is fed to pin 1 of K1 to appear at the output of IC3b with its amplitude modulated. IC2a compares the voltage across emitter resistor R22 with its input, causing the current through T1 to vary linearly with the voltage at pin 3. R19 and C12 are added for stability and potential divider R20/R21 stops the output of IC2a from clipping. The maximum bias current is about 3.5 mA.

The amplification of modulator IC1a/IC3b has purposely been kept a bit below 1 (it can be varied with P2 between 0.5 and 0.6), since 100% modulation will cause the maximum amplitude to be equal to the input voltage thereby overdriving the transmitter. K1 has the same pin-out as the connector on the transmitter board; the supply is taken from pins 3 and 4. The total current consumption is about 25 mA. The output signal from IC3b is connected to pin 2 of K1. The circuit was designed for use in close proximity with the transmitter, so no limiting resistor was added to the output. When a longer (shielded) cable is used between the two, you should connect at least a 47 Ω resistor in series with the output. A fast AD827 was chosen for opamp IC3, so that the modulator can easily cope with the 143 kHz signal.

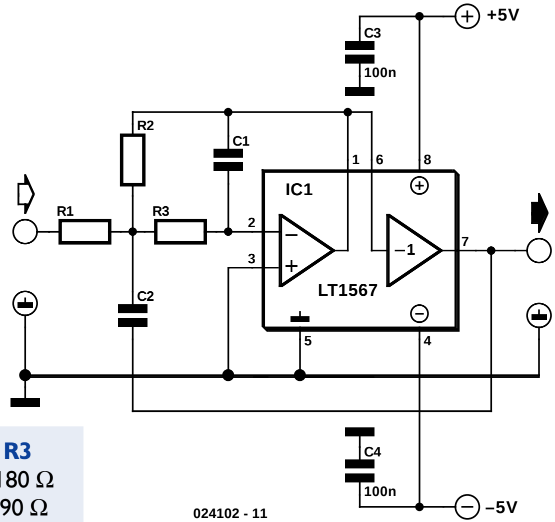
(024077)

5 MHz Active Lowpass Filter

015

In the LT1567 Linear Technology (www.linear-tech.com/pdf/1567i.pdf) has produced a component specially developed for building analogue filters with cutoff frequencies up to 5 MHz. It contains two wide-bandwidth operational amplifiers, the second of which has a fixed configuration as an inverting amplifier with unity gain. To construct a lowpass filter just two external capacitors and three resistors are required.
Example component values for corner frequencies of 1 MHz, 2 MHz, and 5 MHz are shown in the table.

(024102-1)



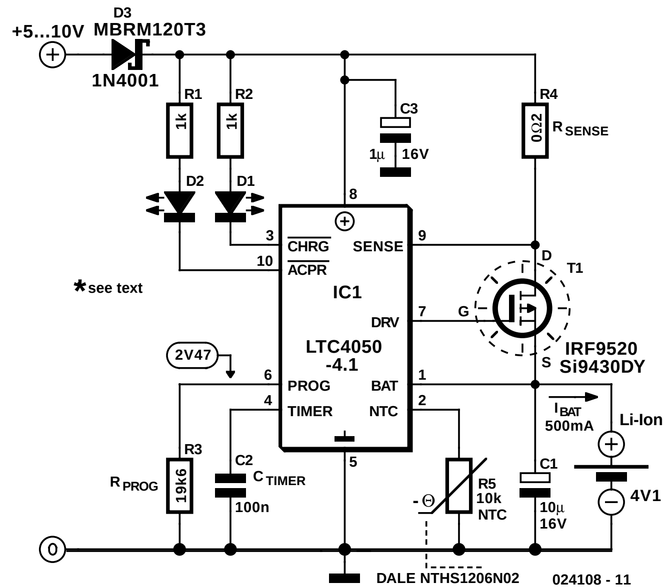
Filter	C1	C2	R1, R2	R3
Chebyschev 0.1 dB ripple, 1 MHz	120 pF	180 pF	1050 Ω	1180 Ω
Chebyschev 0.1 dB ripple, 2 MHz	120 pF	180 pF	523 Ω	590 Ω
Chebyschev 0.1 dB ripple, 5 MHz	120 pF	180 pF	205 Ω	232 Ω
Butterworth, 2 MHz	180 pF	180 pF	604 Ω	309Ω

Lithium-Ion Charger

016

Lithium-Ion cells require a totally different charging protocol to that for NiCd or NiMH cells, a protocol that has to be followed precisely. During the last year we have already published two articles regarding the charging of this type of cell. This time we're using a new IC (so it may still be difficult to obtain!) made by Linear Technology (www.linear.com), which is very small and can therefore be built into the cell permanently, but is also suitable for use as an 'ordinary' charger.

It is designed to charge one cell at a time, at a current of 500 mA. When a new cell is connected and power is applied (in any order), the charging process begins. First the temperature of the cell is checked with the help of the NTC. The charging will only start if the temperature is between 0 and 50 °C. When Lithium-Ion cells have been discharged too deeply they should at first be charged very gently, at a current of only 50 mA, as long as the cell voltage is below 2.49 V.



Above that voltage the charge current is increased to a nominal 500 mA, until the maximum voltage of 4.1 V (or 4.2 V, depending on the type) has been reached. The cell voltage is now held at this level, causing the charge current to gradually decrease until the cell is fully charged. When the charge current has reduced to 50 mA, the charging stops and the cycle is complete. As an extra safety measure the IC also contains a timer that stops the charging process after a specific time, even if the current hasn't yet fallen below 50 mA.

The phases described above are indicated by LED D1. During the charging of the cell it will light up brightly. When the charging stops due to the current having fallen below 50 mA, it is lit dimly. And when the timer stops the charging process, the LED will be off.

When the charging process has completed, the supply is obviously no longer required. The charger circuit itself can be left connected to the cell since it only draws about 5 to 7 µA, so there is no need to worry that the charger would quickly discharge the cell. A new charge cycle will begin when an empty cell is connected and power is applied. A new cycle will also automatically start (as long as power is applied) when the cell voltage drops below 3.88 V (3.98 V).

The charge current can be modified by adjusting R3 and R4 according to the following formula: $I = (2.47/R3) \times (800/R4)$. The maximum charge time is determined by C2; the formula used here is: $\text{time} = (C2 \times 3\ \text{hours}) / 0.1\ \mu F$. The timer doesn't start until the cell voltage reaches 4 V. LED D2 is lit when the voltage supplied to the charger is high enough.

T1 is a P-channel MOSFET, which can be virtually any power type. It could even be replaced by a PNP darlington, with its emitter connected to R4. NTC R5 should be mounted as closely as possible to the cell, so that the cell temperature is measured accurately. It won't be easy to find the NTC used in this circuit, but the accuracy of the 0 en 50°C temperature limits aren't that important. Since its resistance at 25°C is 10 k, it could even be replaced by a fixed 10 k resistor. Obviously the temperature protection will then no longer function.

For D1 and D2 you should use low-current (also known as high efficiency) LEDs. D3 can be any 1 A Schottky diode, or an ordinary diode such as the 1N4001 if it doesn't matter that there is a slightly bigger voltage drop.

There is one final point, which most of you probably know: Lithium-Ion cells may absolutely never be charged at voltages greater than 4.1 V (4.2 V) because they could explode under those circumstances. It should be stated on the cell whether it is a 4.1 V or 4.2 V type, otherwise you will have to refer to information provided by the manufacturer. The LTC4050 comes in two versions, with '-4.1' or '-4.2' as a suffix. The IC is only available in a SMD package (MS10).

(024108-1)

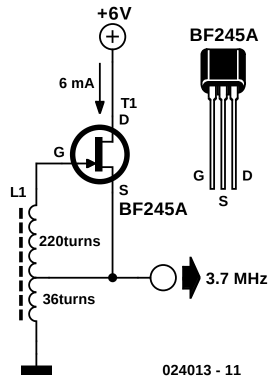
2-component Hartley Oscillator 017

G. Baars

Although Elektor never actually launched a design contest under the name “low component-count oscillators” the author was challenged by ‘Three Component Oscillator’ published in July/August 2001. The result is shown here, representing a reduction in component count of no less than 33.3%! The audio field has been left though in favour of RF.

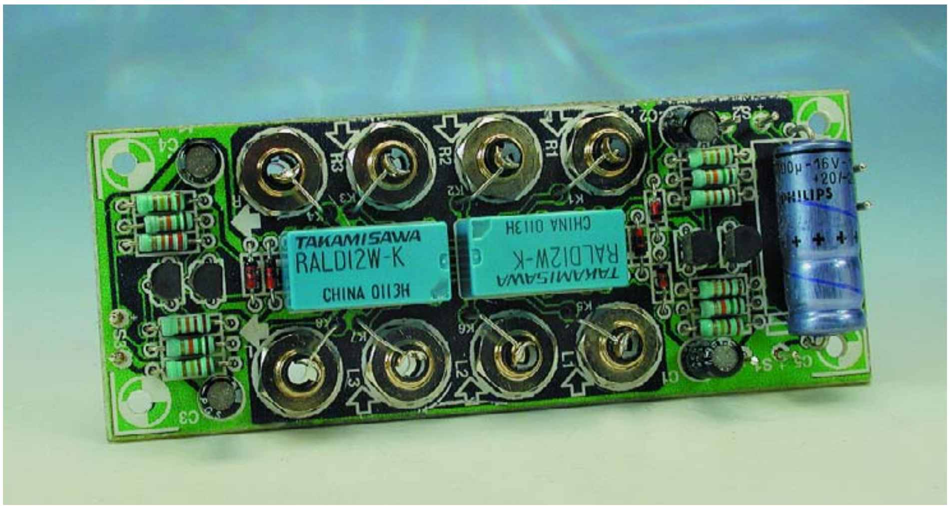
This Hartley oscillator can be built from just one FET and a coil. The coil has a tap to provide the amount of positive feedback the circuit needs to start and maintain oscillation. The stray capacitances presented by the FET gate and the coil wires are enough to make the circuit resonate at about 3.7 MHz with the coil data given in the diagram. The internal diameter of the coil is about 8 mm and no core was used. Moving the tap up towards the gate will reduce distortion but at some point the oscillator will throw in the towel and refuse to start.

(024013-1)



Audio Switchbox

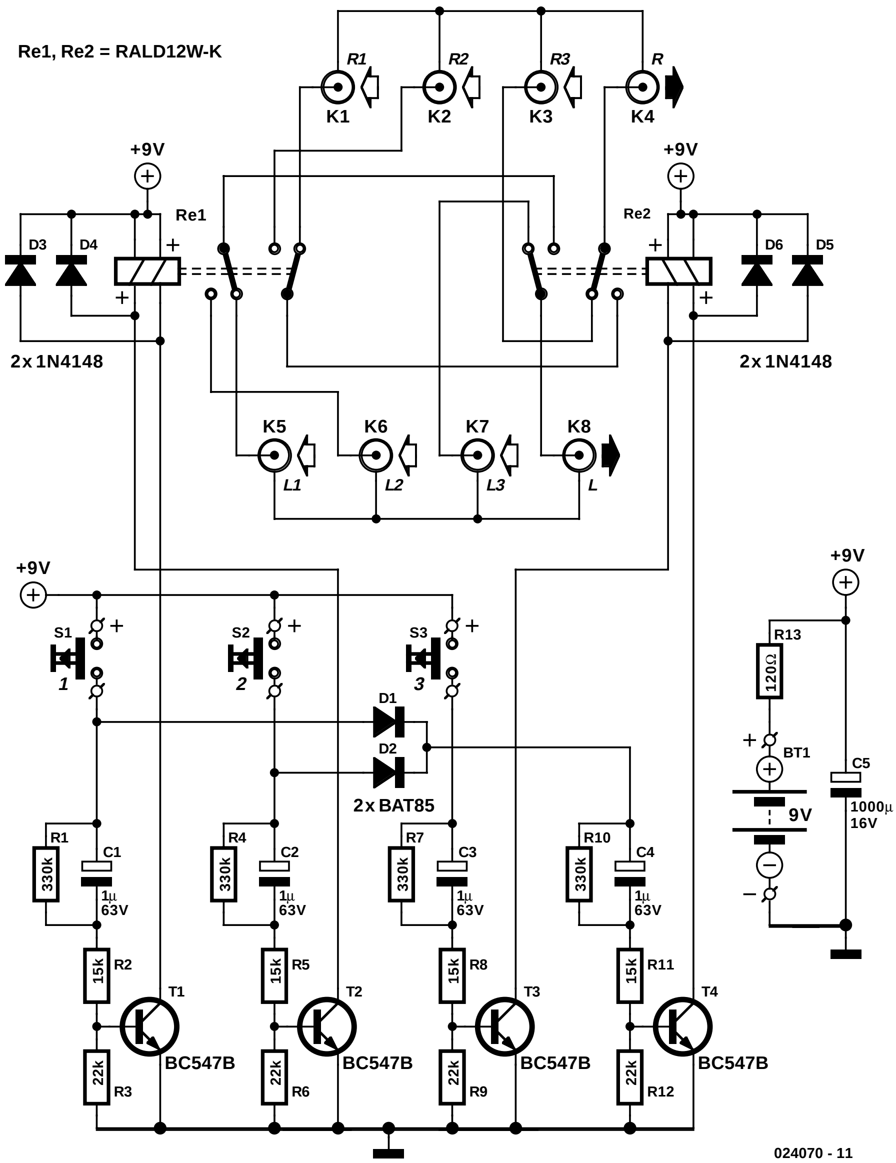
018

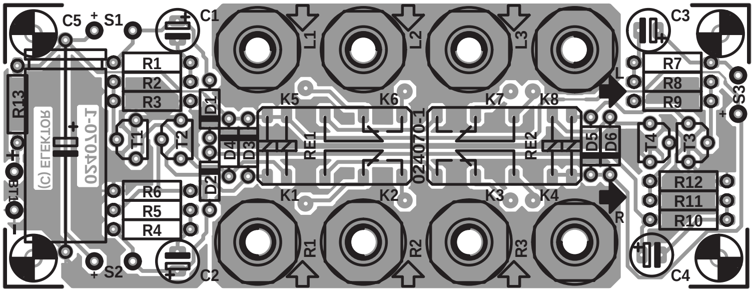


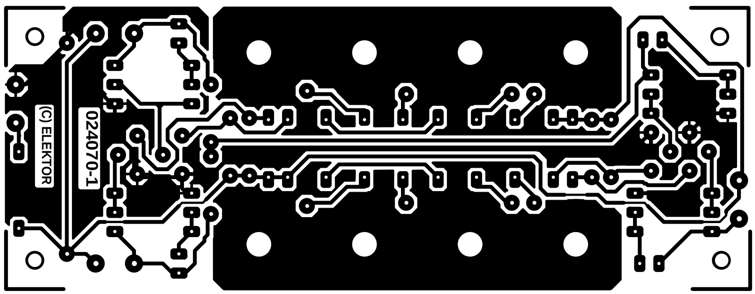
This circuit is intended as an extension for (pre-) amplifiers, to increase the number of inputs. Ever since the introduction of MD-recorders, DVD players, etc. the owners of older amplifiers have had a chronic shortage of inputs. The application of this switchbox also makes it possible to loop the audio outputs from the DVD player and video recorder to the audio system, without the need to turn on the TV. This is very handy when the audio installation is positioned some distance away from the video system and you only wish to listen to the sound from the DVD/MP3 player, for instance.

The circuit makes use of two bistable relays, which have two changeover contacts each. This makes the circuit nice and compact and also avoids the need of exerting considerable force on the shaft of a rotary switch. The relays can now be operated with three small buttons (S1 to S3). The relays are 12-V models, which operate just fine on a 9-V battery. In order to reduce the power consumption to virtually zero, a number of differentiator networks (C1/R2, C2/R5, C3.R8 and C4/R11) have been designed that generate the necessary pulses for the relay windings. Every relay has a SET and RESET winding.

The third stereo input is directly connected, via the normally-closed contacts of relay Re2, to the output. The other two inputs are connected via Re1 to the normally-open contacts of Re2, and from there to the output. To select the third input only Re2 has to be reset. That is the purpose of the small circuit around T3. When pressing S3, T3 is made to conduct for a sufficiently long period (several milliseconds) via C3/R8/R9, to ensure that the relay switches over. R7 is required to discharge C3 quickly when the pushbutton is released. The differentiator networks have the advantage that even when the pushbutton remains activated the current consumption is still very low (<25 µA). Only the initial







COMPONENTS LIST

Resistors:
R1,R4,R7,R10 = 330kΩ
R2,R5,R8,R11 = 15kΩ
R3,R6,R9,R12 = 22kΩ
R13 = 120Ω

Capacitors:
C1-C4 = 1µF 63V radial
C5 = 1000µF 16V axial

Semiconductors:
D1,D2 = BAT85
D3-D6 = 1N4148
T1-T4 = BC547B

Miscellaneous:
K1-K8 = cinch (RCA) socket, chassis mount
S1,S2,S3 = pushbutton
Re1,Re2 = RALD12W-K (12 V/960 Ω) Takamisawa (Conrad # 50 33 98-60)
BT1 = 9V battery with clip

pulse amounts to a little over 0.5 mA. We can therefore expect the battery to last several years (beware of possible leaks).
If the first or second input is desired then Re1 has to be set or reset via S2/T2 and S1/T1 respectively. Re2 has to be set when S1 or S2 is pressed. The fourth differentiator network takes care of this. When S1 or S2 is pressed, D1/D2 and the small circuit around T4 generate a set pulse for Re2, so that Re1 will be selected.
A 1000-µF electrolytic capacitor in parallel with the battery acts as an 'emergency power supply' when the battery is nearly exhausted. Every pulse at 9 V amounts to nearly 10 mA. The relays are industry-standard types and pin-compatible with the common V23042-B2203-B101 from Siemens (called Schrack these days). R13 limits the 'short-circuit current' (through C5) when the battery is first connected.
The connections for the three pushbuttons appear on the PCB in the locations where they were most convenient, in order to keep the PCB as small as possible. Whoever is tempted to use a slide switch has to keep in mind that in that case there is always one closed contact and a small current flow as a consequence. To facilitate testing, the power supply (marked with '+') is available next to each connection. If the three pushbuttons are mounted on a panel a single common '+' will suffice of course. The PCB shown here is unfortunately not available ready-made.

(024070-1)

Fan Monitor

019

The MAX6684 from Maxim (http://pdfserv.maxim-ic.com/arpdf/MAX6684.pdf) is a device for monitoring fans. Available in an 8-pin small outline SMD package the IC is capable of detecting if the fan is jammed or if it is turning too slowly. Although the device itself requires a supply voltage in the range +3.3 V to +5 V, it can be connected to fans operating on up to +24 V and drawing currents of up to 250 mA. An internal sense resistor to ground (PGND) is used to detect the current pulses of the fan motor and the pulses are processed according to their shape and frequency. A power MOSFET, with an on resistance of around 1 Ω, is connected between SENSE and PGND.

The open drain $\overline{\text{FAIL}}$ signal goes low to indicate a fail condition when:

(a) the current drawn by the fan falls below 35 mA_{pp} (AC component);

(b) the current drawn by the fan rises above 600 mA;

(c) the fan rotation rate falls below about 700 revolutions per minute (25 Hz AC component);

(d) the die temperature of the chip rises above +160 °C (over-temperature protection).

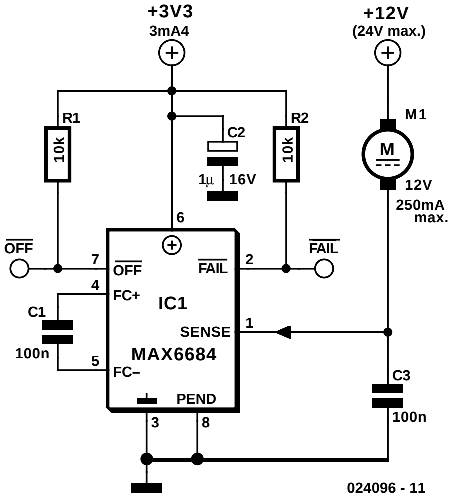
In the first case the fan remains switched on, whereas in case

(b) the device will try to start the fan again every 62 ms. If the over-current condition persists, the MAX6684 will turn off again within 2 ms and wait another 60 ms. In case (c) also the fan remains switched on. The lower rotation rate threshold for this case is mainly determined by the coupling capacitor between FC+ and FC-. The value should be determined experimentally if necessary.

If the fan already has internal protection against jamming it will automatically switch itself off. Then the $\overline{\text{FAIL}}$ signal is only active during the period when the fan tries to restart itself. If the fan rotation rate threshold detector does not work reliably, a 100 µF electrolytic capacitor connected in parallel with the fan may help.

The $\overline{\text{OFF}}$ input can be used to turn the fan on and off.

(024096-1)



Overvoltage Protection

020

W. v.d. Voet

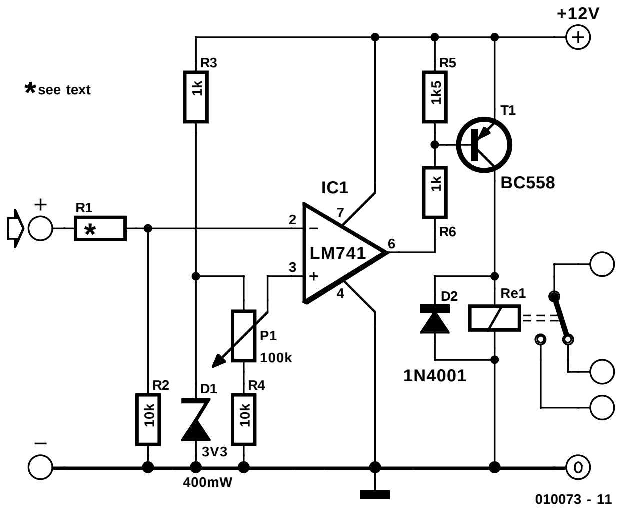
This is a tried and trusted design for the protection against overvoltages that can be configured for your own needs. The circuit can also be used to detect undervoltages; in that case the inputs of IC1 have to be changed over.

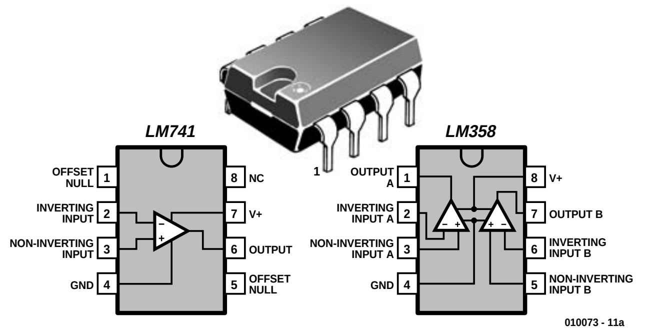
The operation is straightforward. When the input voltage becomes too large, the voltage at pin 2 (inverting input) of IC1 becomes higher than the reference voltage at pin 3 (non-inverting input). The output of the opamp will in that case become 'low'. (It's just like a maths lesson: *higher* times *inverting* can be thought of as + times – and that results in –.) This then drives T1 on, which in turn supplies power to the relay. The contacts of the relay can then be used to isolate the equipment from the supply.

The circuit doesn't have any hysteresis, so in principle there is a possibility that the relay could chatter. In practice it won't be that bad, because the voltage will rise a bit further when the protected equipment is turned off.

We've chosen a supply voltage of 12 V for the circuit, but in practice any voltage between 12 V and 24 V is suitable. The coil of the relay should work at the chosen supply voltage; so select a relay with a working voltage the same as that of the supply, or if you already have a relay, make the supply voltage equal to the operating voltage of the relay. It doesn't have to be exact, as long as you stay within plus or minus 10%. Keep in mind that a BC558 (T1) can switch at most 50 mA. Should the relay require more current, replace T1 with a BC516; this can switch a maximum of 0.5 A (in practice 0.25 A).

The voltage reference is simply derived from a zener diode (D1). The zener voltage isn't critical and a value of 5.1 V may even be better than the 3.3 V used here, because the change





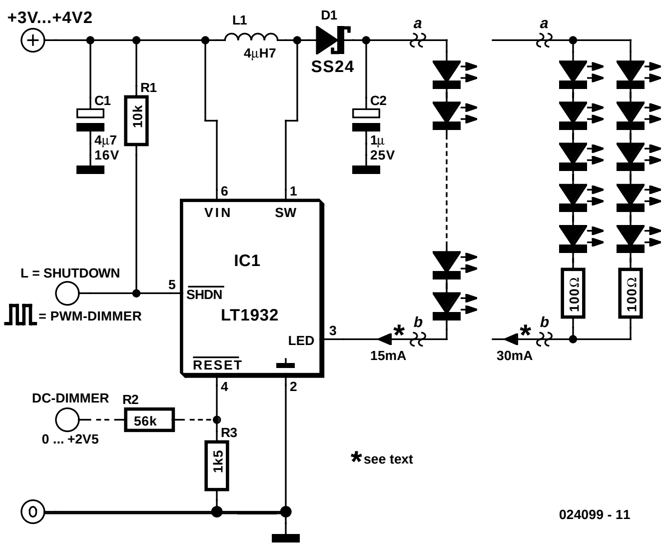
in zener voltage due to a temperature change is then at a minimum. R3 should have a value that lets a current of at least 3 mA flow through the zener. Keep in mind that according to the datasheet for the 741 the input voltage at which the 741

switches should be at least 1.5 V higher than the voltage at pin 4 (this is the so-called common-mode voltage). So the voltage set by P1 at pin 3 should not be less than 1.5 V. In practice a lower voltage is possible, down to about 1 V. It would be better to use a value between 47 kΩ and 100 kΩ for R4; this reduces the adjustment range somewhat, but at least the voltage can never be set too low. If you really want the lower range to extend down to 0 V, then you should choose a different opamp, such as half a LM358.
Resistor R1 forms a potential divider in combination with R2. R1 should have a value such that the voltage at pin 2 is about equal to the voltage at the wiper of P1 when that is in its mid position; that is about 2.4 V. The following formula is used to calculate the value for R1: it is equal to the required turn-off voltage minus 2.4 V, divided by 240 µA. So for protection against voltages greater than 100 V, R1 should be 407 kΩ; in practice you would use 390 kΩ.
The current consumption of the circuit is only a few mA plus the relay current.

(010073)

LED Flashlight

021



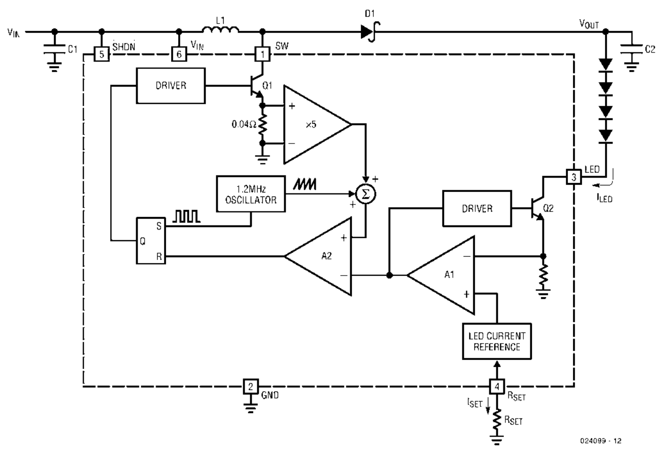
If you need to drive a number of white LEDs to provide display backlighting or for a white-light torch what's the best way to configure them? If they are wired in series then the forward conduction voltage of the chain will be greater than the output from a typical battery. Connect them in parallel and current becomes a problem. To keep the light output constant it is also necessary to maintain a constant current through the LEDs despite a falling battery voltage.

The LT1932 from Linear Technology (www.linear-tech.com/pdf/1932f.pdf) is an efficient solution to the problem. This chip operates at a low supply voltage and contains a switched mode regulator circuit supplying an output current defined by an external resistor at the R_{SET} input. The 4.7 µH inductor must be a type suitable for use in switched mode circuits; it stores energy in the ferrite cores magnetic field during switching.

Up to eight LEDs can be connected in series but if more are required it is possible to use two parallel networks of five series connected LEDs. In this configuration it is necessary to include a resistor in each arm of around 100 Ω to prevent one of the arms from hogging all the current.

A pulse width modulated square wave applied to the Shutdown ($\overline{SHDN}$) input will dim the LEDs. Alternatively an external variable supply connected via a 56 kΩ resistor to the R_{SET} input can also act as a dimmer.

(024099-1)



Number of LEDs	Operating voltage	Efficiency	R_{SET}	I_{LED}
2	1.8 to 3.0 V	75 %	4kΩ53	5 mA
3	1.8 to 3.0 V	75 %	2kΩ26	10 mA
4	1.8 to 3.0 V	75 %	1kΩ5	15 mA
5	2.0 to 3.0 V	70 %	1kΩ13	20 mA
6	2.7 to 4.2 V	75 %	750 Ω	30 mA
8	3.0 to 4.2 V	70 %	562 Ω	40 mA
10 *	2.7 to 4.2 V	75 %		

* *Two parallel arms of 5 LEDs connected in series together with a 100 Ω resistor*

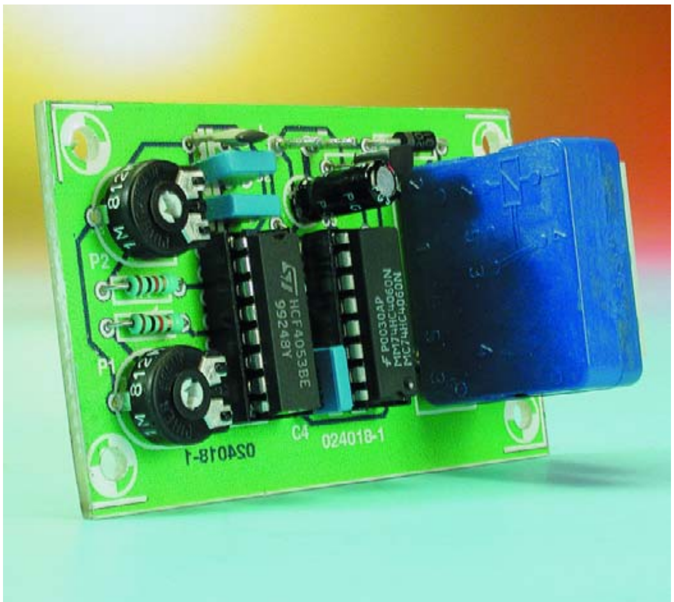
On/Off Timer

022

If you need an adjustable 'on' or 'off' time for some application, then this is the circuit you have been looking for. A problem that often occurs when adjusting timers is that the individual times affect each other. This circuit completely solves this problem because the time defining elements — both R and C — are switched over. That means that there is an RC pair (P1 + R3 and C1) for the 'off' time and another pair (P2 + R4 and C2) for the 'on' time.

The relay is not energised when there is a logical zero at the base of T1. This same zero causes, via input pins 10 and 11 of IC1, pin 12 to be connected to pin 14 and pin 2 to pin 15 of IC1. By contrast, a logical one (relay energised) causes pin 13 to be connected to pin 14 and pin 1 to pin 15.

With the values shown, the oscillator period (this can be measured at pin 9 of IC2) can be adjusted from 4 to 200 ms. Since IC2 divides the frequency by 8,192 the resulting time period is adjustable from 32.8 seconds to 27.3 minutes. If a shorter period of time is desired C1 (or C2) has to be reduced,

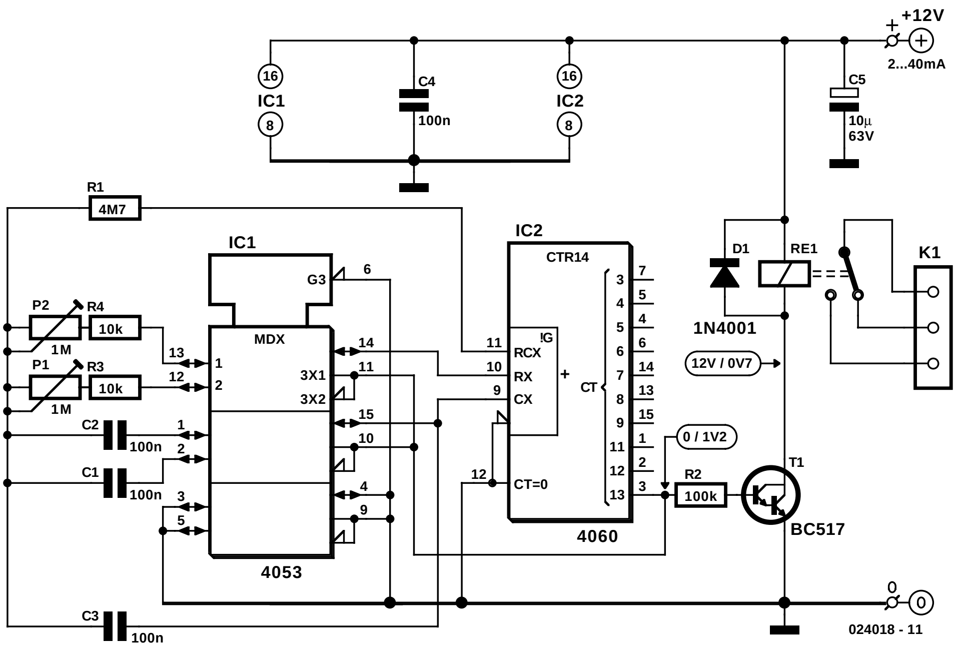


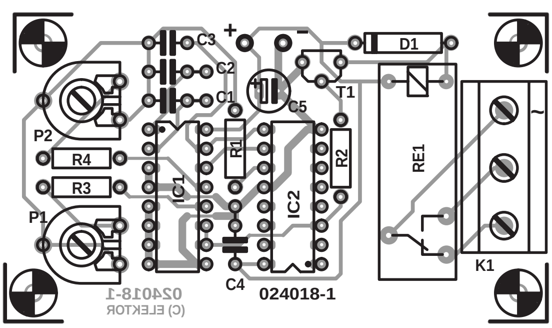
COMPONENTS LIST

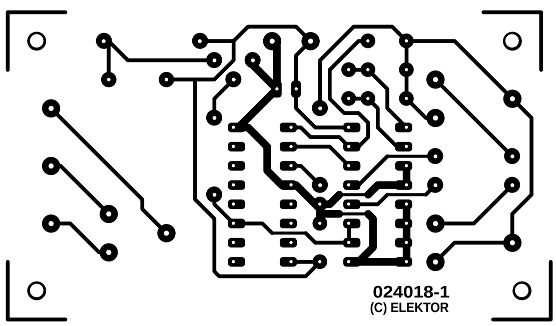
Resistors:
R1 = 4MΩ7
R2 = 100kΩ
R3,R4 = 10kΩ
P1,P2 = 1MΩ preset

Capacitors:
C1,C2,C4 = 100nF
C3 = 100pF
C5 = 10µF 25V radial

Miscellaneous:
D1 = 1N4001
T1 = BC517 (Darlington)
IC1 = 4053
IC2 = 4060
Re1 = E-card relay, 12V coil,
V23057 B0002 A201 (Schrack)







increase for a longer period. C1 and C2 need to be film capacitors or bipolar electrolytics; if these are not available it is possible to make one yourself by connecting two ordinary electrolytics in series, with the positive terminals together.
The power supply voltage for the timer may range from 5 to 15 V. It is preferable that you choose the same value as the rated operating voltage of the relay. The relay shown in the parts list is a 12-V type that is able to switch 230 VAC at several amps. The PCB for this project is unfortunately not available ready-made.

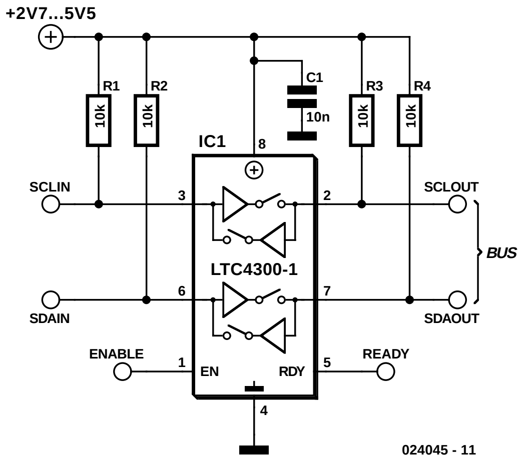
(024018-1)

I²C Hot Swap

Now that we've become used to USB you really appreciate the facility to connect or disconnect devices without having to turn the power off first. This was also true for RS232 (but not with LPT), but you never felt at ease with it. I²C or SMBus devices are unfortunately not hot swappable.

A component that solves half of the problem (the switching part) has been introduced recently by Linear Technology. The LTC4300 is a 2-wire interface buffer that can isolate the signals between peripherals and the bus, making it possible to add another device to the bus at any time, without causing any interference. The next step is more difficult and you will have to find the solution for this yourself: you have to find a way to detect when there is no activity on the bus. At that time the interface chip can be enabled, causing the peripheral to be connected to the bus.

The buffer contains active pull-ups, permitting the use of high-value (10 kΩ) pull up resistors.



More information about the LTC4300 can be found at www.linear.com.

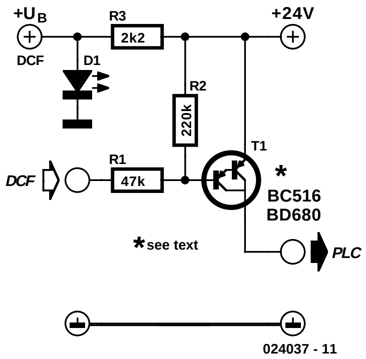
(024045-1)

Level Shifter

024

R. Koegler

In order to connect the open-collector output of a standard Conrad DCF receiver module (order number 121177), which has a relatively low operating voltage of 1.5–15 V and an NPN open-collector output, to a PLC requiring a positive-switching 24-V signal at around 150 mA, level shifting is necessary. This can be achieved in a simple manner using a miniature relay connected between the positive supply voltage for the module and the DCF module output, as shown in **Figure 1**. In this case the current load on the DCF output must not exceed 1 mA.
All that is needed for this solution is a 15-V miniature relay with a suitable coil voltage that does not require the DCF output to sink more than 1 mA. There are no special requirements on the PLC side, since the contacts of just about any relay can switch 24 VDC at 150 mA.
Besides the electromechanical solution, there is also an electronic solution. Two resistors and a transistor are all you need, as shown in **Figure 2**, if you are prepared to do without electrical isolation between the two devices.
Any desired type of Darlington transistor without a built-in resistor can be used, as well as a p-channel MOSFET. This means that a BC516 or BD680 is suitable, but not a TIP142! The circuit grounds of the two devices are connected together. Naturally, this compact level shifter can be used for other

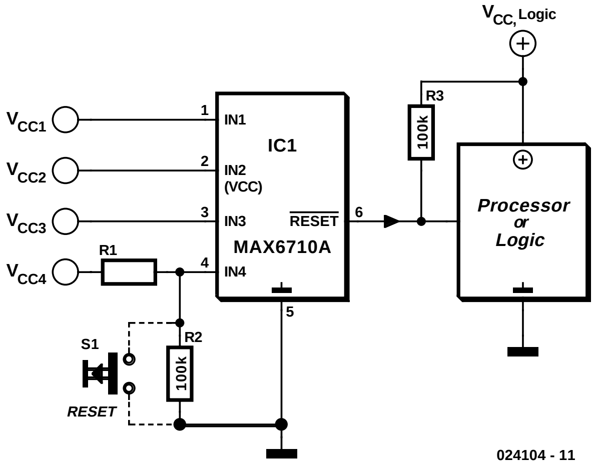


applications as well; the application shown here illustrates how an open-collector output can be connected to a system having a different supply voltage.

(024037-1)

Fourfold Voltage Monitor

025



A Maxim MAX6710 IC (http://pdfserv.maxim-ic.com/arpdf/MAX6710.pdf) can be used to monitor four supply voltages. If any monitored voltage drops below the threshold voltage set by the manufacturer, the IC triggers a Reset signal. This signal remains active for 140 ms after the voltages on all four inputs rise above the threshold voltage, in order to reliably reset the connected system.

The fourth input (IN4) can be freely programmed using an external voltage divider. Its threshold voltage is set to 0.62 V. To calculate the values of the voltage divider resistors R1 and R2, you can choose a value for R2 (100 kΩ, for instance) and then calculate the value of R1 using the formula

$$R1 = R2 \cdot \left(\frac{V_{CC4,th}}{0.62V} - 1 \right) = R2 \cdot \left(\frac{U_{ALARM}}{0.62V} - 1 \right)$$

The IC takes its power from the voltage applied to IN2, with a current consumption of only 35 μA. A faultless reset signal can be generated with a voltage of only 1 V present at IN1 or IN2. The IC is available in an SOT23 SMD package. A selection of the available types is shown in the table.

If it is necessary to be able to generate a reset signal using a manually operated pushbutton, a pushbutton switch can be simply connected in parallel with resistor R2 in order to connect input IN4 to ground when the button is pressed. In this case, R1 is required for current limiting.

(024104-1)

Type	IN1	IN2	IN3	IN4
MAX6710 A	5 V	3.3 V	2.5 V	0.62 V *
MAX6710 B	5 V	3.3 V	2.5 V	0.62 V *
MAX6710 C	5 V	3.3 V	1.8 V	0.62 V *
MAX6710 D	5 V	3.3 V	1.8 V	0.62 V *
MAX6710 E	0.62 V *	3.3 V	2.5 V	1.8 V
MAX6710 F	0.62 V *	3.3 V	2.5 V	1.8 V
MAX6710 G	5 V	3.3 V	0.62 V *	0.62 V *
MAX6710 H	5 V	3.3 V	0.62 V *	0.62 V *
MAX6710 I	0.62 V *	3.3 V	2.5 V	0.62 V *
MAX6710 J	0.62 V *	3.3 V	2.5 V	0.62 V *
MAX6710 K	0.62 V *	3.3 V	1.8 V	0.62 V *
MAX6710 L	0.62 V *	3.3 V	1.8 V	0.62 V *
MAX6710 M	0.62 V *	3 V	2.5 V	0.62 V *
MAX6710 N	0.62 V *	3 V	2.5 V	0.62 V *
MAX6710 O	0.62 V *	3 V	1.8 V	0.62 V *
MAX6710 P	0.62 V *	3 V	1.8 V	0.62 V *
MAX6710 Q	0.62 V *	V_{CC}	0.62 V *	0.62 V *

* programmable using voltage divider R1/R2

Low-Drop Current Source

026

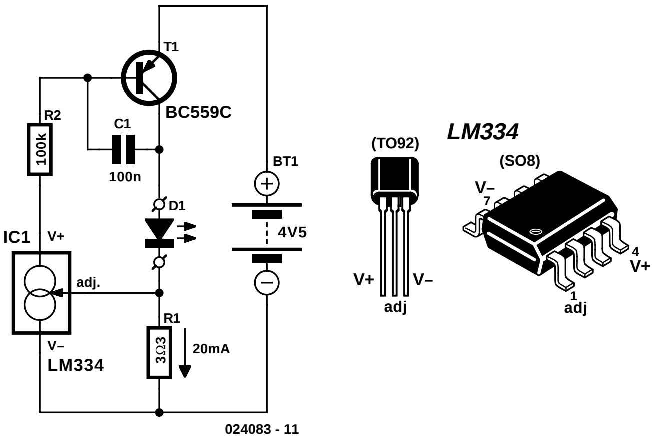
All simple constant-current sources generally operate on the same principle: a current is allowed to flow through a resistance and some sort of regulator is used to try to hold the voltage across this resistance constant. If this is done using a transistor, there must be a voltage drop of approximately 0.6 V over the resistor in order to forward bias the base. However, in some cases this yields an excessive loss, so an opamp with a reference source is used instead. The type LM334 adjustable current source has all of this 'on board' and regulates with a voltage drop of 64 mV. The associated schematic diagram shows a practical example of a current source using this indestructible IC. Here R1 is the sense resistor that determines the current level. Its value is calculated using the formula

$$R1 = 0.064 \div \text{current}$$

For example, for a current of 20 mA the value of R1 must be 3.2 Ω.

The illustrated circuit is exclusively intended to be used for small voltage spans and small currents, since T1 can dissipate 100 mW at most. However, you are free to experiment with other components and component values. With the values shown on the schematic, the circuit is eminently suitable for powering a white LED with an operating voltage of 3.6 V from a 4-V lead-acid battery or a 4.5-V battery.

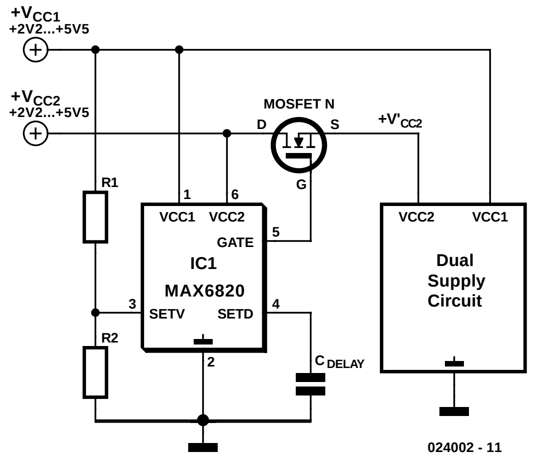
(024083-1)

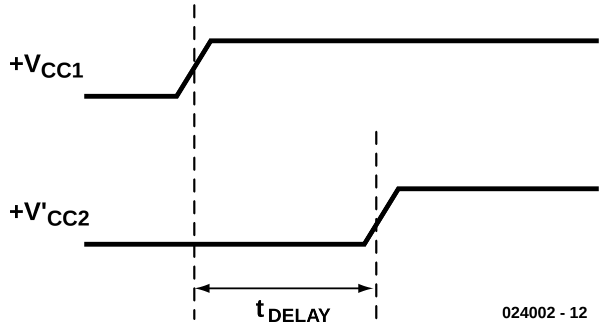


Supply sequencer

027

It is often the case with electrical systems using more than one power supply that the supplies must be switched on in a defined sequence to avoid electrical damage that may otherwise occur. The MAX6820 chip from Maxim (http://pdf-serv.maxim-ic.com/arpdf/MAX6819-MAX6820.pdf) contains all the necessary control circuitry to perform this function in one package. The chip samples the primary supply V_{CC1} at its input SETV (pin 3) via the potential divider formed by R1 and R2. When the voltage at this pin rises above 0.62 V a timer is initiated in the chip. At the end of this timer period the GATE output switches an N-channel MOSFET to connect V_{CC2} through to the secondary circuitry. Both V_{CC1} and V_{CC2} must be greater than 2.125 V otherwise the chip will detect an undervoltage condition and turn off the MOSFET. The values of R1 and R2 can be calculated using the equation:





$$R1 = R2\,[(V_{TH} / V_{TRIP}) - 1]$$

Where

V_{TH} = triggering threshold voltage

V_{TRIP} = 0.62 V

Capacitor C_{DELAY} gives the time delay t_{DELAY} and is calculated using the equation:

$$t_{DELAY} = 2.484 \times 10^6 \times C_{DELAY}$$

Where

The units for C_{DELAY} are µF.

t_{DELAY} is in seconds.

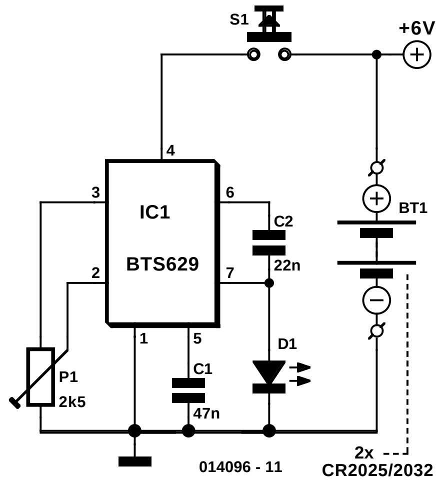
The MAX6820 uses an integrated charge pump to drive the MOSFET gate ensuring that the N channel MOSFET is fully enhanced with a low R_{DS} ON. To select a MOSFET choose one with a suitable R_{DS} ON for a V_{GS} bias of 5 V to 6 V. A BSP17, for example, may fit the bill and comes in a SOT223 package.

The MAX6819 performs the same function but has a fixed time delay of 200 ms. This device has no need for an external timing capacitor so the C_{DELAY} input is replaced by an ENABLE input.

(024002-1)

Lithium Torch

028



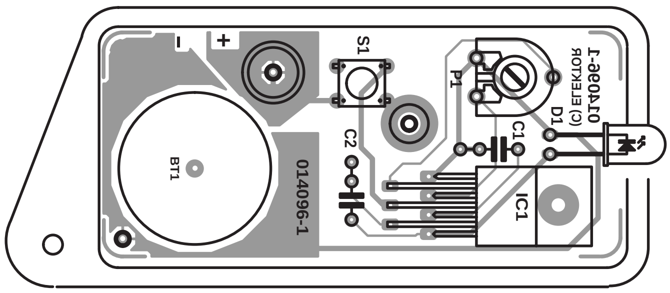
This mini pocket torch combines the advantages of Lithium button cells with a super-bright white LED. The lithium cells are small, have a long shelf life and have very little self-discharge. The LED has very high efficiency, an extremely long life expectancy and a modest current consumption (20 mA). In combination this results in an exceptionally long-life pocket torch.

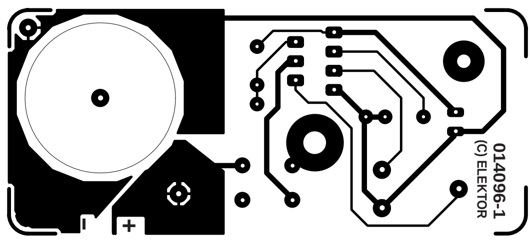
It is unfortunate that a Lithium cell has a voltage of only 3 V, while a super-LED has a forward voltage of 3.5 V. One cell will not suffice and we therefore will have to connect two in series. That results in a power supply voltage of 6 V, and at a current limit of 20 mA we need a series resistor of (6 - 3.5) / 0.02 = 125 Ω. Very annoying, because this resistor causes a power loss of 2.5 × 0.02 = 50 mW. Compared to the power consumption of the LED (3.5 × 0.02 = 70 mW) this would mean that nearly 42% of the energy would be wasted!

By using an integrated pulse-width modulator from Siemens, the BTS629, this power loss can be limited to about 10%. In combination with two CR2025 cells with a capacity of 170 mAh, the pocket torch will have an expected operating time of 15 hours. Two CR2032 (230 mAh) cells will last an astonishing 21 hours! Another advantage of the IC that has been used here, is that the pulse width, and hence the brightness of the LED, can be smoothly adjusted with P1.

The compact printed circuit board for this pocket torch has been designed so that it fits exactly inside the key ring enclosure UM14 from KM. The PCB has a 20 mm diameter hole for the button cells. The negative terminal for the button cells is made with a piece of paperclip soldered to the bottom. For the positive terminal, on the topside of the PCB a flat terminal can be attached with the aid of an M3 bolt and nut, as can be seen in the photograph. The PCB shown here is unfortunately not available ready-made.

(014096-1)





COMPONENTS LIST

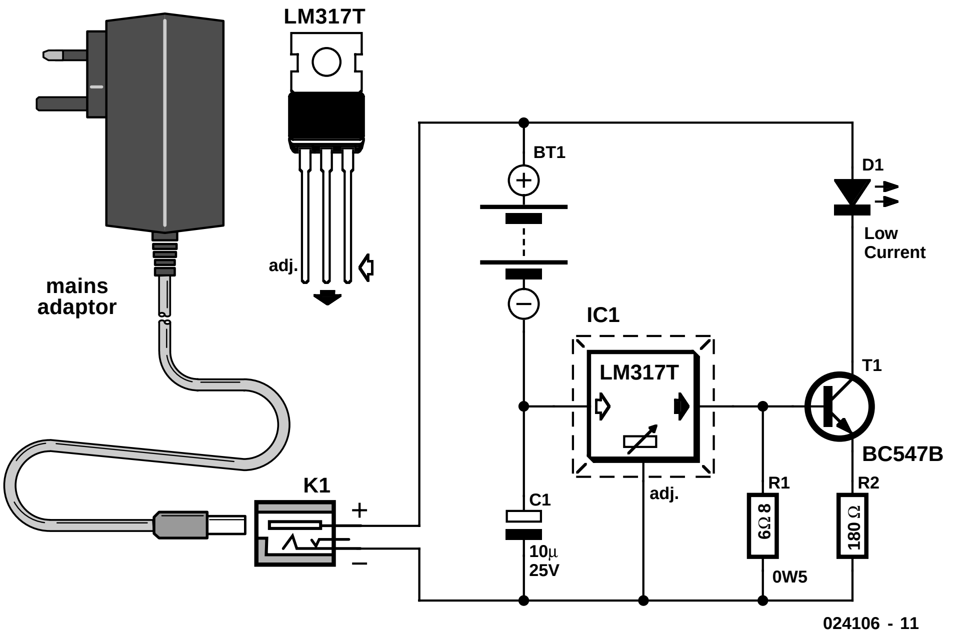
Resistors:
P1 = 2kΩ5 preset

Capacitors:
C1 = 47nF
C2 = 22nF

Semiconductors:
D1 = super HR LED, white
IC1 = BTS629 (Conrad Electronics # 1759 86)

Miscellaneous:
S1 = pushbutton (e.g., Farnell # MCDTS-5M)
BT1 = 2 off CR2025 or CR2032
Case: KM type Box UM14 (Nedis)

NiCd/NiMH Battery Charger 029



Here we have yet another excellent universal battery charger that is easy to build and can be used to safely charge practically all commonly used NiCd and NiMH penlight cells. The only downside of the universal approach is that it is not a fast charger, since it works with the well-known standard charging current of one tenth of the battery capacity in combination with a charging time of 10 to 14 hours.

With the advent of nickel–metal hydride rechargeable batteries, capacities have increased and it is no longer necessary to worry about the memory effect. This means that a topping-up charge can be used at any time, and if this is done using the above-mentioned current of one tenth of the battery capacity the charging time is not critical. In other words, the battery is guaranteed to be fully charged after 10 to 14 hours and there is no danger of overcharging, so it does not matter if you accidentally charge for 20 hours. If you are certain that the battery is only half empty, you can restore its full capacity by charging for around 6 to 7 hours.

Currently, penlight cells (AA) commonly have a capacity of 1500 to 1800 mAh (milliampère–hour), so the charging current should be 150 to 180 mA. If you want to charge several cells at the same time, you can simply connect them in series, since the same charging current will then flow through all the cells and they will all be charged simultaneously.

The question is thus how to obtain a current of 180 mA. The most elegant and accurate solution is to use a current source. Here we have 'misused' a type LM317 voltage regulator as a current source. The well-known LM317 three-lead regulator is designed to adjust its internal resistance between the IN and OUT leads to maintain a constant voltage of 1.25 V between the OUT and ADJ leads. If we chose a value of (1.25 ÷ 0.180) = 6.94 Ω for R1, then exactly 180 mA will flow. Since in practice you cannot buy a resistor with this value, we have chosen a value of 6.8 Ω, which is available. For convenience, an indicator LED has been added to the charger. This LED is illuminated only when current is actually flowing, so it can be used to verify that the batteries are making good contact.

In order to allow a current of 180 mA to flow, we require a certain voltage. The maximum voltage across a cell during charging is 1.5 V, and the current source needs around 3 V. If you charge only one cell, a supply voltage of 4.5 V is adequate. If you charge several cells in series, you need 1.5 V times the number of cells plus 3 V. For four cells, this means a supply voltage of 9 V. If the supply voltage is too low, the charging current will be too low. A supply voltage that is greater than necessary is not a serious problem, since the circuit ensures that the charging current cannot exceed 180 mA.

The required voltage can be conveniently obtained from a standard unstabilised mains adapter (or 'battery eliminator'), with a 300-mA type being highly suitable for supplying the required 180 mA. It is usually possible to select several different voltages with such an adapter, and it is recommended to choose the lowest voltage for which the indicator LED of the current source still lights up well.

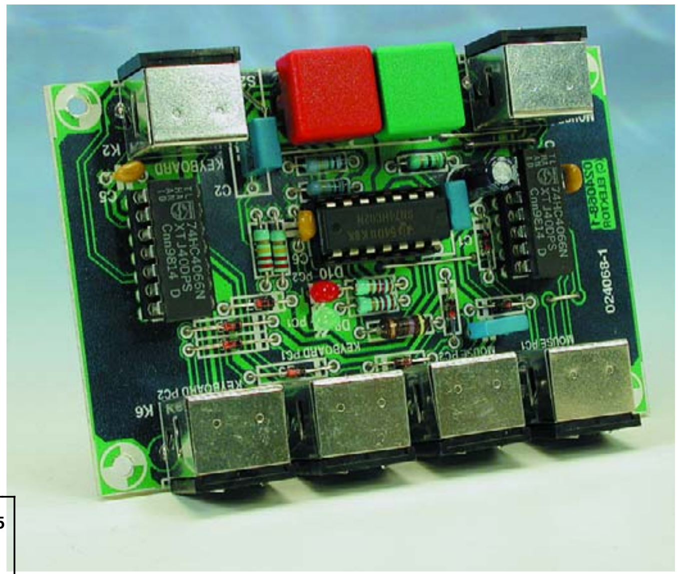
We should mention a couple of practical points. First, any desired colour of LED may be used, but it must be a high-efficiency (low-current) type, since such LEDs are brightly illuminated with a current of 2 mA as used here. When charging several cells in series, the cells should naturally be placed in a battery holder. Although it is not all that important for use with this charger, we would like to point out that the quality of most battery holders is very poor. The interconnecting springs sometimes have a resistance of as much as 1 Ω (!), which can result in considerable losses (a cell loaded at 1 A will then provide a voltage of only 0.2 V…).

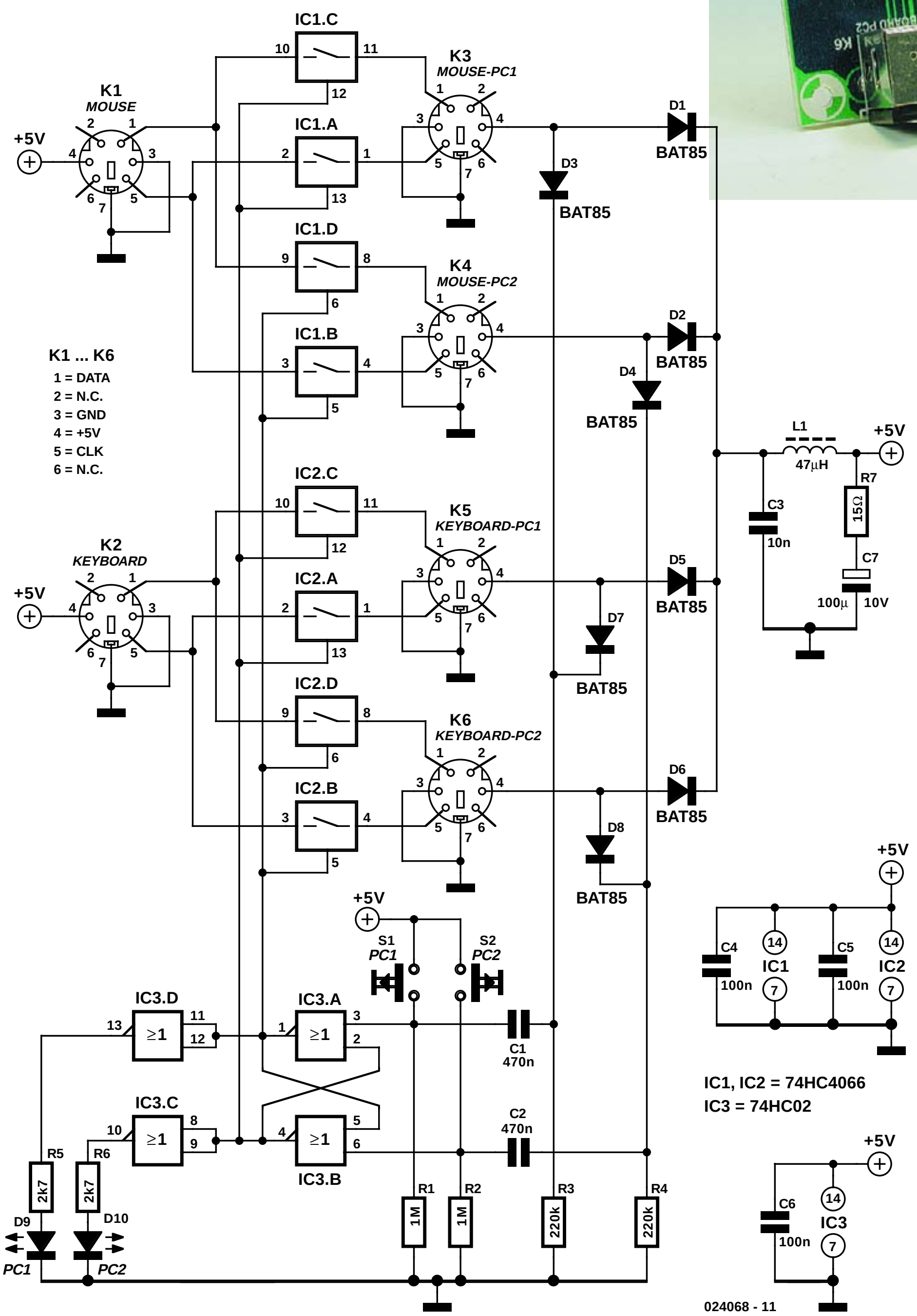
Finally, note that the LM317T (the 'T' refers to the package type) must be fitted with a heat sink. Although it will not be destroyed by being overheated, it's no fun to burn your fingers and it's naturally not particularly good for the charger to become so hot. A Fischer SK104 heat sink (approximately 10 K/W, available from Dau Electronics) is a suitable type.

(024106-1)

Keyboard/Mouse Changeover Switch 030

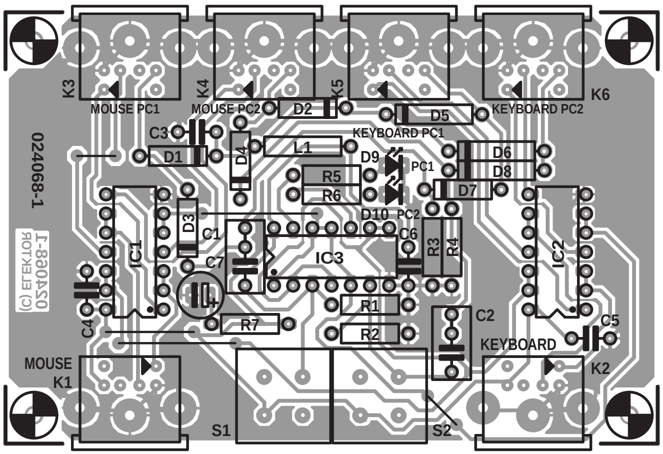
Old PCs are usually not immediately discarded after the purchase of a new one, but often remain in use for certain tasks. To avoid being tied to two keyboards and two mice when operating two PCs it would be nice if there was some kind of simple switch that would easily change the keyboard and mouse over from one PC to the other. That is the purpose of this circuit. The connections are made with four PS/2-cables (male/male). If one of the PCs requires a 5-pin DIN connector, an adapter cable or plug is also necessary. For the mouse an adapter plug from PS/2 to 9-pin Sub-D is also available, if required.

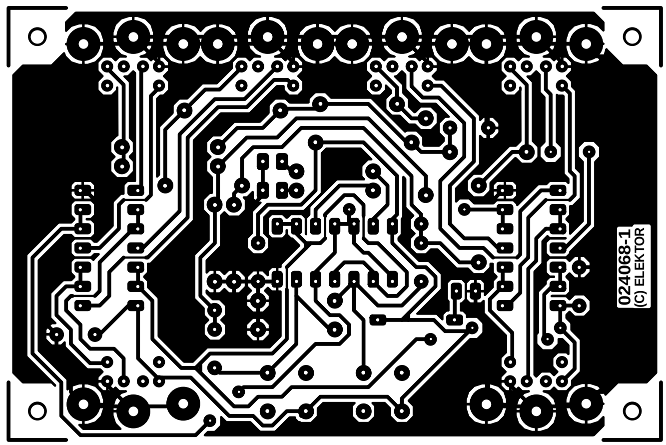




The operation of the whole circuit is quite simple. The power supply is obtained, with the aid of diodes D1, D2, D5 and D6, from the four connections to the PCs. Only one of the connections, K3 through to K6, to the PC needs to be powered up for this circuit to operate. L1 prevents any noise that may be present on the PC power supply from reaching the circuit. The power supply so obtained is directly connected with the keyboard and mouse.

Standard analogue switches from the HC-logic family (74HC4066) are used for the changing-over of the signal connections (clock and data). IC1 and IC2 are put to work as double pole changeover switches. To prevent short-circuiting the signals from both PCs, IC1 and IC2 are driven by a flip-flop with NOR-gates (IC3a and IC3b). Pushbuttons S1 and S2 change the state of the circuit. If both inputs of the flip-flop are made active, both outputs of the flip-flop will be 'low' and neither of the PCs is connected through. The LEDs D9 and D10 indicate which PC is currently connected to the keyboard and mouse. If no PC is connected then both LEDs are lit (this happens when





both S1 and S2 are pressed simultaneously). For a visually pleasing effect, the parts list indicates switches from ITT-Cannon with a red and green actuator so that the colour of the switches corresponds to the colour of the LEDs.

It is important that when the PC first starts up or re-boots the switch is in the correct position or no keyboard or mouse will be detected. In the first case the voltage that appears when the PC is switched on is used to trigger the flip-flop into the correct state: D3 and D7 for PC1, and D4 and D8 for PC2. R1/C1 and R2/C2 generate the reset pulses; the duration of nearly half a second is more than sufficient. R3 and R4 are necessary to discharge C1 and C2. Electrolytic capacitor C7 ensures proper decoupling of the power supply. The job of R7 is to limit the charging current through the BAT85 power supply diodes when the PS/2 cable is first connected.

The current consumption is about 1 mA and is almost entirely attributable to the LEDs. We conclude with a few things that are worth knowing. When shutting down the PC it appears that under Windows the mouse from certain manufacturers becomes deactivated. Make sure then, that when shutting down one PC, the other one is selected as quickly as possible, or this PC may not respond to the mouse any more. Also make sure that both PCs have the same mouse driver installed.

(024068-1)

COMPONENTS LIST

Resistors:
R1,R2 = 1MΩ
R3,R4 = 220kΩ
R5,R6 = 2kΩ7
R7 = 15Ω

Capacitors:
C1,C2 = 470nF
C3 = 10nF, lead pitch 5mm
C4,C5,C6 = 100nF ceramic, lead pitch 5mm
C7 = 100µF 10V radial

Semiconductors:
D1-D8 = BAT85
D9,D10 = red and green LED, 3mm, high efficiency
IC1,IC2 = 74HC4066
IC3 = 74HC02

Inductors:
L1 = 47µH

Miscellaneous:
K1-K6 = mini-DIN6/PS2 socket (female), PCB mount
S1,S2 = pushbutton, e.g., ITT/Cannon switch D6-C-40 (square, red) and/or D6-C-50 (square, green); optional snap-on button BTN-D6-40 (red) BTN-D6-50 (green)
PCB, order code **024068-1**

Battery Voltage LED

031

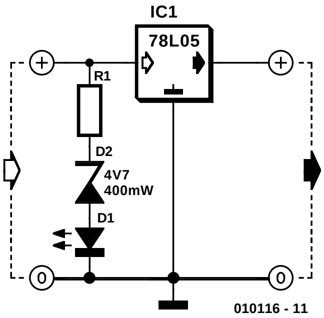
H. Bartelink

In many small battery powered apparatus it is common to have an LED act as a combined indicator for ‘power-on’ and ‘battery state’. The most commonly seen circuit fir this is shown in Figure 1. Although it is assumed that a 78L05 is used in all circuits to be discussed, the concepts can be extended to other (low-drop) voltage regulators as well.

The 78L05 requires a minimum input voltage of 6.5 V to work properly. In Figure 1, the LED voltage will be about 1.8 V, with the zener diode dropping 4.7 V and the resistor, the excess voltage above 6.5 V. Note that a low-current LED is recommended because of its modest current requirement of just 2 mA. Once the battery voltage drops below the sum of the zener voltage and the LED voltage, the LED is turned off.

In the case of the circuit in **Figure 2**, assuming that the load current is more than a few milli-amps, the current flows through the LED-resistor combination. The value of the resistor is worked out to pass slightly less than the minimum load current. In that case, the LED will help to shunt some current past the regulator, while not wasting battery power as in Figure 1. For loads over 20 mA, work out the resistor value so as to let the regulator do more work. As the battery is drained

1



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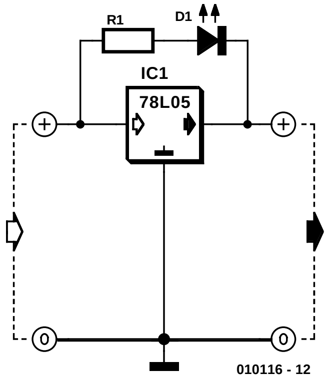
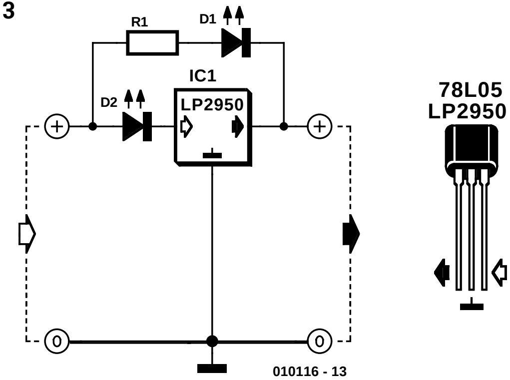


3



the LED will dim and eventually turn off as the battery voltage starts to approach the minimum working voltage of the regulator.

In **Figure 3**, two LEDs are used with different turn-on voltages (i.e., different colours). If the regulator used is a low-drop type with a minimum voltage drop smaller than about 0.1 V, then LED D1 is the low-battery indicator and LED D2, the power-on indicator. For this to work, D1 must have a turn-on voltage which is about 0.2 V higher than that of D2.

The LEDs in Figures 2 and 3 may be 2-mA types or the more commonly found standard 20 mA ones. Note that the maximum current through a normal LED should not exceed 50 mA or so.

(010116-1)

Telephone Watchdog

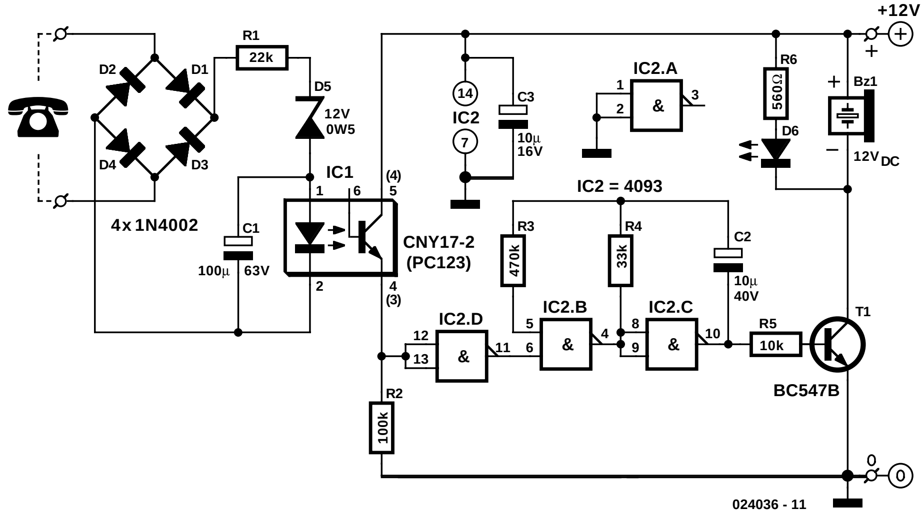
032

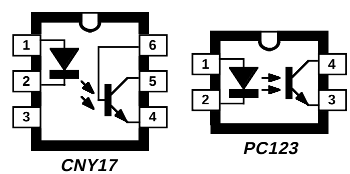
T. Hareendran

This simple circuit will tell you, by sounding a buzzer and flashing a LED, that a telephone line is in use.

If the telephone line is free, it will carry a steady direct voltage of something between 48 and 60 volts. This is well above the zener voltage of D5 so optocoupler IC1 is switched on and consequently the oscillator built around IC2 is held disabled.

When the line is in use, i.e., a telephone extension is ‘off hook’, the line voltage drops to a level below the zener voltage of D5, and the oscillator is enabled. LED D6 will flash and buzzer Bz1 will produce intermittent beeps. Note that the buzzer is a DC type.



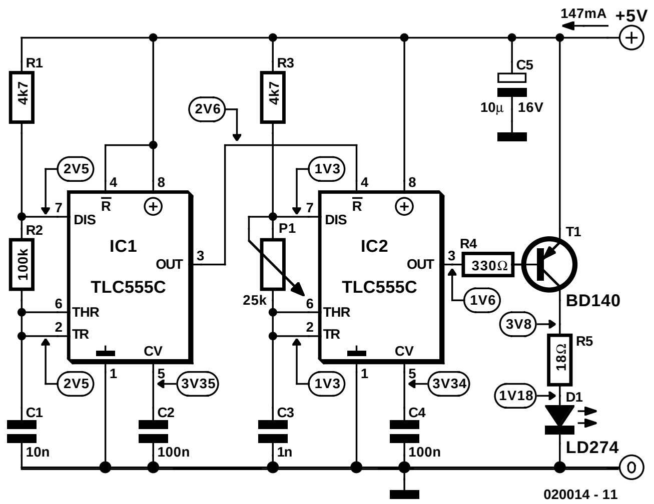


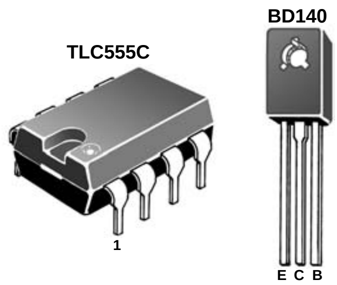
The circuit is best (and most safely) powered by a mains adaptor with an output of 12 Vdc. Since the current consumption is minimal at just 25 mA or so, the smallest adapter you can find will be okay to use.

Both optocoupler types indicated in the circuit diagram meet a 5-kV isolation breakthrough specification. The PC123 from Sharp though has ‘bent out’ pins and so meets the isolation distance requirement of 6 mm for equipment connected to the mains, which may well be extended to public telephone systems. The CNY17 does not meet the 6 mm spec and is strictly speaking not safe to use unless you bend out the pins yourself and refrain from using an IC socket.

(024036-1)

Simple Infrared Light Barrier 033





Pradeep G.

This infrared alarm can be used to detect persons passing through doorways, corridors and small gates. The transmitter emits a beam of infrared light which is invisible to the human eye. The buzzer at the output of the receiver is activated when the light beam is interrupted by a person passing through it. The transmitter and receiver circuits shown here have been designed for a range of several meters, almost independent of ambient light conditions. Only in the rare case of the receiver sensor being exposed to bright, direct sunlight, some screening measures have to be added.

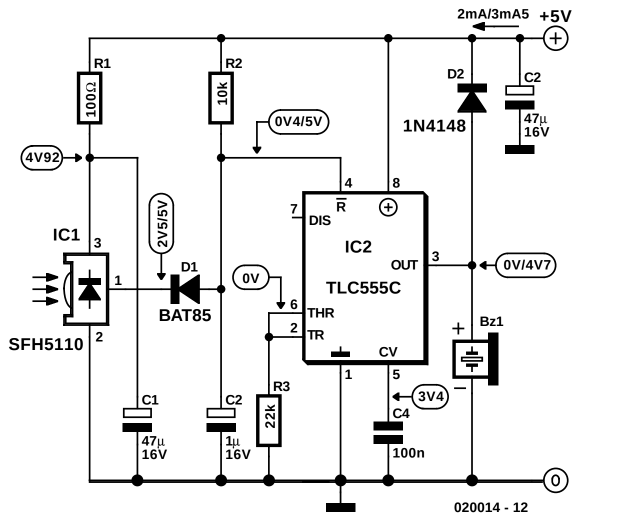
The transmitter does not emit a continuous infrared signal, Rather, it is modulated, that is, the 36-kHz carrier used to pulse the IRED (infrared emitting diode) on and off is itself switched on an off at a rate of about 300 Hz. The reason for doing so is that most infrared sensors, including the ones suggested in the diagram do not respond very well to continuous incidence of infrared light. Switching the IR source off, even for a small period, allows IR detectors to 'recuperate', and so optimise their ability to minimize the response to ambient light.

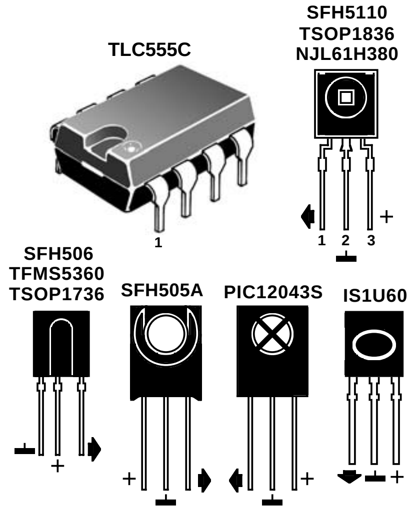
The transmitter consists of two oscillators built around the ubiquitous 555 IC. Here, the current-saving CMOS version TLC555 (or 7555) is used. Alternatively, the two 555's may be replaced by a single TLC556 (or 7556). IC1 is the 300-Hz generator, IC2, the 36-kHz source. The IRED type LD274 is pulsed at a relatively high peak current via driver transistor T1. If in your application the distance covered by the IR beam is relatively short, the value of resistor R5 may be increased to save on current consumption. Preset P1 is adjusted for a carrier frequency of 36 kHz exactly (failing test equipment, adjust it for optimum range).

The receiver is equally simple and also based on a CMOS 555. As long as the sensor picks up infrared light from the transmitter, the reset input of the 555 IC is held low and the buzzer is silent. Components D1 and C2 act as a low-frequency rectifier to cancel the effect of the 300-Hz modulation on the transmitter signal. When the infrared light beam is interrupted, the oscillator built around the 555 is enabled and starts to produce a warning tone.

Finally, the test values indicated in the circuit diagram are average dc levels measured with a DVM, under light/no light conditions. In fact, most test points carry rectangular or sawtooth waveforms.

(020014-1)





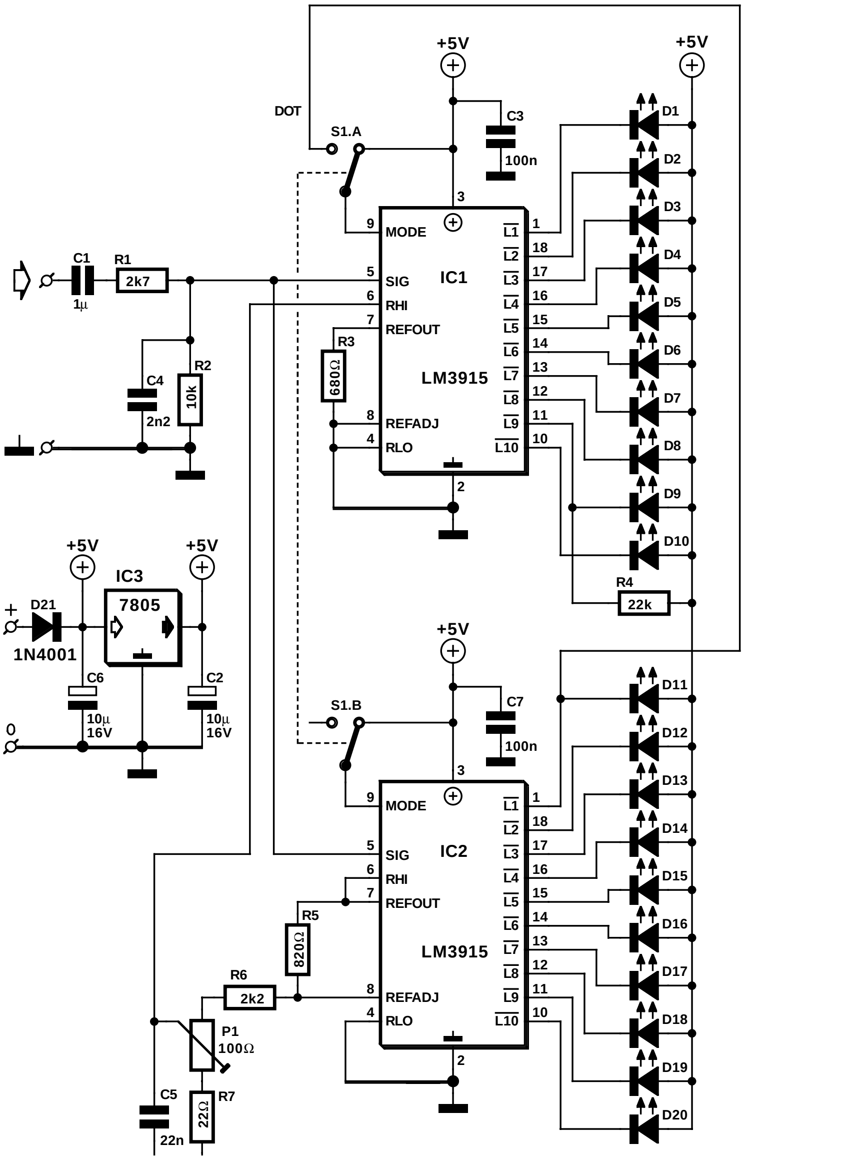
60-dB LED VU Meter

034

Rikard Lalic

Most of the analogue audio media, including radio broadcasting, stick within 60 dB dynamic boundaries. This VU Meter was designed to be used as a desktop instrument with home audio appliances, so it has its own power supply. Driven by an AC musical signal taken directly from low impedance loudspeaker connectors, i.e. **in** parallel with loudspeakers, and having linear frequency response, it covers 60 dB dynamic range in 3 dB steps using 20 LEDs in a bargraph. Low component count and simplicity enable the circuit to be housed in a small box or behind a transparent shield like a small acrylic desktop photo holder.

The LM3915 IC from National Semiconductor senses voltage levels and drives 10 LEDs, providing a logarithmic 3 dB/step analogue display, and so covering a 30 dB range. LED current drive is programmable and regulated. The IC contains adjustable voltage reference source and an accurate 10-step 22-kΩ voltage divider array. A ground referenced, ±35V-proof input buffer amplifier, capable of sensing down to ground, is driving 10 comparators referenced to the voltage divider. Applying an additional resistor in series with the input raises input protection to ±100 V. Two LM3915N (IC1 and IC2) are cascaded here to achieve a total dynamic range of 60 dB. R5 programs the LEDs current on IC2 while network R5-R6-P1-R7 sets the reference voltage that determines the full-scale input signal level of IC2. In this case it is set to 5.0V. The full scale level of IC1 is derived from this reference and shifted 30 dB below that of IC2. It is precisely preset using the P1, with R3 programming the LED current supplied by IC1. The value of R3 is smaller than R5 to compensate for IC2's internal voltage divider which is connected in parallel with the reference voltage source in IC2. The adapted value of R3 ensures that that there is no difference in LED brightness between IC1 and IC2.



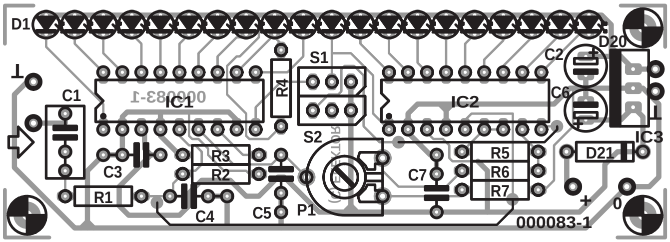
The audio signal to be measured arrives at pins 5 of IC1 and IC2 via C1-R1-R2-C4. R1 and R2 form a voltage divider and C4 is added for RF suppression. With R1 at 2.7 kΩ as shown in the schematic, full-scale indication is reached at 6.4 V_{rms} (which equals 10 W across 4 Ω). Depending on the output power of your amplifier, suitable values fro R1 and C4 may be selected from **Table 1**. As the VU-meter input is connected across the loudspeaker, power, *P*, and voltage, *U*, equate like

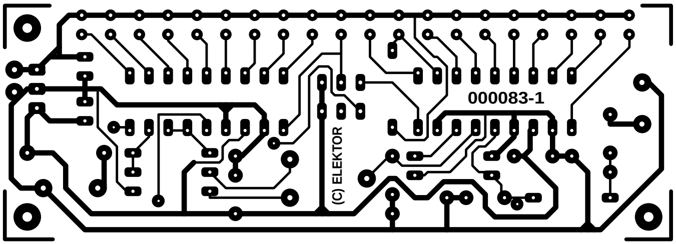
$$P = U^2 / Z$$

where *Z* is the loudspeaker impedance in ohms. Each lower

Table 1.

Loudspeaker	4 Ω	4 Ω	4 Ω	8 Ω	8 Ω	8 Ω	16 Ω	16 Ω	16 Ω
Power	10 W	50 W	100 W	10 W	50 W	100 W	10 W	50 W	100 W
R1	2.7 kΩ	18 kΩ	30 kΩ	6.8 kΩ+1.1 kΩ	30 kΩ	47 kΩ	15 kΩ	47 kΩ	68 kΩ+2.2 kΩ
C4	2.2 nF	470 pF	330 pF	1 nF	330 pF	330 pF	470 pF	330 pF	270 pF





COMPONENTS LIST

Resistors:

R1 = 2kΩ7 (see text)
R2 = 10kΩ
R3 = 680Ω
R4 = 22kΩ
R5 = 820Ω
R6 = 2kΩ2
R7 = 22Ω
P1 = 100Ω

Capacitors:

C1 = 1µF 63V (MKS, MKC)
C2,C6 = 10µF 16V radial
C4 = 2nF2 (see text)
C3,C7,C9 = 100nF
C5 = 22nF

Semiconductors:

IC1, IC2 = LM3915N
IC3 = LM7805
D1-D20 = LED

order LED in the chain indicates 50% power or 70.71% voltage of the first higher LED.

The threshold for LED #1 is just 7.0mV, so both noise and internal buffer and comparator offset voltages may influence the readout at the very low end of the LED bargraph (first few LEDs). Capacitors C4 and C5, proper wiring and correct PCB layout should help to maintain a good degree of noise immunity.

For **a** stereo version of the VU meter the metering circuits shown here should be duplicated. The power supply has already been dimensioned for a stereo version. A mains adaptor with an output voltage of about 8 Vdc is an inexpensive and safe way to power the circuit. The LED voltage is reduced to +5.0 V by regulator IC3 in order to keep the power dissipation of IC1 and IC2 within safe limits.

A double-pole switch, S1, allows the readout to be switched to ‘dot’ mode instead of ‘bar graph’.

Although its artwork is shown here, the printed circuit board designed for the LED VU meter is not available ready-made. IC3 needs no heatsinking.

The VU meter requires only one, simple adjustment. Connect a DVM to pin 6 of IC1 and adjust preset P1 to see 158 mV (5.0 V / 31.62), that is, –30 dB relative to the voltage present on pins 7 and 8 of IC2.

Finally, this VU meter must not be used with BTL type of audio amplifiers which could be found in some car radio receivers but only with common-ground type of amplifiers.

(000083-1)

Active Band-pass Filter up to 5 MHz

035

Constructing an active bandpass filter is greatly simplified by using the LT1567 from Linear Technology (www.linear-tech.com/pdf/1567i.pdf). The circuit shown here uses relatively few external components to build a 2nd order bandpass filter with a centre frequency of 1 MHz and a –3 dB bandwidth of 0.71 MHz. The centre frequency gain will be unity. The circuit should be driven from a low impedance source so that R1 will define the circuit gain. To recalculate the bandpass filters components for other frequencies the filter bandwidth ($BW_{-3\,dB}$) is given by:

$$BW_{-3\,dB} = 1 / (2\pi RC).$$

while the centre frequency (f_0) is given by:

$$f_0 = BW_{-3\,dB} \times \sqrt{(A_0 + 1)}$$

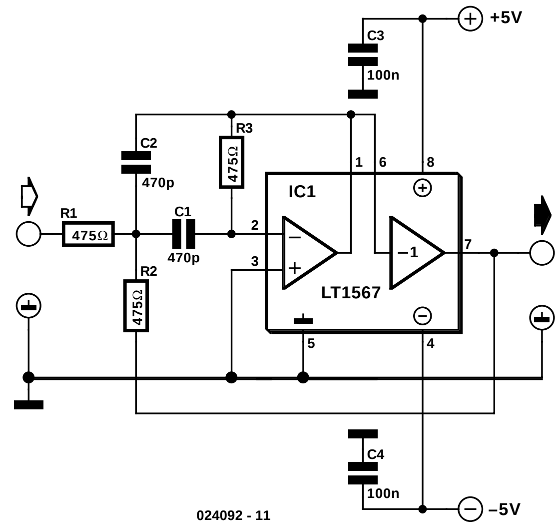
Where:

R=R2=R3 and C=C1=C2.

A_0 (gain at f_0) = R / R1.

$f_{0(max)}$ (maximum centre frequency) = 5 MHz / A_0

(024092-1)



Low-cost Position Sensor

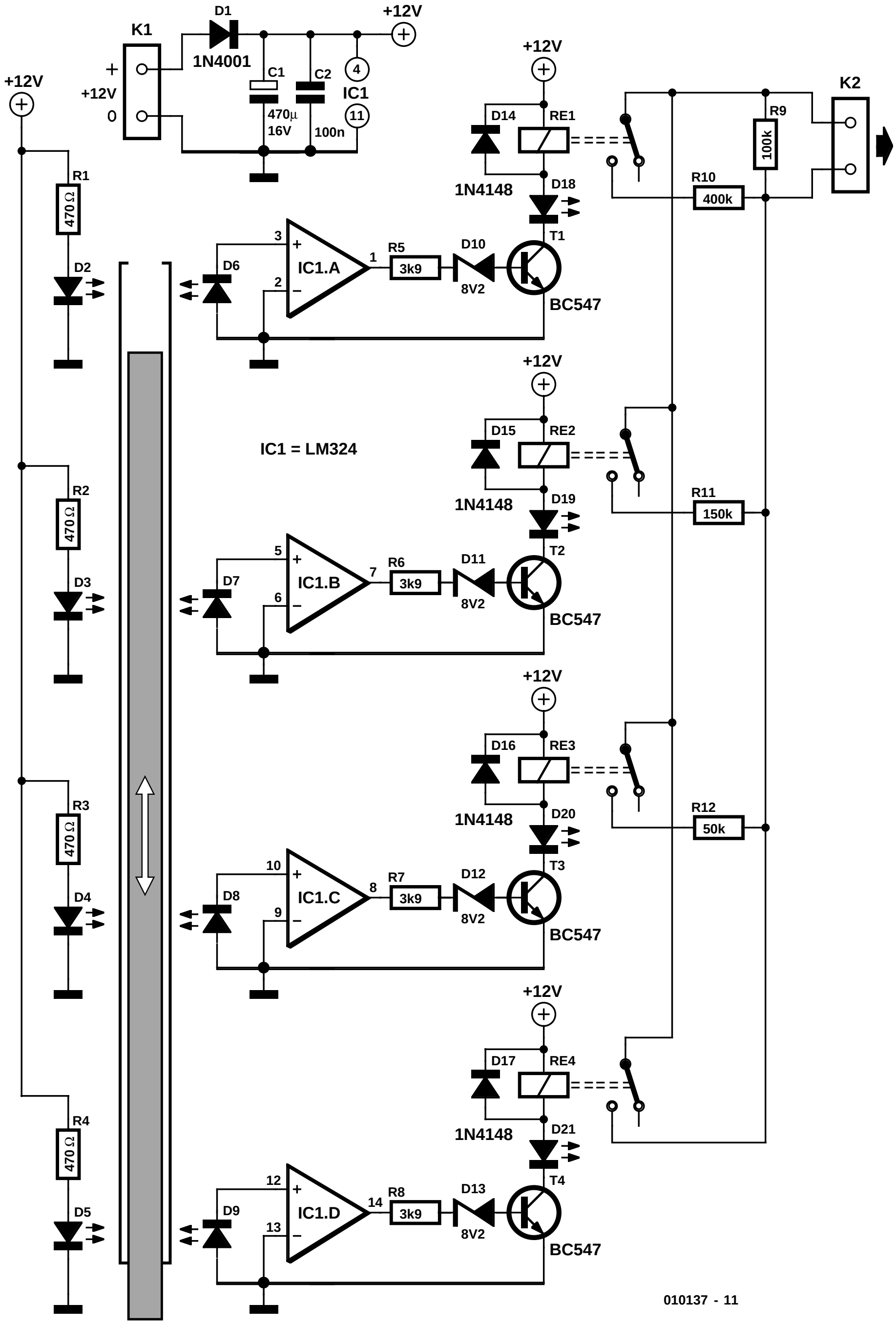
036

H. Lindberg

Ever wanted to build a positioning encoder? Here's one and you do not even need any sensors. The circuit uses an LED as a light emitter and another LED as a light receiver. This is possible due to the ability of the LED to generate a very tiny current when light is shining into it, almost like a small solar panel. A red LED is used because tests indicated that they gave the best results. It is possible though to use a different colour than red and the best combination might be found by trial and error. Please note that not all red LEDs give the same result, so you might have to get hold of a few different types and then find the best one. When a receiver LED is not illuminated by its associated sender device, it will prevent a tiny current of about 50 nA from flowing out of opamp pin 3 to ground. Once the LED is illuminated, this current is allowed to flow. In fact, even more current flows because the 'solar cell' quality of the LED causes pin 3 to go negative with respect to ground.

Because of the series-connected LEDs D18-D20, the supply voltage, Vcc, will have to be a few volts higher than the relay coil voltage. If that is the case, the values of R1-R4 are subject to experimenting because the operation of the light barriers depends on the supply voltage and the efficiency of the LEDs used.

The LEDs (transmit and receive) are mounted onto the sides of a tube and can 'see' each other through two 3-mm holes. The light barrier is then "broken" when a rod is inserted into the tube. This approach can of cause be multiplied as many times as required. Here, four barriers are shown, which allow up to five rod positions to be detected. Each position, including 'rod not inserted' is translated into a discrete resistance value which is created by connecting resistors in parallel, and may be measured on the output connector for further processing by an ADC or what have you. The values of the resistors on the relay contacts may be adapted to suit your needs.

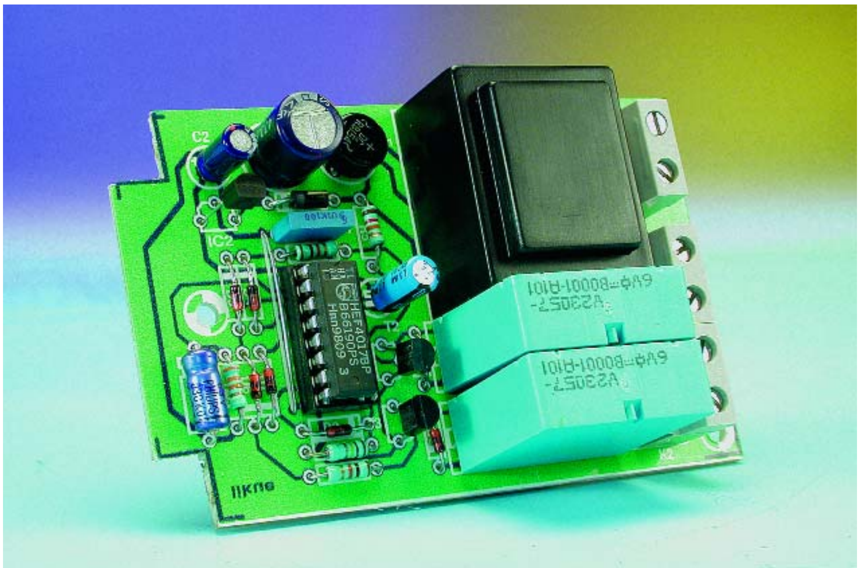


The LM324 opamp is a quad type, making the layout a bit easier. On the output of each opamp, a series resistor and a zener diode ensure proper voltage levels for the buffer transistor, which drives the relay. Across each relay coil sits a back EMF protection diode (always a good idea to add). As a matter of course, the transistor driving the relay has to be able to handle the required coil current.

(010137-1)

Multi-Position Mains Switch

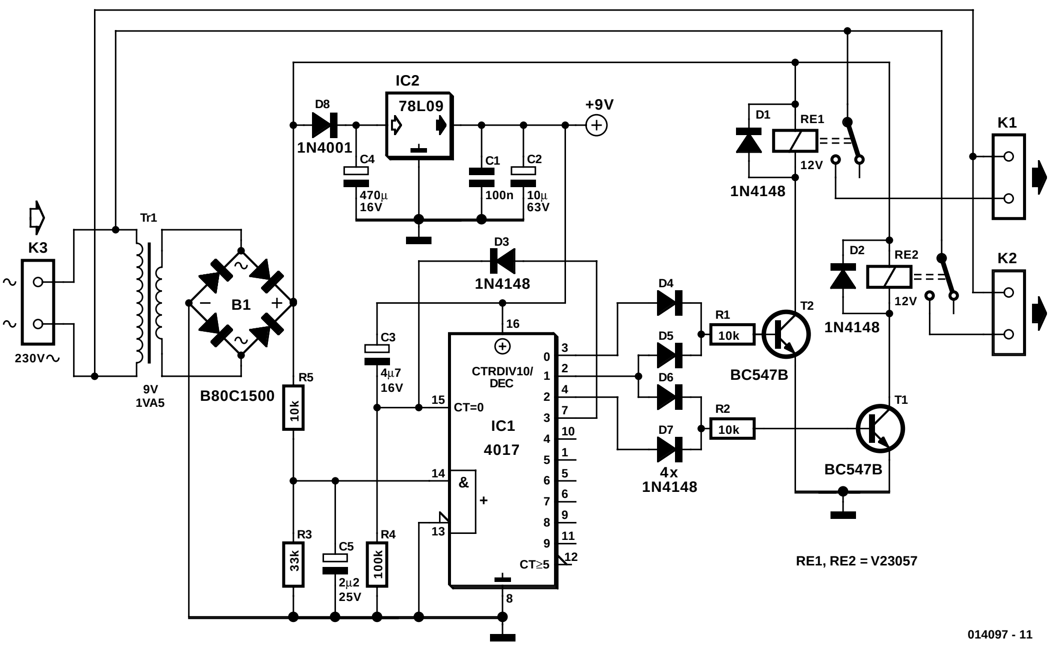
037



The circuit shown here was born out of necessity after one of our colleagues had just renovated his kitchen and realised afterwards that there were not enough switches. Obviously he was not too keen to partially demolish the kitchen to install a few additional wires in the already tiled wall. That's how the idea arose to develop a clever electronic circuit that would operate two lamps with only one switch.

All this appeared to be easy to realise by adding a small circuit, consisting of a decade counter, a diode network, two relays and a low voltage power supply. The schematic shows how simple the design of the 'multi-position' extension really is. K3 is connected to the switched wires that go to the original light. K1 and K2 are the connections for the two new lamps.

The operation is simply based on the fact that at every low to high transition at the CLK input of IC1 the active output moves over by one position. In combination with the diode network D4 through D7 this ensures that with a single wall switch it becomes possible to control two outputs. When the mains voltage is applied to K3 for the first time, Q0 will be high and Re1 will be energised. When the mains switch is briefly switched off and then on again it will have no consequences for the 9-V power supply, because C4 is quite large. But this will result in a trigger pulse on the CLK input, so that Q1 will now be high and via D5 and D6 both relays are energised. After another off/on cycle of the mains switch, Q2 will be high, relay Re1 will de-energise and only Re2 is still activated. If we repeat the off/on cycle once more we're back at the starting position and only Re1 is



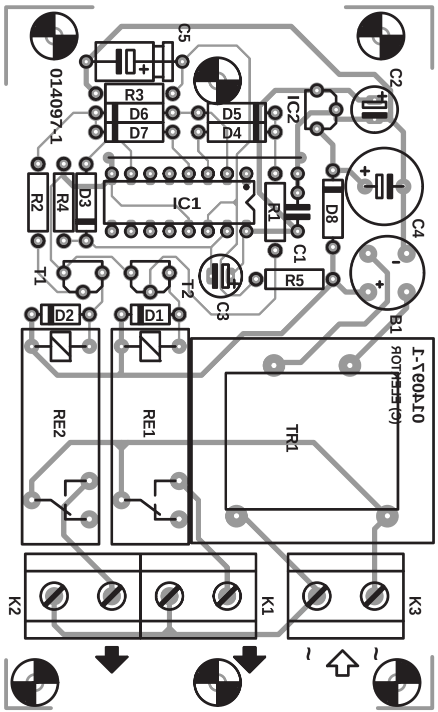
COMPONENTS LIST

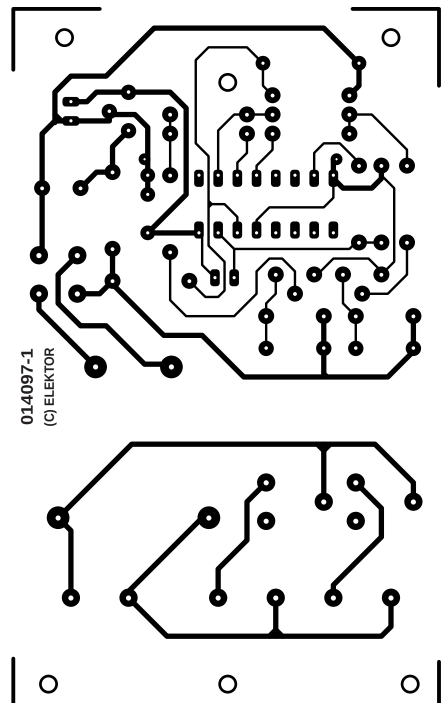
Resistors:
R1,R2 = 10kΩ
R3 = 33kΩ
R4 = 100kΩ
R5 = 10kΩ

Capacitors:
C1 = 100nF
C2 = 10µF 63V
C3 = 4µF7 63V radial
C4 = 470µF 16V radial
C5 = 2µF2 63V axial

Semiconductors:
D1-D7 = 1N4148
D8 = 1N4001
T2,T3 = BC547B
IC1 = 4017
IC2 = 78L09

Miscellaneous:
K1,K2,K3 = 2-way PCB terminal block, lead pitch 7.5mm
Tr1 = mains transformer 9V 1.5VA, e.g., Hahn EI 3022021
B1 = B80C1500 (round case) (80V piv, 1.5A)
Re1, Re2 = 12V relay, e.g., V23057 B0002 A101 12V (Schrack)
Case, Bopla, OKW or Schyller





energised. If the switch remains in the 'off' position then both relays will also be off.

A printed circuit board has been designed for this extension so that the entire circuit will fit without any problems in a waterproof enclosure from OKW, Bopla or Schyller. The 9-V transformer is also fitted on the PCB. PCB screw terminals can be used for K1, K2 and K3. Since the circuit is directly connected to the mains voltage we emphasise that the well-known safety rules need to be observed. When making any measurements or performing other operations on the circuit is it absolutely necessary to first break the connection to K3! Unfortunately, the PCB shown here is not available ready-made.

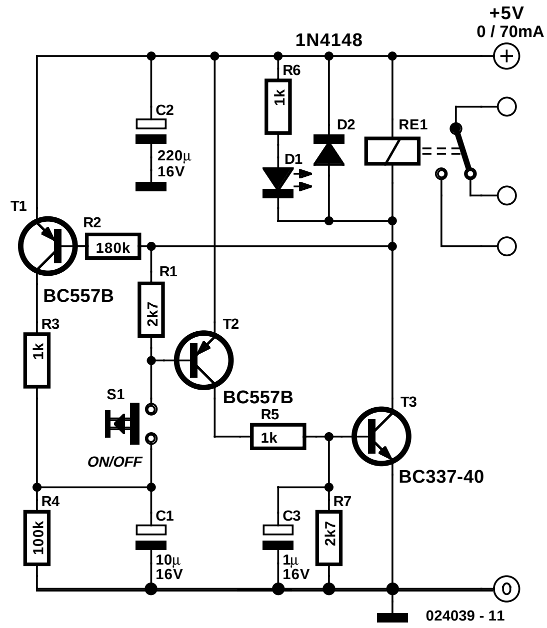
(014097-1)

Pushbutton Switch

038

K. Lorenz

This circuit acts like a two-position switch but is operated using a pushbutton. After power has been applied, the circuit is in the following initial state: the bases of T1 and T2 are at the positive supply potential and the base of T3 is at ground potential. All transistors are cut off. The other contact of the pushbutton is at ground potential. No current flows through the relay coil and the status LED is off.
If the pushbutton is pressed, T2 and (after a slight delay due the RC network) T3 switch on. The collector of T3 is now nearly at ground potential, so current flows through the relay coil and the function LED is illuminated. T1 can also switch on. This situation is stable, since ground potential can reach the base of T2 via R1, so nothing changes when the pushbutton is released. C1 is charged via R3 to cause a positive potential to be present at the pushbutton. If the pushbutton is now pressed again, it connects a positive potential to the base of T2 instead of the ground potential. This causes everything to toggle back into the initial state.
Similar operation can be obtained using a thyristor circuit, and in fact T2 and T3 form a sort of thyristor. However, the circuit shown here is largely independent of the voltage and current demands of the connected load. The relay coil should be suitable for the supply voltage (5–12 V) and should not draw more



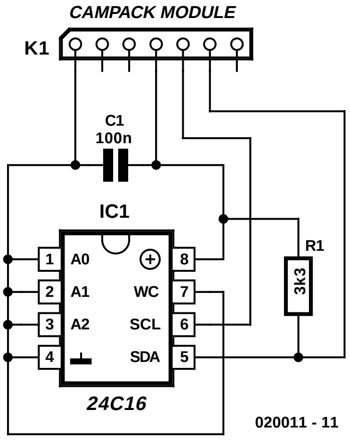
than 250 mA, since otherwise T3 will go up in smoke. With our lab prototype, we measured a current consumption of 70 mA in the ‘on’ state and less than 0.1 mA in the ‘off’ state.

(024039-1)

Memory Expansion for Futaba R/C Transmitters

039

1



2

COMPONENTS LIST

Resistors:
R1 = 3kΩ3

Capacitors:
C1 = 100nF

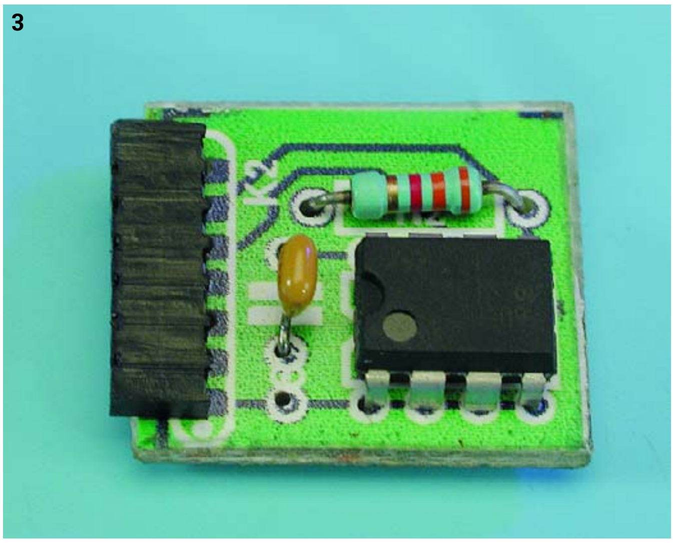
Semiconductors:
IC1 = 24C16

Miscellaneous:
K1 = 7-way connector (2 mm lead pitch!), e.g., Farnell # 672-348

In the model buildingworld a lot of transmitters are used with a built-in microprocessor. This processor occupies itself with carrying out, among other things, mixing functions, null point control of the servo, reversal of the servo, etc. This provides two significant advantages. On the one hand there is no need to purchase expensive analogue mixers again, because all of that is now implemented in software. On the other hand, you can use different settings for different models, because the various settings can be stored in non-volatile memory. However, many transmitters have very limited capacity in this respect. If desired, the memory can be expanded by purchasing a memory module. Expansion of the memory has the advantage that you can copy the current settings via the transmitter and then experiment with the settings in order to optimise them without losing the original data.

The circuit diagram, **Figure 1**, shows such a module. This module is intended as a replacement for the CamPack modules for use with Futaba transmitters. It is obvious from the schematic that the circuit doesn't amount to much. The memory element is a 24C16 EEPROM. This device, with a capacity of 2 kilobytes, has an I^2C interface, so that only a few connections are required to access it.

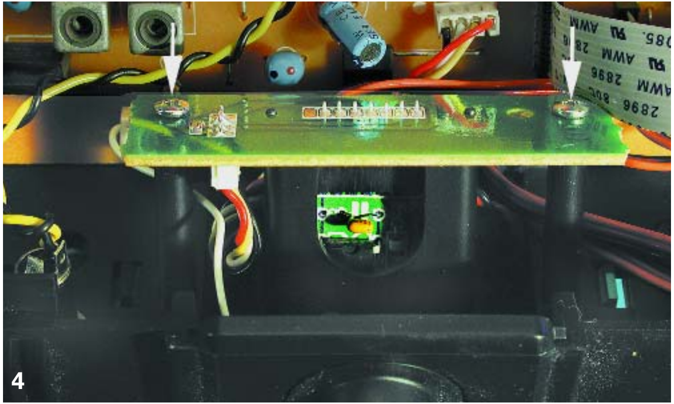
3



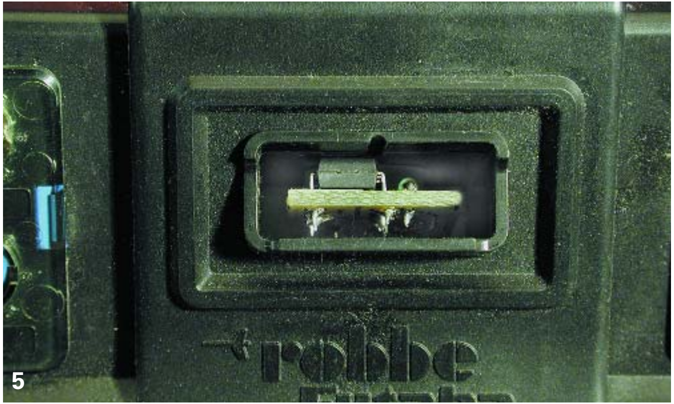
The assembly of the printed circuit board (**Figure 2**) can be accomplished in record time. It is important to note that IC1 (as can be seen in **Figure 3**) is not fitted in an IC socket, otherwise the circuit will not fit inside the transmitter! Also note that the connector used has a pitch of 2 mm, instead of the more usual 2.54 mm! After assembly has been completed (and the PCB has been carefully checked for short-circuits), the circuit can be fitted in the transmitter. The component side of the board needs to point in the direction of the antenna. It will also fit the other way around, but the consequences are less pleasant…

It is rather difficult to install the PCB in the slot in the transmitter that is specifically intended for this purpose, because our PCB is much smaller than the commercially available CamPack. It is therefore much better to take a different approach. First we have to open the transmitter, using exactly the same procedure as when replacing the battery. After this, the two screws that hold the small PCB with the expansion connector have to be removed (**Figure 4**). This small PCB can be found in the vicinity of the antenna. Once this little PCB has been removed it is easy to plug in the memory module (**Figure 5**: note the correct orientation!). Finally, fasten the original PCB again and replace the cover on the transmitter.

As soon as the Futaba radio control transmitter is turned on you will see a message indicating that the memory is being formatted. This takes a while, but only happens the first time. Once formatting has been completed the memory module is ready for use. From now on, instead of the original two,



4



5

we are able to store no less than 27 models in memory. Those of you who think that is still not enough can always build additional modules (also useful as back-up for the present settings). The PCB for this project is unfortunately not available ready-made.

(020011-1)

Audio Combiner

040

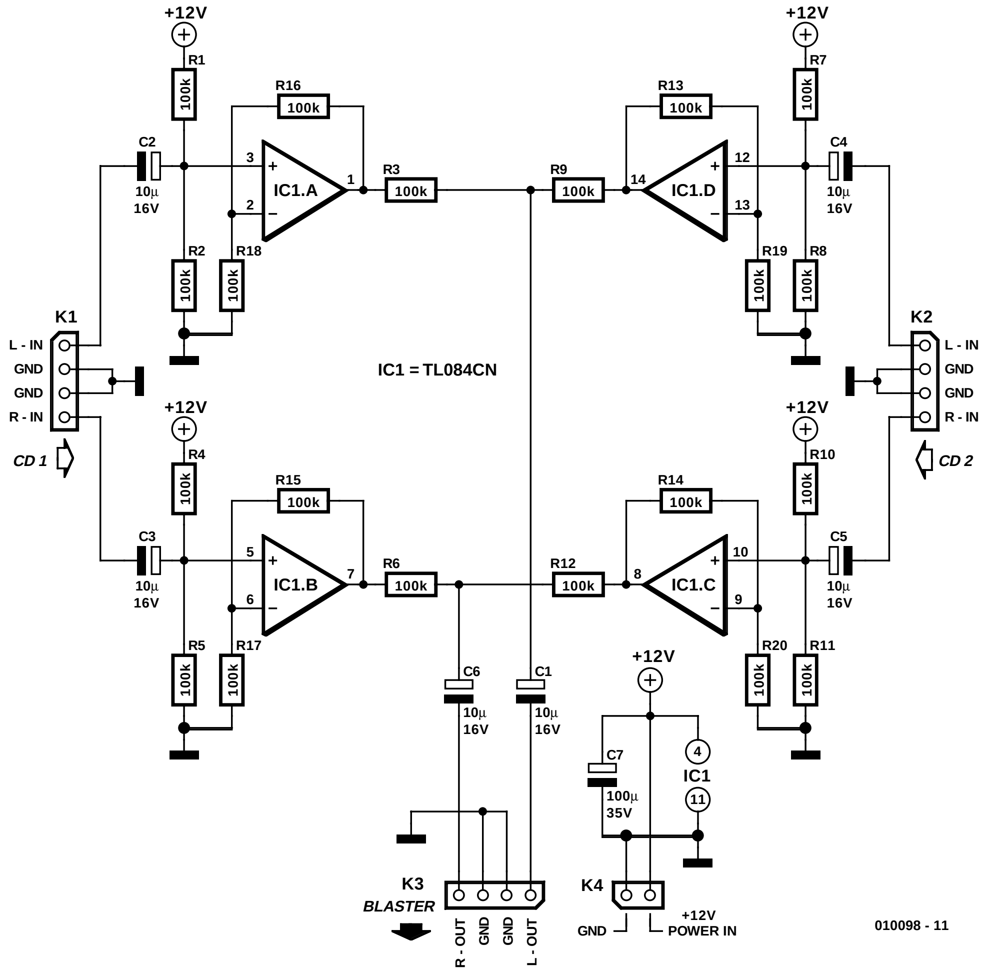
P.v.d.Hurk

This circuit arose from the need to couple the outputs of two CD-ROM drives to the input of a single sound card. Simply 'stacking' the connectors is possible but not particularly elegant, and the result is somewhat unpredictable due to mutual interference between the two drives. Using a quad opamp and handful of passive components, it is possible to quickly put together a mixer that first buffers the signals before adding them together. This completely eliminates any possibility of feedback or mutual interference.

The actual addition of the two stereo signals is performed by R3/R9 and R6/R12. Since these resistors also form voltage dividers that reduce the amplitude of each signal by half, the gain of the four buffer amplifiers is set to 2 as compensation.

The required 12-V supply voltage can be drawn from the PC without any problems. The current consumption of the entire circuit is practically negligible.

(010098-1)



Mini Audio DAC

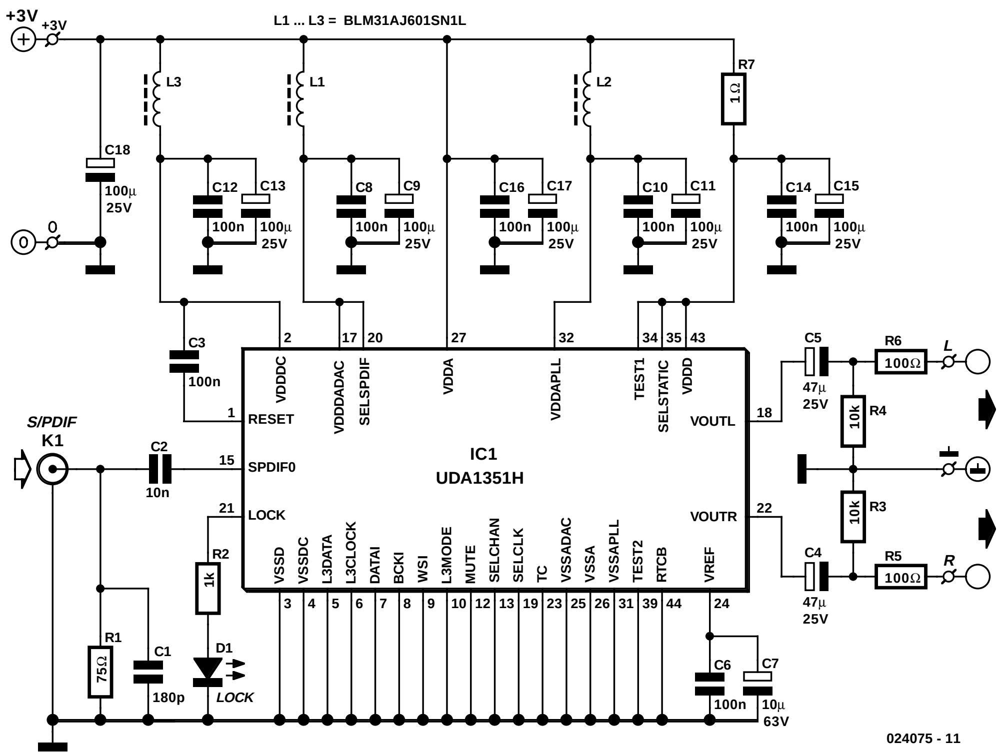
041

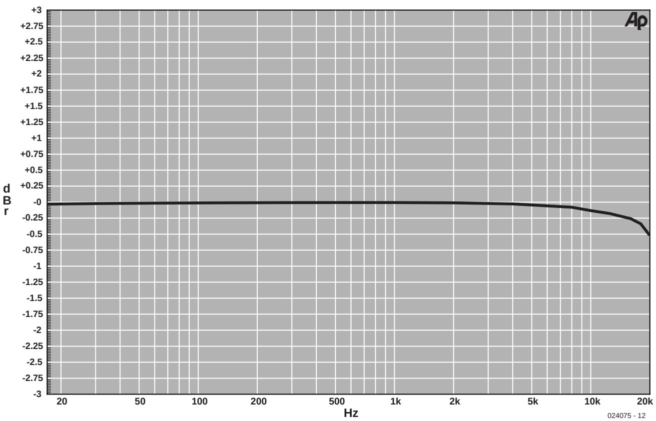
In the February 2002 issue of *Elektor Electronics*, we described an S/PDIF tester, consisting of a special decoder IC with integrated D/A converter. Two implementations of the 'IEC 60958 audio DAC' from Philips could be used here: the UDA1350ATS or the UDA1351TS. The latter has a frequency range up to 96 kHz. However, a number of readers were disappointed that this circuit was not accompanied by a PCB design. That is the reason that we present a similar 'Mini Audio DAC' here, this time with matching PCB layout.

The IC that has been used here, the UDA1351H, belongs to the same family as the aforementioned types, but is packaged in a different housing. The advantage of the SOT307-2 (QFP44) package is that the pins are a little further apart (0.8 mm instead of 0.65 mm), which makes soldering with a normal soldering iron considerably easier. The single sided PCB has been made as compact as possible and is fitted with components on both sides. The majority of components are on the actual component side, but six ceramic SDM capacitors have been placed on the copper side close to the IC, to obtain optimum decoupling. The are also three decoupling inductors in SMD packages, which are also on the solder side for the same reason. Including the cinch-connector the PCB is not larger than 51?37 mm.

For short duration use, two AA batteries may be used as power supply, but at 44 kHz the circuit draws a current of 22 mA and at 96 kHz, 33 mA, which is a little bit too much for a battery power supply. D1 indicates that a usable input signal has been detected. R3 and R4 ensure that output electrolytics C4 and C5 are charged even when no load is present, while R5 and R6 are the usual limiting resistors for capacitive loads. The output consists of three PCB pins, so that you can, depending on the application, fit cinch (RCA) chassis connectors or a jack to connect a headphone.

For a detailed description of the IC we refer you the article mentioned above and the datasheet for the UDA1351H. We conclude by listing the specifications, measured at a power supply voltage of 3 V.

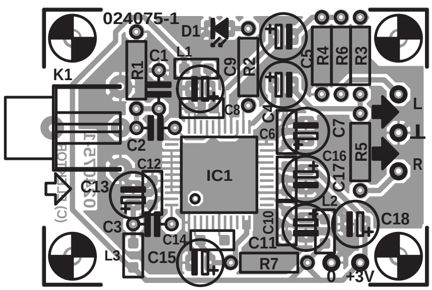


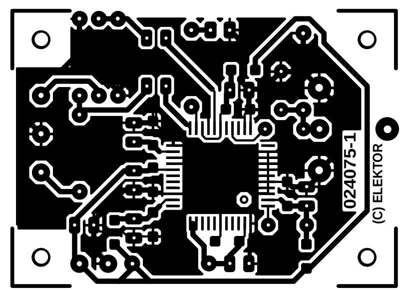


– I_{supply}:	8 mA (no signal, LED off)
	22 mA (f_s = 44.1 kHz)
	33 mA (f_s = 96 kHz)
– Nominal output signal:	900 mV
– THD+N (1 kHz, f_s = 44.1 kHz):	0.0033 % (B = 22 kHz)
	0.04 % (B = 80 kHz)
– THD+N (1 kHz, f_s = 96 kHz):	0.003 % (B = 22 kHz)
	0.011 % (B = 80 kHz)

The graph shows the amplitude characteristic, measured with a test CD. As can be seen, the attenuation is only about 0.5 dB at 20 kHz! The PCB shown here is unfortunately not available ready-made.

(024075-1)





COMPONENTS LIST

Resistors:
R1 = 75Ω
R2 = 1kΩ
R3,R4 = 10kΩ
R5,R6 = 100Ω
R7 = 1Ω

Capacitors:
C1 = 180pF
C2 = 10nF ceramic, lead pitch 5mm
C3 = 100nF ceramic, lead pitch 5mm
C6,C8,C10,C12,C14,C16 = 100nF, SMD shape 1206
C4,C5 = 47µF 25V radial
C7 = 10µF 63V radial
C9,C11,C13,C15,C17,C18 = 100µF 25V radial

Inductors:
L1,L2,L3 = Murata BLM31AJ601SN1L (Farnell # 581-094)

Semiconductors:
D1 = high-efficiency LED, dia. 3mm
IC1 = UDA1351H (Philips)

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

Digital Transformer

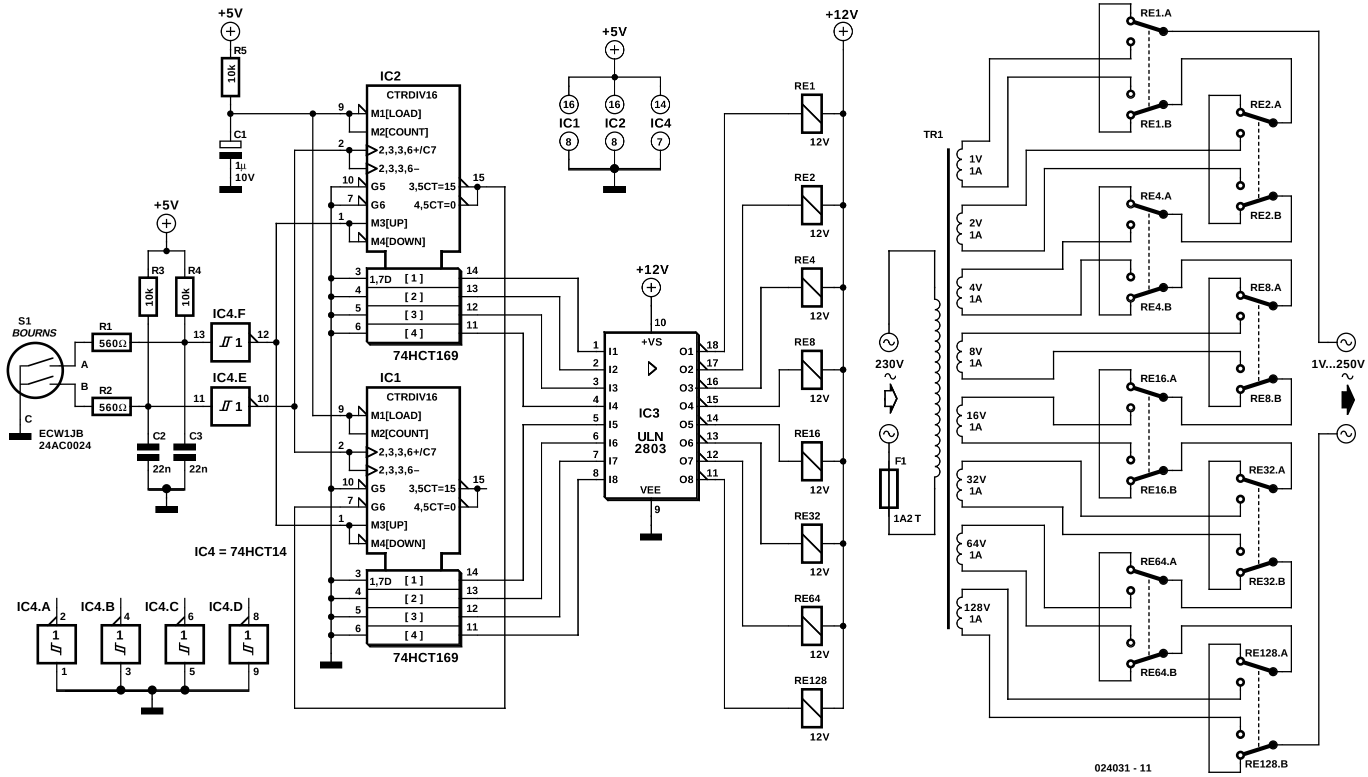
042

D. Barth

An electrically isolated variable transformer is actually an obligatory piece of equipment in every electronics lab. However, many people are put off by the high price of such a device. In a hobby lab, a professional variable transformer is usually grossly overdimensioned, so it's possible to save a lot of money with a little bit of handiwork. All that you need is a transformer kit for the desired power level and a bit of electronics (**Figure 1**). This approach can be used to construct a binary-controlled transformer for ac voltages in the 1 to 255-V range with 1-V steps.

The working principle is based on a binary-valued series of eight isolated transformer secondaries, each having a voltage that is twice that of the previous one (1, 2, 4, 8, 16, 32, 64 and 128). These can be combined to achieve the desired ac output voltage. In order to allow the output voltage to be continuously adjusted (to the extent that this can be said to be the case with 1-V steps), a rotary pulse generator connected to an up/down counter is used. The pulses from the generator are offset with respect to each other, which makes it possible to determine the direction of rotation. After passing through a network that suppresses contact bounces, the pulses are applied to the cascaded counters to cause them to count up or down.

The counter outputs energise a set of relays via the ULN2803



power driver, and the relay contacts tap off the corresponding voltages. The output voltage should be checked by connecting an ac voltmeter.

Please observe the usual safety precautions for equipment carrying mains voltages when building this digital transformer.

(024031-1)

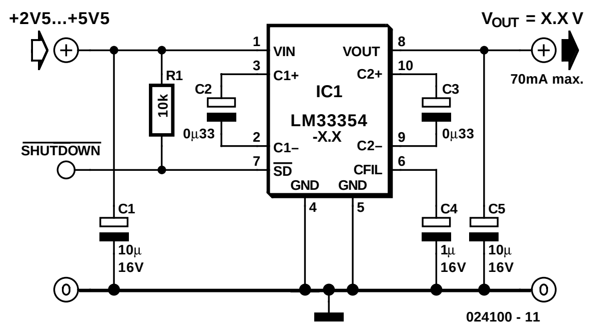
Constant Voltage

043

When batteries are used to power a circuit, there is always the problem that the battery voltage drops during operation. A National Semiconductor type LM3354 IC (www.national.com/0ds/LM/LM3354.pdf) can provide a solution to this problem. It contains a dc/dc converter that operates as a buck converter in step-down mode when the input voltage is too high, and which can also handle input voltages that are too low by operating as a boost converter in step-up mode. All of this works without any coils using switched-capacitor technology. Several capacitors are charged to the input voltage and then connected by an internal matrix switch either in series (step-up) or parallel (step-down). The output voltage is regulated by varying the duration of the switch-on phase. The LM3354 is available in several versions for various voltages.

The IC is clocked at 1 MHz and works with input voltages between +2.5 V and +5.5 V. At maximum output current, the minimum input voltage is 2.9 V (3.4 V for the 5-V version). A Low signal level applied to the Shutdown input can be used to disable the converter. The efficiency of the IC ranges from 75% to 85%. Thermal overload protection prevents damage to the IC from overloads.

(024100-1)



Type	Minimum input voltage	Output voltage	Current (max.)
LM 3354-1.8	2.5 V @ <80 mA	+1.8 V	90 mA
LM 3354-3.3	2.5 V @ <70 mA	+3.3 V	70 mA
LM 3354-4.1	2.5 V @ <40 mA	+4.1 V	90 mA
LM 3354-5.0	2.9 V @ <30 mA	+5.0 V	90 mA

Two Keyboards on one PC

044

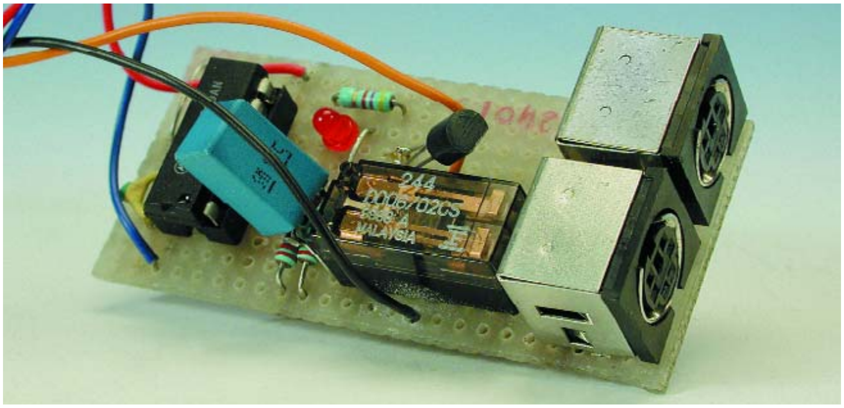
This circuit does the exact opposite of what most of this type of changeover switches attempt to do. Usually a changeover switch is used with two PCs and one keyboard. This version however makes it possible to operate one PC with two keyboards. K1 is connected to the PC for this purpose while K2 and K3 are connected to two keyboards.

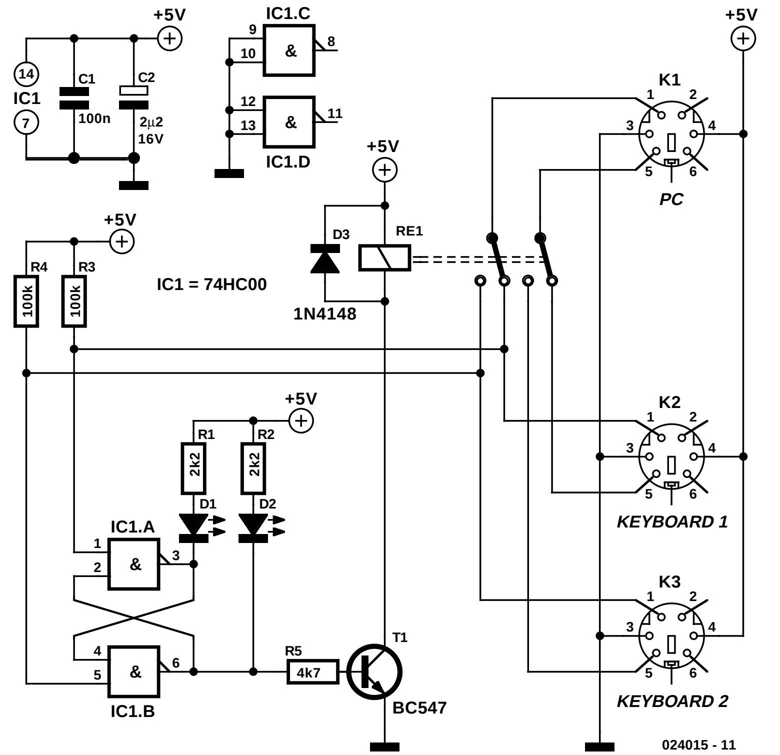
The data outputs of the keyboards are high in the idle state. As soon as a key is pressed the keyboard will serially transmit the data. The data line will now also be low from time to time. This low level is detected and remembered by the flip-flop circuit around IC1.

If the signal originates from K3, the output on pin 6 will be high. Transistor T1 will conduct via resistor R5, which causes the relay to activate. The signals on K3 are then connected straight through to plug K1. This situation will persist until a signal on the data line of K2 is transmitted. In that case the flip-flop will reverse and pin 6 will be low. The signal at resistor R5 will no longer cause transistor T1 to conduct and the relay will be in the rest position. The signals at connector K2 are now connected to plug K1. The LEDs D1 and D2 indicate which keyboard is connected to the PC.

The changeover of the signals via the relay is relatively slow, so the first keystroke is not properly transmitted to the PC. This means that when changing over the first keystroke will always be lost. Also take into account that when the PC is first switched on, the state of the flip-flop is random, so it is not clear which keyboard is initially connected to the PC.

(024015-1)

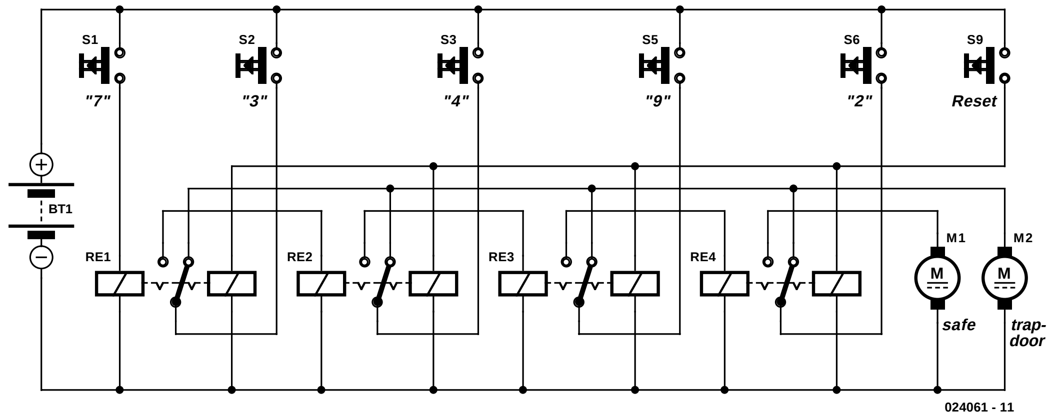




Treasure Chest using Relays 045

B. Kainka

There exist special types of relay that have peculiar characteristics. Next to the 'normal' types in the catalogue we find the so-called 'latching' types which have two stable states. There are two types of latching relay, with either one or two coils. Types with just one coil are reset by applying a reverse-polarity voltage to the coil, while those with two coils are reset by applying a voltage (of the same polarity) to the second coil. It is the version with two coils that we will use in this circuit. It takes only a brief pulse of current to switch the relay's state; then it will remain in that state until reset using the other coil. Suitable latching relays are available, for example, from Conrad Electronics: they offer versions with coil voltages of 6 V (at 110 mA), 12 V (at 50 mA) and 24 V (at 27 mA), with one or two sets of changeover contacts.



The combination lock for a treasure chest can be constructed from such relays, rather than the microcontrollers more commonly employed these days. To open the treasure chest, the rightful owner needs simply to press the keys in a set sequence, whereupon the relays switch over in turn from left to right. After the last press a motor is switched on that operates the mechanism which unlocks the treasure chest. Once the valuables have been taken out, the circuit is reset by pressing the reset button.

The unauthorised intruder, should he dare to try the circuit, will find that as soon as he for example presses button 4, without having pressed 7 and 3 first in the correct order, a second motor starts up — which releases the trapdoor, dropping the intruder into the snakepit below!

(024061-1)

3-dB Chebyshev Filter/Amplifier 046

Elsewhere in this collection of small circuits a 1-dB version of a third-order Chebyshev filter can be found. This 3-dB version is a bit steeper after the corner frequency. The inherent disadvantages are the increased ripple in the pass-band and more ringing in response to a square wave. The indicated frequency is the corner frequency at –3 dB. This time as well, two tables are printed here. **Table 1** shows an implementation with three equal resistors for the filter section and the theoretical values for the capacitors. In the case of a high-pass filter, three equal capacitors can be used. The resulting 'oddball' values can be synthesised by combining values from the E96 range.

Table 2 indicates more practical values for the filter section, by selecting E12 values for the capacitors and theoretical values for the resistors (which can be built by combining 1% components). It is best to use a very fast opamp so that the transfer function is not unduly influenced, particularly with higher gains or at higher frequencies.

(024116-1)

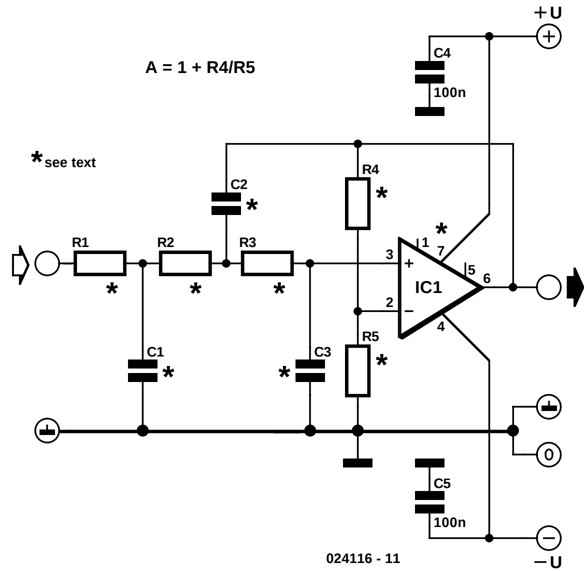


Table 1 : 3x10kΩ, 1 kHz (f_c = –3 dB)

A[dB]	C1	C2	C3
0	57.571 n	619.02 n	403.12 p
5	65.696 n	23.996 n	10.205 n
6	66.756 n	21.061 n	11.442 n
10	70.926 n	14.003 n	16.197 n
14	75.498 n	10.032 n	21.240 n
20	83.776 n	6.3885 n	30.059 n

Table 2 : 1 kHz (f_c = –3 dB), C's : E-12

A[dB]	C1	R1	C2	R2	C3	R3
0	56 n	10.299 k	560 n	10.493 k	390 p	12.171 k
5	68 n	9.3222 k	22 n	11.270 k	10 n	10.235 k
6	68 n	9.8636 k	22 n	9.6024 k	12 n	9.4615 k
10	68 n	10.665 k	15 n	9.5068 k	15 n	8.6424 k
14	82 n	8.7960 k	10 n	10.416 k	22 n	9.7338 k
20	82 n	10.508 k	6.8 n	9.4462 k	33 n	8.8083 k

Super Simple NiCd/NiMH Charger

047

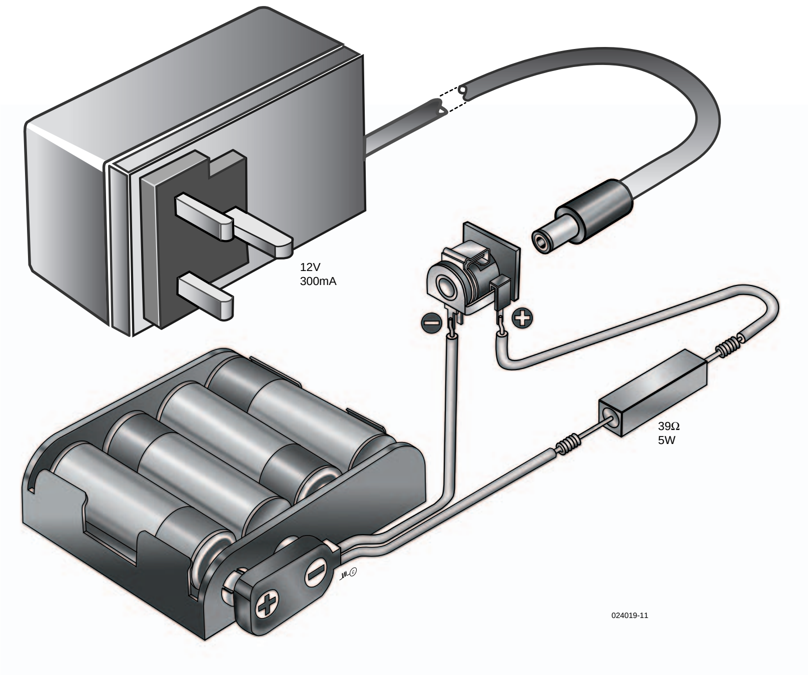
A few years ago the charging of Nickel Cadmium cells was a fairly standard procedure. An AA sized NiCd cell used to have a capacity of 500 or at most 600 mAh. You bought or built a charger that was rated for that capacity and you could use it to charge virtually all AA NiCd cells. The capacity of these popular rechargeable cells has increased enormously, especially since the advent of Nickel Metal Hydride cells. Nowadays AA cells are available in any imaginable capacity up to 1800 mAh (at the time of writing). This is of course a welcome development, but it also introduces a few problems. When you already have a charger for 600 mAh cells it is irritating to have to buy a new charger when you buy some new 1800 mAh cells. To avoid the need to purchase two or three expensive chargers, we'll describe a simple charging process that is suitable for any NiCd or NiMH cell and which can be built in only quarter of an hour.

For a safe charging process we choose a charging current of 1/10 the value of the cell capacity, *C*. Why? At this current the cell can never be damaged, even if it is accidentally left in the charger for several days. At higher currents it can be fatal for the cell if it is overcharged. This can occur when we forget to remove the cell from the charger, but also when the cell is still half full when charging starts. The normal charging period will then be too long as well. At a current of 0.1 *C* these problems won't occur. We don't have to worry about the notorious memory effect; new NiCds haven't had this for several years and NiMH cells never suffered from this in the first place. It is therefore no longer necessary to discharge the cells first. (There are some exceptions to this rule: when large peak currents are drawn from the batteries, such as in model cars and cordless drills, the recommendation is that the cells should occasionally be discharged and charged at high currents.)

After following the next example, anybody should be able to make his or her own charger:

We'll assume that the cell has a capacity of 1800 mAh. That means that the cell will be able to deliver a current of 1800 mA for one hour, or 900 mA for two hours, and so on. It will be charged with a current of 1/10 the current it can deliver for one hour, making 1800/10 = 180 mA. After 10 hours the cell should be fully charged, but because there are always losses and we want to be completely sure that the cell is fully charged, the charging period should be 14 hours. As we said previously, the cell may be charged for longer, but if we charge it for a shorter period it may not be fully charged.

For the power supply we'll use an ordinary 12 V mains adapter. Keep in mind that an unregulated adapter usually has a voltage of 13 V or more; if you want to know the exact figure you should measure it with a multi-meter during the charging process. In this example we'll assume 12 V. To draw a current of 180 mA (= 1/10 C) at a voltage of 12 V Ohm's law tells us that the value of the required resistance is 12/0.18 = 66.7 Ω. But there is also a voltage drop across the cell of about 1.4 V when it is being charged. Keeping that in mind, the resistor should be (12–1.4)/0.18 = 58.9 Ω. The resistor will become rather warm, so we should choose one that can dissipate at least (12–1.4) × 0.18 = 1.9 W. In practice we'll choose a 5 W or even a 10 W type, otherwise it could become too hot.



It is of course not possible to buy a 58.9 Ω resistor, so we have to choose one of the nearest standard values of 56 Ω or 68 Ω. It is always best to choose the slightly higher value (in this case 68 Ω), because there will be a reduced current and that is safer. The calculations don't have to be exact, because the capacity of the cell will most likely differ from that stated by the manufacturer. The cell is put in the battery holder, the resistor and plug are connected and Bob's your uncle. Do take care that the polarity of the connection is correct: the negative terminal of the cell goes to the negative output of the adapter and the positive terminal of the cell goes to the positive output of the adapter via the resistor. If the cell is connected the wrong way round it will be **discharged** instead, and could well become damaged!

It is also possible to recharge more than one cell at a time. Take a battery holder for the correct number of cells and calculate a new value for the resistor. For two cells you should subtract 2 times 1.4 = 2.8 V from the 12 V supply, for three cells 3 times 1.4 = 4.2 V, and so on. The maximum number of cells that may be charged at a time is six; the value of the resistor is then (12–8.4)/0.18 = 20 Ω and the heat dissipated is 0.65 W. In this case we should choose a 22 Ω resistor rated at 5 W, so it won't become very hot.

You may think that you could charge a few more cells, such that the resistor becomes unnecessary; after all, the resistor only wastes energy. But if you try that, you will find that the charging current becomes overly dependent on factors over which you have no control, such as the value of the mains voltage and the charging voltage. We don't mind sacrificing some energy to obtain a stable charging current. That is what the resistor is for.

And finally a warning: Lead-acid batteries and Lithium-Ion cells should **absolutely not** be charged with this charger!

(024019-1)

Optical CD-ROM Output

048

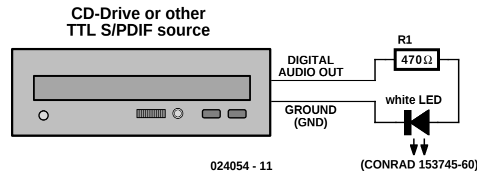
Many CD-ROM drives have in addition to the analogue output also a digital S/PDIF output, in the form of two pins, typically next to the analogue connections and which are generally unused. It is quite straightforward to connect a Toslink module to these pins to add an optical output. The 5-V power supply that this module requires can easily be tapped off the power supply connector of the drive. But it can also be done simpler and cheaper: just connect a series resistor and an LED (**Figure 1**).

It is necessary however, to use an LED that emits light at roughly the same wavelength as a Toslink module (660 nm). An ordinary red LED is fairly close and it appears that others have succeeded in getting an optical connection to work with one of these components (for example, refer to http://members.tripod.com/~Psych/super-cheap-toslink.html). However, a considerable amount of current was required, and in general it is not specified how much current the digital output can safely deliver. Therefore, the author of the circuit proposes to use a buffer IC between the CD-ROM drive and the LED (**Figure 2**).

We experimented in the lab with different red LEDs that we had on hand, but none of them provided satisfactory results. It did work, though, with a super-bright, white, 5-mm LED, the current through which could be reduced to 3 mA. A load current this small is unlikely to worry or damage any output.

Also, Thomas de Bruijn provides on www.minidisc.org/cdrom_opticalout.htm a very good suggestion for an enclosure for the LED: the plastic housing from a 3.5-mm jack plug is eminently suitable. The connector part is removed, which then provides sufficient room in the back shell for the 5-mm LED and the resistor. An additional advantage is that a Toslink connector fits nicely in the thread of the back shell.

1



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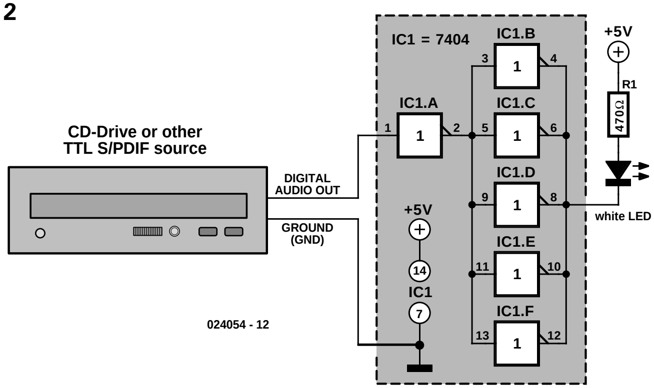


NB: when Windows Mediaplayer version 7 is used to play back CDs, the default value 'digital copying' is set. This means that the PC reads the CD data (music in this case) from the IDE interface and is not available at the S/PDIF output. To solve this, in Mediaplayer go to 'Options' under the Tools menu and select the tab 'CD Audio'. Disable the 'digital copying' option.

(024054-1)

Elektor Electronics USB Interface under Linux

049

H. Jung

With the use of Linux becoming increasingly common, including among the readers of *Elektor Electronics,* there is growing interest in Linux drivers for *Elektor Electronics* interface projects. One of the most successful projects in recent years was the USB interface presented in the September 2000 issue, for which a driver for Windows 98 was developed, but not for Linux. It's time to set that right!

Linux supports USB in kernel versions 2.4.0 and above. For kernel versions 2.2.16 and above, there is also limited USB support. In principle, there are two options for accessing a USB device:

- access via a user-supplied module incorporated into the kernel that communicates with the interface;
- access via the USB file system (usbdebfs), which can also execute the Control Request provided by the USB interface.

Here we take the approa ch of using our own kernel module, which is a piece of code that can be dynamically loaded into the operating system kernel as well as dynamically deleted, and which implements a particular function – such as accessing the *Elektor Electronics* USB interface. This not only makes it very easy to test the code, it also means that the operating system kernel is not burdened with unnecessary code.

The `cy3640.o` kernel module is a modified version of a driver that we found on the Internet for the Cypress Starter Kit, which uses the same hardware as the *Elektor Electronics* USB interface.

This module provides access to all of the functions implemented in the interface using `ioctl routines`, which are special calls for devices that do not fit into the normal read/write scheme. Each `ioctl` employs a 4-byte structure that is used to transfer values and return results. A typical call sequence is shown in **Listing 1**. The calls that have been implemented are listed in **Table 1**.

The file descriptor is hardcoded in the module and employs USB Major Number 180 and Minor Number 128.

In order to activate the module, it must be loaded into the kernel using the command `insmod cy3640.o`. The command `lsmod` can be used to verify whether this was successful. In addition, the file descriptor must be created once using the command

```
mknod /dev/usb-elektor
  c 180 128
```

Listing 1. Typical ioctl calling sequence

```
// build the transfer structure
cmd.val1 = CY3640_READ_ROM;
cmd.val2 = addr;
cmd.val3 = 0;
cmd.val4 = 0;

// call the ioctl
ioctl(fd,CY3640_READ_ROM,&cmd);

// output the results
printf("rom at addr 0x%02x is
0x%20x\n",addr,cmd.val2);
```

Table 1. ioctl functions

Ioctl	In	Out
CY3640_PING	-,-,-,-	status,-,-,-
CY3640_SET_BRIGHTNESS	-,brightness,-,-	status,-,-,-
CY3640_READ_TEMP	-,-,-,-	status,temp_low,temp_high,button
CY3640_READ_PORT	-,port,-,-	status,value,-,-
CY3640_WRITE_PORT	-,port,value,-	status,-,-,-
CY3640_READ_RAM	-,address,-,-	status,value,-,-
CY3640_WRITE_RAM	-,address,value,-	status,-,-,-
CY3640_READ_ROM	-,address,-,-	status,value,-,-

Table 2. Access library functions

Function	Remarks
void set_device (char *device)	Sets the name of the device file. Must always be the first function called.
void brightness (int val)	Sets the brightness of the green LED.
unsigned char read_port (int port)	Reads the specified port.
void write_port (int port, int val)	Writes the specified port.
unsigned char read_ram (int addr)	Reads the specified RAM address.
void write_ram (int addr, int val)	Writes the specified RAM address.
unsigned char read_rom (int addr)	Reads the specified ROM address.
float read_temp (void)	Reads the temperature.
int read_button (void)	Reads the state of the pushbutton.
int ping_device (void)	Tests whether the interface is operational.

and its access privileges must be set to read/write for everyone using the command

```
chmod 0666 /dev/usb-elektor
```

Root privileges are required for these operations.

The module must be reinstalled each time the system is started up. Automatic installation using `usbmgr` or hotplug scripts is not possible.

In order to avoid having to use the rather tedious ioctl programming every time, an access library containing the functions listed in **Table 2** has been created. This hides the `ioctl` calls from the user.

The Linux counterpart to Visual Basic in Windows is called Tcl/Tk. Tk is an interpreter that provides a graphic user interface and into which dynamic libraries can be loaded. These features make it an ideal testing tool for rapidly generating user interfaces.

The download file for this project, number **010065-11**, contains the previously described access libraries and the kernel modules for kernel versions 2.2.x and 2.4.x. It also contains a shared library for use with Tcl/Tk and several demo programs, as well as the illustrated Tcl/Tk application in both German and English versions.

(010065-1)

Recommended literature:

The Universal Serial Bus (USB).
Elektor Electronics September 2000
(description of the USB interface).

www.linux-usb.org
Introductory site for USB under Linux, including access to the USB mailing lists

The Linux USB subsystem, Brad Hards, Sigma Bravo Pty Ltd.

Programming Guide for Linux USB Device Drivers. Detlef Fliegel.
http://usb.cs.tum.edu

A USB Driver for the Cypress USB Starter Kit. Craig Peacock.
www.beyondlogic.org/usb/cypress.htm

Practical Programming in Tcl and Tk. Brent B. Welch.
Prentice Hall, 1999.

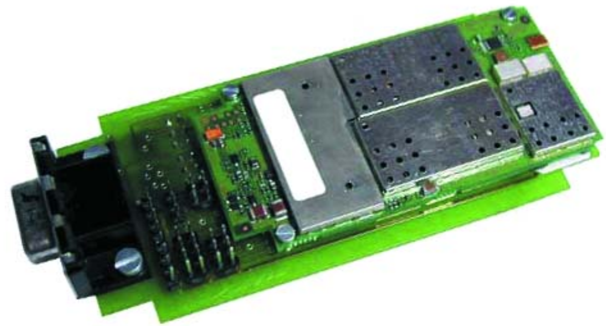
GSM Modem

050

Interfacing your own equipment to GSM devices has been made difficult by the manufacturers because they don't use standard connectors, very little information is made available, and a few other similar reasons. This is a pity since the AT command set has been extended especially for GSM. These AT commands have been used for years in ordinary modems and are fairly easy to use. The arrival of GSM has led to the addition of a number of commands, making the sending of SMS messages and such like possible.

If you tried to interface an off-the-shelf GSM phone to your own circuit, you would first have to find out which signals go to which pins on the GSM connector. If you only need to use AT commands, then it is sufficient to look for the RxD and TxD pins and a ground on this connector. Unfortunately this is only half the work, since you still have to find a plug that fits into the GSM connector. These two obstacles prevent many people from experimenting with GSM. And that is despite the fact that there are many applications where the addition of a GSM link would be very useful.

UbiCom, a German company, has recently begun to market several GSM products, of which the GSM Triband Modem and GSM Dualband Modem are highly suitable for experimentation with GSM. These GSMs have been specifically designed to be incorporated in existing equipment and therefore don't have a display, a keypad or any other unnecessary extras. Obviously this also keeps the cost down, which is a bonus.



These modems can be used with a standard RS232 cable; just connect a power supply and antenna and you're ready to start experimenting! You do of course have to insert a valid SIM card into the GSM.

As far as driving the GSM is concerned you are on your own, but it is not difficult to control the GSM with AT commands using a microprocessor. As an example, this could be used to transmit the current room temperature or to receive an SMS message that causes the microprocessor to set your room thermostat to a specific temperature.

More information about these GSM modems can be found on the Internet (www.ubicom.de). We're not sure how widely these are available, but it should always be possible to order the devices directly from UbiCom. (024105-1)

Audio input Selector

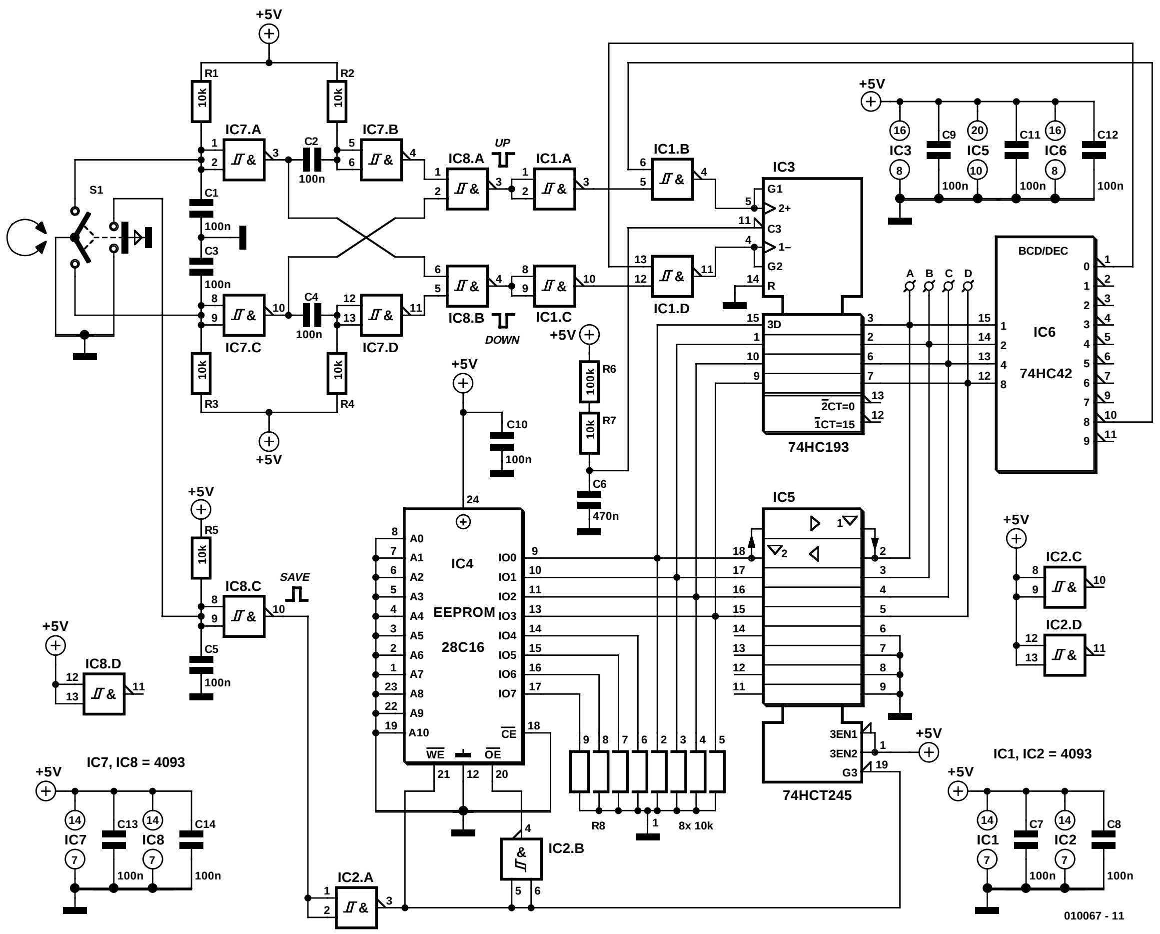
051

F. Lux

This circuit, designed for audio applications, allows the user to select which input signal will be enabled when the system is switched on. The setting is not fixed, but is stored in an EEPROM which is read when power is applied. Fortunately, because of the falling price of EEPROMs, this is not an expensive proposition. The switch used here is a shaft encoder with rotation outputs and a pushbutton (Conrad Electronics catalogue number 70 55 94). The circuit to determine the direction of rotation has appeared before in *Elektor Electronics*. A debounce circuit has been added for the pushbutton. The heart of the circuit is a presettable decimal or binary up/down counter (IC3, either a 74HC193 or a 74HC192) with separate clock inputs.

The clock pulses from the shaft encoder are first inverted by IC1.A and IC1.C. The signals are then passed to NAND gates IC1.B and IC1.D: the other inputs of these gates are used as control inputs to enable the clock pulses to be passed through or not. Assuming the control input is high, the clock signal is passed on to counter IC3. Each negative edge causes the counter to count up or down by one. The outputs of the counter are connected to BCD-to-decimal decoder IC6. The '0' and '8' outputs of this decoder are connected to the control inputs of the NAND gates on the 'down' and 'up' clock inputs respectively. This means that negative edges from the shaft encoder will be blocked when the counter is in the appropriate state, and the counter can therefore only count in the opposite direction.

When the pushbutton on the shaft encoder is pressed the output of the counter at that moment is stored in the non-volatile EEPROM. The pushbutton signal is connected to the enable input of bus driver IC5 via IC2.A, a NAND gate connected as an inverter. If the button is not being pressed, the enable input is high, so all outputs are high impedance, and IC5 is effectively not present. However, when pin 19 goes low the 4-bit value on the output of counter IC3 is driven on to the preset inputs of the counter and on to the I/O pins of EEPROM IC4. At the same time the EEPROM is put into write mode by taking its WE input low and its OE input high. The data are then stored at location 0 (since the address pins A0 to A10 are held low). As soon as the pushbutton is released, the data outputs of the bus driver return to the high impedance state. The EEPROM



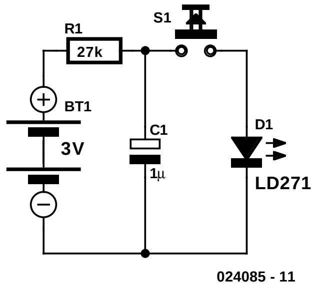
switches back into read mode (with WE high and OE low), and presents the stored counter value to the preset inputs of the counter. As long as the load input of the counter IC3 remains high, the output of the counter is not affected. When the load input goes low, the values at the preset inputs are transferred to the count outputs A to D. This is exactly what happens when power is applied to the circuit: C6 charges slowly, applying an active-low pulse to the load input of IC3. Since the EEPROM is in read mode at power-up, the data stored at address 0 will be available at the preset inputs to the counter and be transferred to the outputs by the low pulse on the load input.

(010067-1)

Simple IrDA Transmitter

052

B. Kainka



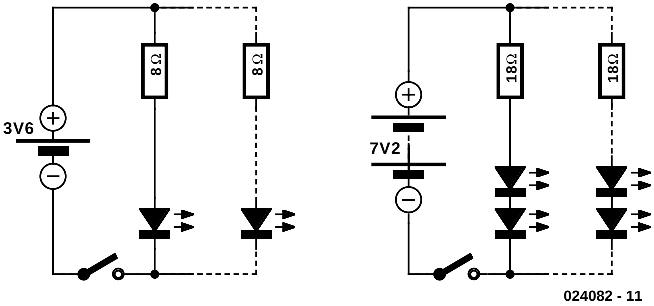
Communication over the IrDA port uses a relatively complex protocol but its possible to send a single character using just a few components. In many cases this will be sufficient to control a Palmtop. This circuit generates a single pulse of IR light that is interpreted by the Palmtop as a data byte with the value 255. The circuit could hardly be simpler, a 1 µF capacitor is charged via the 27 kΩ resistor and when the switch is pressed this charge will flow through the diode to generate an IR light pulse. The values have been optimised for a communication speed of 9600 Baud but the actual pulse length is not critical. The listing shows a simple receiver program for this IrDA signal. The IR interface is opened with the command `ir` and the line `get$(#5,0)` reads a single byte from the interface, if nothing has been received it will return –1. Each time a character is received a counter will be incremented and displayed on the screen and a short sound will be generated.

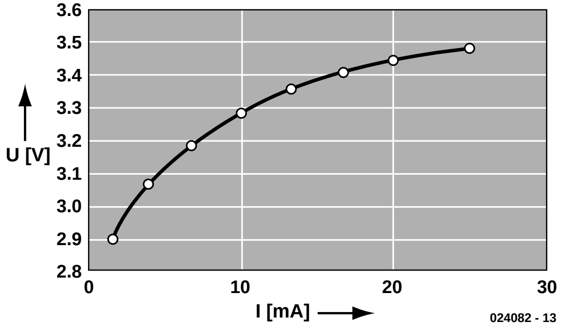
```
#irdacount.bas
open "com1:",9600,ir as #5
z=0
draw -1
while 1
n =get$(#5,0)
if n>-1
  z=z+1
  t$=str$(z)
  draw t$,75,60,2
  sound 800,100,63
endif
wend
```

(024085-1)

Economical White LED

053





The general assumption is that white LEDs operate at a voltage of 3.6 V and a current of 20 mA, which is about right. Lithium Ion cells coincidentally have a voltage of exactly 3.6 V, which seems to be convenient. However, we can't just connect an

LED to a voltage source (the cell), because the current could become too large and the LED could be damaged. That is why they are usually driven by a current source, but the energy dissipated in the current source is of course lost. Besides, a current source can only function properly when it drops a few volts, which we don't have in this case.

But is it necessary to have a 'real' current source? The amount of light given off by an LED is obviously dependent on the current flowing through it, but our eyes are easily fooled. It is easy to tell the difference in brightness between two different LEDs mounted next to each other, but when you turn on an LED momentarily and then turn it on again a bit later at a different brightness you will barely notice the difference. So as far as the eyes are concerned, there is not much difference whether the LED operates at 10, 20 or 30 mA (!). The conclusion is that we don't really need an accurate current source, but that a 'bad' current source will suffice, limiting the current to safe levels.

With that in mind we get a very simple yet efficient design, where the current source consists of a resistor of a few ohms combined with the internal resistance of the LED, which is about 10 ohm at 20 mA. You can add as many branches in parallel as you like.

It can often be difficult to obtain a single 3.6 V cell, but camcorder battery packs with two cells (7.2 V) are widely available. The circuit remains simple at 7.2 V: two LEDs in series with a current limiting resistor of about double the value. Here too you can have as many branches as you like.

To determine the value of the current limiting resistor you should look at the graph, which shows the relationship between the operating voltage and current of a white LED. As an example we'll show the calculations for the current limiting resistor for an LED current of 20 mA: (3.6-3.44) / 0.02 = 8 Ω. So at 3.6 V the current is 20 mA, at 3.7 V it is about 27 mA and at 3.5 V about 16 mA. In practice the values shown in the circuit of 8 Ω at 3.6 V and 18 Ω at 7.2 V may be increased a little; values of 15 Ω and 33 Ω respectively still work well.

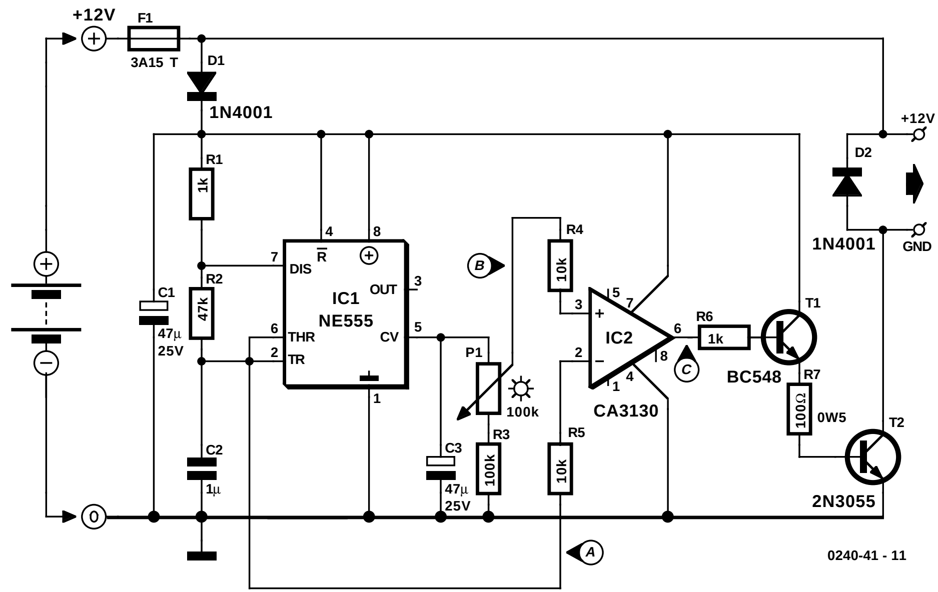
(024082-1)

Direct Current Dimmer

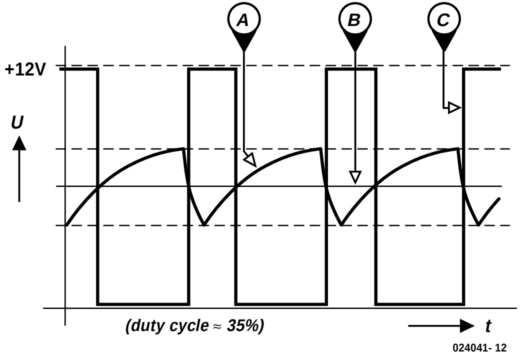
054

A. Schilp

This energy saving, 12-V controller is nearly universally applicable. In addition to controlling battery powered lighting in a car, boat or caravan it is also quite able to control the speed of a model train. In essence, this circuit converts the 12-V DC voltage into a rectangular pulse train with a duty cycle that is adjustable from 0 to 100%. The circuit can be divided into four sub-circuits: the saw-tooth generator built around IC1, reference network P1/R3/C3, comparator IC2 and driver stage T1/T2. The comparator compares the generated sawtooth voltage (1) with the reference voltage (2). This reference voltage is adjustable between the lower and upper limits of the sawtooth voltage with P1. When the saw-tooth voltage is greater than the reference the output of the comparator will be 'high'. Since the saw-tooth voltage, with its fixed frequency, is continuously crossing the reference voltage, a rectangular waveform (3) appears at the output of the comparator, the duty cycle if which can be determined with P1.



The driver stage, with its large current amplification, ensures that the voltage up to a load current of 3.15 A will remain sufficiently square. By varying the reference with P1 we can change the width of the pulses in the pulse train. This affects the average voltage to the load and therefore also the power drawn by it. The current through T2 is always largest when the voltage drop across it is smallest (saturation) and smallest when the voltage drop is the greatest. T2 therefore, needs to dissipate only very little power and needs to be cooled only when used with highly inductive loads. Diode D2 protects



against reverse connections and acts as freewheeling diode for inductive loads.

Returning to the sawtooth generator, IC1 is a 555 timer configured as an AMV, which is tuned to 65 Hz with R2/C2. With such an AMV the square wave at pin 3 is typically used, but this time we are more interested in the charge/discharge voltage across C2. This is, strictly speaking, not a pure sawtooth but is nonetheless very suitable as a sawtooth for this controller. If it turns out that the controlled lamp flashes visibly, it is possible to raise the frequency by lowering the value of C2. Because of the behaviour of the load and the dissipation of T2 it is not recommended to increase the frequency beyond 200 Hz, even though the circuit will work without problems at frequencies greater than 10 kHz.

C2 is charged and discharged by the 555 between the bottom limit of 1/3 and the upper limit of 2/3 of the power supply voltage. These limits are defined by three internal resistors of 5 kΩ each. These also give the IC its name. In order to adjust the frequency the upper limit is made available to the outside world via the control input (pin 5). This voltage is stabilised by C3 and directly made available to P1. The lower limit is defined by making the resistance of P1 and R3 equal, so the voltage division is the same as the internal resistors in the 555, with which they are effectively in parallel. (024041-1)

LED Voltage Tester

055

B. Kainka

A universal voltage tester should respond to both dc and ac voltages. The usual types with glow-discharge lamps only work with voltages greater than around 100 V. The circuit shown in **Figure 1** uses a Darlington circuit formed by two NPN transistors and can detect voltages of less than 1 V. It can also be used to test continuity. Here the positive terminal of the battery serves as the 'ground' connection. Consequently, an input current flows even with a high-impedance connection, but this current increases when a voltage source with the proper polarisation is included in the loop. A supplementary piezoelectric buzzer allows the circuit to also be used as an audio-frequency signal tracer.

The tester can be used in the following manners:

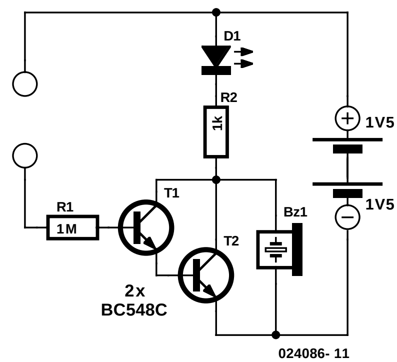
– Continuity between the two terminals or connection via the fingers: the LED lights.
– Testing a battery with the positive terminal connected to the input: the LED becomes brighter.
– Testing a voltage with the negative terminal connected to the input: the LED becomes darker or is off.
– With an ac voltage, the LED current is modulated, so the LED flickers and the buzzer sounds.

All of this can be built into the enclosure of a key finder, since the essential components are already present: a battery holder, a LED and a piezoelectric transducer. Alternatively, the tester can be fitted into the case of a ballpoint pen or a length of plastic tubing (see Figure 2).

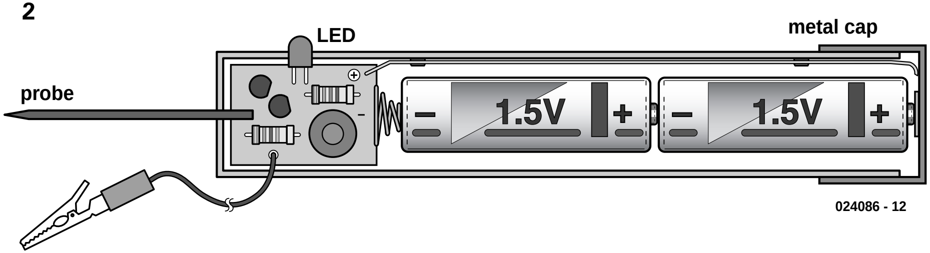
An interesting experiment can be performed using this circuit. One person holds the probe tip, while a second person holds the opposite terminal. If they walk over a carpet or synthetic floor covering, the LED lights up with each step. This is a consequence of charge separation between the floor and the shoes.

(024086-1)

1



2



Instrumentation Amplifier

with an electrically isolated input

By B. Schädler

The instrumentation amplifier described here has two important features. Its input is electrically isolated and it has no fewer than 16 input and output ranges that can be selected for many types of signal conversion.

Most commercial instrumentation amplifiers provide a number of input ranges, but their outputs are usually restricted to one or two voltage ranges and a similar number of current ranges. In this circuit there is a wide choice of input and output ranges: seven unipolar voltage ranges (varying from 0 to 100 mV to 0 to 10 V), seven bipolar voltage ranges (±100 mV to ±10 V) and two current ranges (0 to 20 mA and 4 to 20 mA). The output can be chosen to follow the input on a 1:1 basis, but conversion between any of the ranges is also possible.

Selecting the input and output range is simply done using two DIP switches. The accuracy of all ranges is very good; with careful selection of the resistors a precision of 0.1% is achievable. Two voltage references (IC12 and IC13) are used to maintain accuracy, and presets are used at several critical points.

The electrical isolation is achieved using an optoisolator made by HP (HCNR200 or -201). Its linearity is 0.01%. An alternative part would be the Siemens IL300.

The circuit can be split into three parts:

1) Input stage (with quad opamp IC1) and optoisolator IC3.
2) Amplification and conversion of the optoisolator output into a voltage output.
3) Current output stage.

Input stage

S1A is used to select between a voltage and current driven input. The open position corresponds to the voltage input. The input impedance then is about 1 MΩ, determined by resistors R3 and R1/R2. Capacitor C1 is used to suppress any spikes in the input signal, although a suppression diode could be used instead. Closing S1A changes the input to current mode with an impedance of 50 Ω.

IC1.B functions as a buffer, with IC1.A providing gain/attenuation as required. The gain is changed using switches S1B through S1E. Switch S1F is opened when a current range of 4 to 20 mA is required. For all other ranges the negative voltage reference is connected to the output of IC1.B via R8 and has a negligible effect.

All unipolar inputs are converted to a range of 0 to –1 V at the output of IC1.A; bipolar inputs become +1 V to –1 V. The next stage around IC1.C is used to turn a bipolar ±1-V-signal into a voltage of 0 to +1 V. When S1G is open the –1 V-signal is simply inverted by IC1.A; with the switch closed the gain of IC1.C is halved, while at the same time a reference voltage is added via R16/R17, causing the output to be in the range of 0 to +1 V.

This is followed by the optoisolator (IC3). The internal LED illuminates two photodiodes. One of these is connected back to the input of the output opamp; the other goes to the inverting input of opamp IC1.D. During normal operation the current through R18 and R19 is 'neutralised' by an identical current, but of opposite polarity, delivered by the photodiode. In other words, the opamp drives the LED such that the above condition is satisfied. Assuming that no current flows into the input of the opamp, the photodiode current will be equal to the driving voltage (0 to +1 V) divided by R18+R19. The photodiode current therefore varies from 0 to 50 µA, a value that keeps IC3 operating in its optimum linear range.

Since any current flow into the inverting input of the opamp affects the linearity of the whole circuit, a type has been chosen with a very low input bias current: the OP497 made by Analog Devices. The LT1097 made by Linear Technology is also suitable. If high linearity is not so important it is possible to use a cheaper FET opamp, such as the TL074.

Output stage

The second photodiode in IC3 is connected directly to the inverting input of IC2.A. Since there will be slight differences in the currents sourced by the diodes in the optoisolator, the

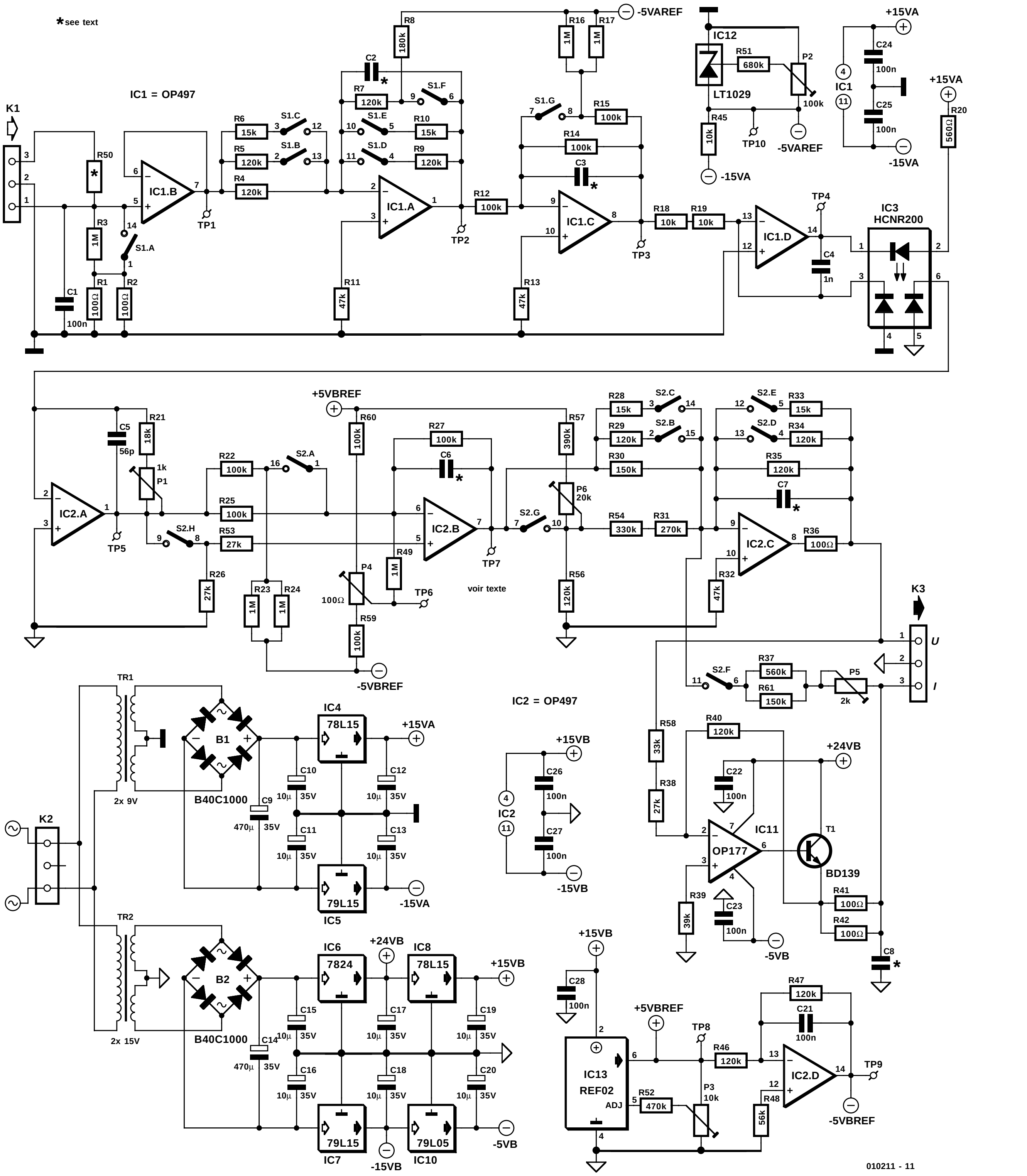


Figure 1. The input of the instrumentation amplifier is electrically isolated from the output by optoisolator IC3.

gain of this opamp has been made adjustable by the addition of P1. When correctly adjusted, the output of IC2.A will be 0 to +1 V.

The next stage with IC2.B is used to give either a unipolar or bipolar output. With S2A open the output is a unipolar 0 to –1 V, with S2A closed the output is a bipolar –1 V to +1 V. At this point any offset errors introduced in the amplifier can be minimised by P4.

The voltage output stage is more or less the same as that at the input; S2B - S2E set the required gain/attenuation. When the voltage output is selected S2G needs to be closed and S2F open. With S2F closed and with the gain of the voltage output stage set at 0.5, IC11 and IC2.C form a voltage to current converter. The gain of IC11 itself is set at a factor of 2.

Table 1. Setting the DIP switches for the various input and output ranges.								
Input		**S1-G**	**S1-F**	**S1-E**	**S1-D**	**S1-C**	**S1-B**	**S1-A**
0..+10V		0	1	1	1	0	0	0
0.. +5V		0	1	1	1	0	1	0
0.. +2V		0	1	0	1	0	0	0
0.. +1V		0	1	0	0	0	0	0
0.. +500mV		0	1	0	0	0	1	0
0.. +200mV		0	1	0	1	1	1	0
0.. +100mV		0	1	0	0	1	1	0
-10V.. +10V		1	1	1	1	0	0	0
–5V.. +5V		1	1	1	1	0	1	0
–2V.. +2V		1	1	0	1	0	0	0
–1V.. +1V		1	1	0	0	0	0	0
–500mV.. +500mV		1	1	0	0	0	1	0
–200mV.. +200mV		1	1	0	1	1	1	0
–100mV.. +100mV		1	1	0	0	1	1	0
0.. 20mA		0	1	0	0	0	0	1
4.. 20mA		0	0	0	0	0	0	1
Output	**S2-H**	**S2-G**	**S2-F**	**S2-E**	**S2-D**	**S2-C**	**S2-B**	**S2-A**
0..+10V	0	1	0	0	0	1	1	0
0.. +5V	0	1	0	0	1	1	1	0
0.. +2V	0	1	0	0	0	0	1	0
0.. +1V	0	1	0	0	0	0	0	0
0.. +500mV	0	1	0	0	1	0	0	0
0.. +200mV	0	1	0	1	1	0	1	0
0.. +100mV	0	1	0	1	1	0	0	0
–10V.. +10V	0	1	0	0	0	1	1	1
–5V.. +5V	0	1	0	0	1	1	1	1
–2V.. +2V	0	1	0	0	0	0	1	1
–1V.. +1V	0	1	0	0	0	0	0	1
–500mV.. +500mV	0	1	0	0	1	0	0	1
–200mV.. +200mV	0	1	0	1	1	0	1	1
–100m.. +100mV	0	1	0	1	1	0	0	1
0.. 20mA	1	1	1	0	1	0	0	0
4.. 20mA	1	0	1	0	1	0	0	0
1 = closed								

The reason for the choice of these gains lies with the supply voltage of IC11: in order for the current output to drive a large resistive load (at least 1 kΩ), it needs a high supply voltage of +24 V. There is therefore no choice but to use a reduced negative supply of –5 V, which obviously restricts the negative output swing of the opamp. It is for this reason that the gain of IC2.C is set at a factor of 0.5. Since IC11 is only required to supply a positive output and its output swing is very large, setting the gain of IC11 to 2 causes the overall gain of the current source to be unity.

When calibrated correctly, the maximum output current depends only on the driving voltage (0 to +1 V) of the current source and parallel resistors R41/R42 (50 Ω). Another condition is that the feedback resistor has to be exactly 120 kΩ – 50 Ω; which can be adjusted precisely with P5.

To turn the current mode of the amplifier to 4 to 20 mA, switch S2G is opened, causing a small voltage to be added to the input of the current source. Since the maximum output current should remain at 20 mA, the gain is adjusted slightly at the same time.

Since the current source requires a positive input voltage, inverting opamp IC2.B is effectively bypassed by closing switch S2H. Preset P6 is used to set the range exactly to 4 to 20 mA.

And finally, capacitors C1 -C7 should be mentioned. These capacitors are used to limit the bandwidth of the circuit. With a value of 1 nF the bandwidth is about 100 Hz and with 100 pF it is about 10 kHz, with an input signal of ±10 V. Choosing too small a value for C4 and C5 can give rise to unwanted oscillations, with C4 being very critical.

Power supply

As shown in the circuit diagram, the power supplies for the input and output stages are completely isolated. The supply for the input stage is kept fairly simple since only a symmetrical ±15 V is required. This is provided by a standard circuit using two voltage regulators (IC4 and IC5). The supply for the output stage consists of four voltages: ±15 V for IC2 and +24 V/–5 V for IC11. As can be seen, the rectified voltage of TR2 is first fed to IC6 and IC7, providing +24 V and –15 V. These supplies are then fed to IC8 and IC9 to give +15 V and –5 V.

The power consumption of the complete circuit is very low, making it possible to use small 1.5 VA PCB mounted transformers for TR1 and TR2.

Calibration

As far as the construction of this circuit is concerned, it is very much a DIY project. The author did design a PCB for his own use, but it had undergone so many modifications during the development of the circuit that the layout was no longer suitable for reproduction. Aspiring hobbyists will therefore have to design their own PCB for this circuit.

The calibration of the circuit is a meticulous task, which should be followed carefully. Fortunately, it isn't very difficult.

1) Switch on the amplifier and all calibration instruments (power supply, multimeter) and let them settle for about fifteen minutes.
2) Adjust P2 and P3 to set the voltage references to their nominal value (test points TP8, TP9 and TP10). Remember that there are two ground points!
   - Adjust P4 for an offset of 0 V at its wiper (TP6).
3) Use Table 1 to set the DIP switches for an input of 0 to +1 V and connect a voltage of 1.000 V to the input. Check that TP1 is +1.0 V, TP2 is –1.0 V and TP3 is +1.0 V.
4) Connect the multimeter to the other ground and set S2 to an output range of 0 to +1 V.
5) Adjust P1 to get +1.0 V at the output of IC2.A (TP5).
6) Short the input to ground. Use P1 and P4 to adjust the output to its exact value. Do the same again with an input of 1.000 V. The voltages at TP7 should now be 0 to 1 V for the unipolar range and +1 V to –1 V for the bipolar range.
7) With an input range of 0 to +1 V switch over to a current output and connect an ammeter between pin 3 of K3 and ground. Adjust P5 with a minimum and maximum load (1 kΩ) to give a 20 mA output current. Use P4 to set the 0 mA point.
8) Switch over to the 4 to 20 mA range and adjust P6 to give exactly 4 mA. If necessary, P1 and P4 can be adjusted to provide the best accuracy.
9) With bipolar inputs and outputs use P4 to adjust the offset to 0 V (input at 0 V). Next, adjust the gain (P1) with a positive input, then connect a negative input, but don't check the output. Again (with the input at 0 V) adjust the offset and with a positive input, the gain. Step 9 should be repeated until the best accuracy has been achieved.

Finally

The ground reference point for input signals and the multimeter should always be that at the input connector. The measurement of the input voltage should also be taken at the input connector. It is recommended to check that none of the opamps is oscillating. Especially IC1.D and IC2.A can be sensitive to this; increasing C4 and C5 in value should help.

If required, R41/R42 can be changed to select different output current ranges. Do take care that the supply transformer is rated for that current.

By choosing different values for resistors R1, R2/R3, as well as R50, virtually any input voltage can be measured.

(010211-1)

Despite the fact that the lack of a PCB prevented us from testing this circuit thoroughly in the Elektor labs, we felt that this design was worthy of publication. We can't comment directly from our own experiences on the usefulness of this circuit. Theoretically, the circuit seems to be in order and it is clear that the author has put a great deal of thought into the design. (Ed.)

Speed Measurement System
March 2002, p. 12-17
(010206-1).

In circuit diagram no. 010206-11 (lower part of Figure 4), the pin numbers on connector K1 should be mirrored, i.e., pin 1 should be pin 5, pin 2 should be pin 4, etc. In diagram 010206-12 (upper part of Figure 4), the designations "IR sender" and "IR receiver" should be transposed.

These corrections do not affect the PCB designed and supplied for the project.

DTMF Codelock
March 2002, p. 56-57
(010110-1).

Pins 4 and 5 of comparators IC3-IC6 should be connected to ground, not to +5 V as shown in the circuit diagram.

DCI PLC
June 2001, p. 10-18
(000163-1)

The article text fails to mention that connector K1 is mounted at the underside of the board.
The photographs on pages 16 and 17 show an early prototype of the DCI PLC with K1 mounted at the solder side.

Serial Interface
for Dallas 1-Wire Bus
April 2002, p. 30-34
(020022-1)

On the component overlay (Figure 2), the labels for solder pins 'D' and 'ground' should be transposed.

The content of this note is based on information received from manufacturers in the electrical and electronics industries or their representatives and does not imply practical experience by Elektor Electronics or its consultants.

QV381m0 Record/Playback Module

meets the Basic Stamp 2

QV381m0 designed by James Evans Electronics (UK), manufactured by Quadravox Inc. (USA)

The QV381m0 is the latest in a family of speech recording and playback modules designed to complement the existing range of Quadravox speech products.

The QV381m0 module employs a sophisticated micro-controller running a dynamic message allocation algorithm, which in-conjunction with the ISD 5008 device offers the user considerable quality and flexibility. The circuit diagram of the module is shown in **Figure 1**.

The module permits messages to be to be recorded at different sampling rates. The sampling rate selected will depend very much on the desired quality of the sound being reproduced and also the length of the required message/messages. The sampling rate can be varied from message to message as required. After having selected a suitable sampling rate the user can record, playback and erase individual messages on command.

The user can select a unique message number to record in the range 1-255, which can be of any length within the limits of the ISD total storage capacity; e.g. one long message or hundreds of short messages, which fill the device to capacity.

The user has the ability to remotely control the playback volume and also has control over an external amplifier (if required) via the

/$\overline{\text{PWR}}$ and AUX pins. For battery powered applications the module can be put into a sleep mode to conserve power.

QV381m0 control options

The Record/Playback module may be controlled by a PC, via the QV430P Programming Cradle which is also supplied by Quadravox.

The QV430P programming cradle is a convenient and *deluxe* means of making the necessary connections to program and test Quadravox ISD-based playback modules. Used in conjunction with the QV381 test program, it enables your PC to configure the module and to record and playback phrases . Additionally an enhanced module configuration program, known as QV300s2, which is currently able to configure several other modules within the Quadravox range, will be available for the QV381m0. The actual recording algorithm is present in every QV3xx

Table I

Command	Description
234 (0xea)	Query module identification
235 (0xeb)	Query current volume level
236 (0xec)	Query current sampling rate
237 (0xed)	Query next available message number
238 (0xee)	Query number of clusters available
239 (0xef)	Query duration of recorded message
240 (0xf0)	Play to speaker
241 (0xf1)	Play to auxiliary output
244 (0xf4)	Record from microphone
245 (0xf5)	Record from auxiliary input
246 (0xf6)	Toggle $\overline{\text{PWR}}$ pin
247 (0xf7)	Toggle AUX pin
249 (0xf9)	Set sampling rate
250 (0xfa)	Sleep mode
251 (0xfb)	Erase message
252 (0xfc)	Set volume
253 (0xfd)	Format device
230 (0xe6)	Stop playback or record

Error /Response Codes	
220 (0xdc)	Invalid number
221 (0xdd)	Message number does not exist
222 (0xde)	Message number to record already exists
223 (0xdf)	No free message numbers
224 (0xe0)	ISD device is full; no free clusters
254 (0xfe)	OK – Confirmation

Features

- Totally dynamic message recording allowing messages to be of variable length.
- Intelligent message allocation recording and playback algorithm
- 255 discrete message capacity
- Individual message erase function
- Digital volume control — 8 discrete steps
- User selectable sampling rate 4.0 kHz, 5.3 kHz, 6.4 kHz, 8.0 kHz
- Serial interface — simple RS232 (TXD & RXD)
- Controlled using simple 2-byte commands
- User controllable /$\overline{\text{PWR}}$ and AUX outputs to control external amplifiers.
- Sleep function for very low power applications
- QV430P programming adapter may be used for pre-programming the prompts
- free recording software (QV381) from Quadravox website.
- small size: 58.4 x 45.7 mm

module; the 430P is a 'dumb' adapter for electrical and logical connections.

Alternatively, the module may be connected using a simple TxD/RxD link to your PC's serial port or another microcontroller circuit. The voltage levels have to be limited +5 V/0 V.

This Application Note, however, will explore the connection of the QV381m0 to the extremely popular **Basic Stamp 2 from Parallax**. Here, it is assumed that the BS2 is plugged in on the Board of Education (BoE).

QV381m0 command set

Table 1lists all the command and error/response codes that are required to control the module and correctly interpret its response. All commands sent to the module consist of two bytes. The first byte send will always be one of the above commands. The value of the second byte sent will depend on the command. The function of each of these commands is explained in detail in the *QV381m0 Users Manual* which is available for downloading from the Quadravox website. It is assumed that a PC (Personal Computer) is being used to control the module.

If an LED is connected to the $\overline{\text{LED}}$ output of the module it will illuminate for the duration that a task is being carried out. The $\overline{\text{BUSY}}$ line on the module will also go low for over the duration that it is carrying out a task and is unable to respond to serial commands.

For commands of very short duration the LED will only 'flicker'.

Interfacing the Basic Stamp 2 to the QV381m0

The QV381m0 is extremely well suited to stand-alone operation in conjunction with a microcontroller. A great candidate for this function is the popular Basic Stamp 2 (BS2) which may be programmed in… yes, BASIC! Here, the wonderful **Board if Education** (BoE) is used to host the BS2. The BoE is available from Parallax Inc. and its distributors (UK: Milford Instruments). It was prominently featured in 'BASIC Stamp Pro-

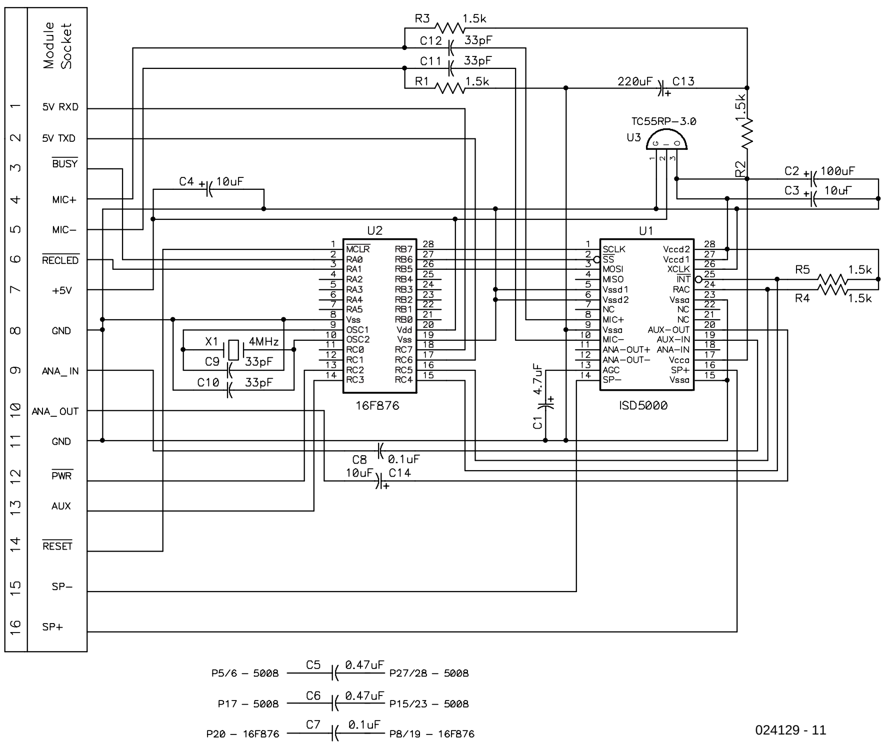


Figure 1. Circuit diagram of the QV381m0 Record/Playback module.

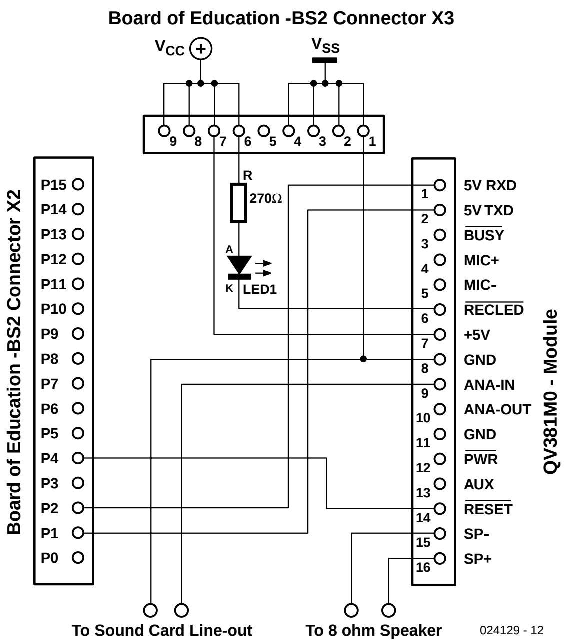


Figure 2. Basic Stamp Board of Education BS2 connection details to QV381m0 Record/Playback module.

gramming Course (1-8)' published in the September 1999 through April 2000 issues of *Elektor Electronics*. The BS2 was described in 'More Power and Performance from the BASIC Stamp', see *Elektor Electronics* July/August 2001.

As seen in the schematic diagram in **Figure 2**, only a simple interface is required to control the module. The Basic Stamp 2 test program (discussed further on) can be easily modified or expanded and provides an easy starting point for user experimentation.

All connections to the QV381m0 module are brought out to its 16-pin edge connector. These are marked on its PCB. and when compared with the schematic, their physical position relates directly to the module.

The schematic diagram assumes that the Basic Stamp 2 is used in conjunction with the Board of Education and its associated hardware (breadboard area and regulated 5V supply).

The QV381m0 can be directly plugged into the breadboard area on the BoE.

The interface between the BS2 and the module makes use of three port pins on the Basic Stamp 2. These are configured for use as the serial transmit and receive lines to communicate with the module and a reset line to initialise its microcontroller.

Connector X3 on the Board of Education can provide the power and ground connections for the module, as shown in the schematic diagram.

Only an additional LED is required to provide a visual feedback to the user.

The LED will require a current limiting resistor (e.g. 270 ohms as shown in the schematic) and should be connected with its anode to +5V, since it is active low in operation.

In order for the module to record sound, an audio source can be connected to its ANA-IN input or a suitable electret microphone can be connected to its MIC+ and MIC– pins.

The ANA_IN audio input can be directly connected to the analogue line- out of a sound card of a personal computer with a suitable 3.5-mm stereo jack plug and short length of screened cable, which can easily be constructed.

Since the line-out jack socket from a PC sound card is typically stereo and the QV381m0 module records only a single channel, only on a single signal and ground connection will need to be made to the stereo jack plug. The effect of playing back the sound to a single channel can be compensated for with the software, which comes with the users sound card. Some experimentation may be required to determine the optimum recording level on the module to avoid distortion.

The flying end of the cable will require some sort of termination so that it can be plugged into the breadboard area to make connections to the ANA-IN and GND pins of the module. Individual header pins would be ideal for this purpose.

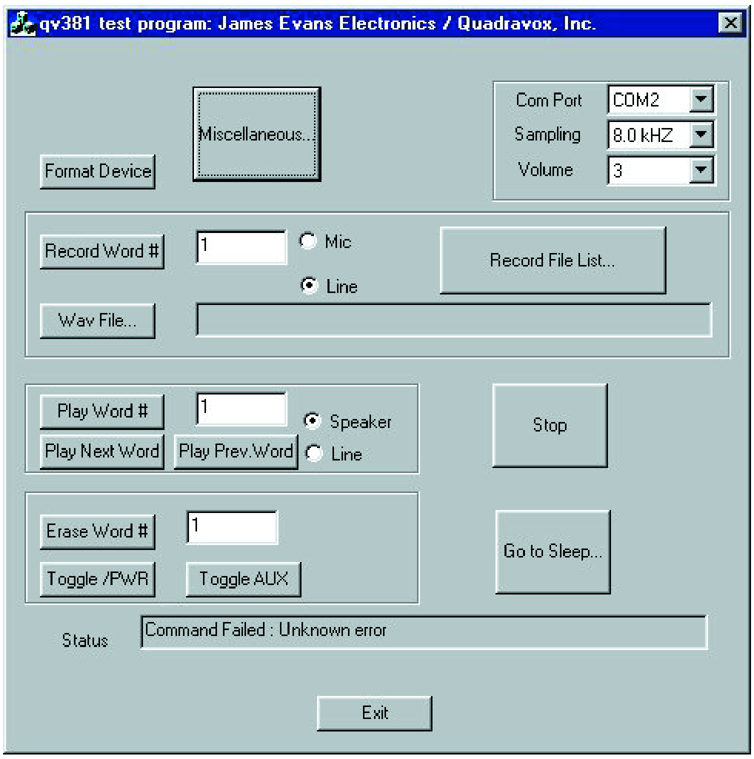


Figure 3. The QV381m0 comes with this test program called QV381. The frugal main control screen belies the power of this software utility.

The test program

The test program shown in **Listing 1** assumes the module will record audio from the ANA-IN input, and will require a change to one instruction in the test program to record from the microphone.

If the user prefers to record audio from a microphone, a sub-miniature electret microphone is ideal (See QV381m0 detailed pin descriptions datasheet).

Connections can easily be made by soldering short lengths of solid tin copper wire to the microphone so that it can be plugged directly into the breadboard area, to make the appropriate connections to the modules MIC+ and MIC– pins (observing the polarity, — one connection of the microphone will go to its metal casing and can usually be seen, otherwise it can be checked for continuity with a multi-meter. This terminal should connect be the MIC– pin on the module).

To make the test program record from the microphone, find the subroutine in the test program called

```
do_record:
```

then change the first line, which reads

```
serout
TRANSMIT_LINE,NBAUD9600,
[QV_RECFRAUX,MessNum]
```

to

```
serout
TRANSMIT_LINE,NBAUD9600,
[QV_RECFRMIC,MessNum]
```

The only change is the command that is sent to the module — sound will now be recorded from the microphone input.

In order to hear the playback, a suitable 8-ohm loud speaker can be connected to the SP+ and SP– pins of the module (See QV381m0 detailed pin descriptions datasheet).

The Basic Stamp 2 test program assumes that sound will be played back to a speaker. Alternatively, sound can be played back to the ANA_OUT connection on the module. The ANA_OUT connection can be directly connected to an external amplifier as required (see QV381m0 detailed pin descriptions datasheet). To make the module playback to the ANA_OUT pin, find the subroutine called

```
do_play:
```

and change the first line, which reads

```
serout TRANSMIT_LINE , NBAUD9600
,
[QVPLAY2SPEAKER , MessNum]
```

to

```
serout TRANSMIT_LINE , NBAUD9600
,
[QVPLAY2AUX , MessNum]
```

The only change has been to the command

Listing 1

```
'QV381M0 Test Program for Basic Stamp-2
'================================================
'Quadravox, Inc (C) 2002 ,J.E.E. (C) 2002

'This Program tests some of the features of the
'QV381M0 Record/Playback Module

'Please see schematic diagram for connection
'details between the QV381M0 and
''Board of Education'
'incorporating the 'Basic Stamp-2

'{$STAMP BS2}

RECEIVE_LINE   con     1
TRANSMIT_LINE  con     2

RESET_LINE   con  4
NBAUD9600    con  84
SMP8KHZ      con  0
SMP6p4KHZ    con  1
SMP5p3KHZ    con  2
SMP4KHZ      con  3

'===========================================
' I/O Definitions

 out2 = 1              'Transmit line high
 output TRANSMIT_LINE
 out4 = 1              'Reset high
 output RESET_LINE
'===========================================
' Aliases for QV381M0 commands
'
QV_QUERY_ID          con  $EA
QV_QUERY_VOLUME      con  $EB
QV_QUERY_SAMPLING    con  $EC
QV_QUERY_MSG_NUM     con  $ED
QV_QUERY_NUM_CLUST   con  $EE
QV_QUERY_DURA        con  $EF
QV_PLAY2SPEAKER      con  $F0
QV_PLAY2AUX          con  $F1
QV_RECFRMIC          con  $F4
QV_RECFRAUX          con  $F5
QV_TOGGPWR           con  $F6
QV_TOGGAUX           con  $F7
QV_SETSMPFR          con  $F9
QV_SLEEPMODE         con  $FA
QV_ERASEMSG          con  $FB
QV_SETVOL            con  $FC
QV_FRMTDEV           con  $FD

QV_OK                con  $FE
QV_INV_NUM           con  $DC
QV_NOMESSNUMEX       con  $DD
QV_MESSNUMEX         con  $DE
QV_NOFREEMESSNUM     con  $DF
QV_NOFREECLUST       con  $E0
QV_STOP              con  $E6

'===========================================
' variables
'
serData     var  byte
serData1    var  byte
MessNum     var  byte
qvVolume    var  byte
qvSampling  var  byte

'===========================================
' Program start
'===========================================


debug  cls
debug "QV381M0 Demonstration ",CR
debug "———————————",CR
```

```
gosub do_reset                'Reset the module
gosub do_format               'Format the module
for first use
gosub query_msg_num           'Return the next
avaialble message number
gosub query_id                'Return the module
identification number

qvVolume=0                    '0=loudest, 7=qui-
etest
gosub do_set_volume           'Set the playback
volume

gosub query_volume            'Return the current
volume setting

qvSampling=SMP8KHZ            'Set the sampling
rate to 8Khz
gosub do_set_sampling

gosub query_sampling          'Read the Sampling
rate setting back

MessNum=1                     'Record   message No
1
gosub do_record               'Record   from line-
in for 10 seconds

MessNum=1                     'Playback message No
1
gosub do_play                 'Playback to speaker

MessNum=1
gosub do_erase_mess           'Erase Message   No 1


debug "End Of Program",CR

STOP

endProgram:


'==================================================
'Error Conditions
'==================================================
playNok:
debug "Play Command Refused",CR
stop

endrecordNok:
debug "Error On End Of Recording",CR
stop

recordNok:
debug "Recording Refused",CR
stop

formatNok:
debug "Formatting failed",CR
stop

query_msg_Nok:
debug "Query Msg Num Failed",CR
stop

query_id_Nok:
debug "Query Id Failed",CR
stop

query_sampling_Nok:
debug "Query Sampling Rate Failed",CR
stop

query_volume_Nok:
debug "Query Volume Failed",CR
stop

samplingNok :
debug "Sampling Rate Refused",CR
stop

volumeNok
debug "Volume Not Set",CR
stop

EraseNok
debug "Message cannot be erased",CR
stop


'==================================================
'Command Subroutines
'==================================================

query_msg_num:

debug "Query Msg Num ",CR
serout
TRANSMIT_LINE,NBAUD9600,[QV_QUERY_MSG_NUM,0]
serin
RECEIVE_LINE,NBAUD9600,1000,query_msg_Nok,[ser-
Data]
debug "Next Msg: ",HEX2 serData,CR,CR
return

'==================================================
query_id:

debug "Query ID ",CR
serout TRANSMIT_LINE,NBAUD9600,[QV_QUERY_ID,0]
serin
RECEIVE_LINE,NBAUD9600,1000,query_id_Nok,[ser-
Data,serData1]
debug "Module Series: ", DEC2 serData ,"     Ver-
sion: ", DEC2 serData1, CR,CR
return

'==================================================
query_sampling:

debug "Query Sampling Rate ",CR
serout TRANSMIT_LINE,NBAUD9600,[QV_QUERY_SAM-
PLING,0]
serin  RECEIVE_LINE,NBAUD9600,1000,query_sam-
pling_Nok,[serData]
debug  "Sampling Rate: ",DEC2 serData," Khz",CR,CR
Return

'==================================================
query_volume:

debug "Query Volume ",CR
serout TRANSMIT_LINE,NBAUD9600,[QV_QUERY_VOLUME,0]
serin
RECEIVE_LINE,NBAUD9600,1000,query_volume_Nok,[ser-
Data]
debug "Volume: ",HEX2 serData,CR,CR
return

'==================================================
do_reset:
```

```
OUT4 = 0
pause 1000
out4 = 1
debug "Device Reset",CR,CR
return

'==================================================
do_format:

debug "Formatting Device.....",CR
serout TRANSMIT_LINE,NBAUD9600,[QV_FRMTDEV,0]
serin
RECEIVE_LINE,NBAUD9600,16000,formatNok,[serData]
if serData<>QV_OK then formatNok
debug "Formatting OK....",CR,CR
return
'==================================================

do_set_sampling:

debug "Altering Sampling Rate: (0)=8.0KHZ
,(1)=6.4KHz, (2)=5.3Khz ,(3)=4.0Khz ", CR
serout TRANSMIT_LINE,NBAUD9600,[QV_SETSMPFR,qvSam-
pling]
serin RECEIVE_LINE,NBAUD9600,1000,samplingNok
,[serData]
if serData <> QV_OK then samplingNok
debug "Sampling Rate  = ",HEX2 qvSampling,CR,CR
return

'==================================================

do_record:

serout  TRANSMIT_LINE,NBAUD9600,[QV_RECFRAUX,Mess-
Num] 'Record from line-in
serin
RECEIVE_LINE,NBAUD9600,1000,recordNok,[serData]
if serData <> QV_OK then recordNok
debug "Start Recording For 10 Seconds",CR
pause 10000 'wait 10 seconds
serout TRANSMIT_LINE,NBAUD9600,[QV_STOP]     'end
of record command
serin
RECEIVE_LINE,NBAUD9600,1000,endrecordNok,[serData]
if serData <> QV_OK then endrecordNok
debug "End Of Recording -OK-",CR,CR
return

'==================================================
do_play:

serout
TRANSMIT_LINE,NBAUD9600,[QV_PLAY2SPEAKER,MessNum]
'Play to speaker
serin  RECEIVE_LINE,NBAUD9600,1000,playNok,[ser-
Data]
if serData <> QV_OK then playNok
debug "Playing -OK-  ",CR
serin  RECEIVE_LINE,NBAUD9600,[serData]
debug "Playback Finished",CR,CR
return

'==================================================
do_set_volume:

debug "Set Volume (0 to 7) ","  0=Max, 7=Min", CR
serout TRANSMIT_LINE,NBAUD9600,[QV_SETVOL,qvVol-
ume]
serin RECEIVE_LINE,NBAUD9600,1000,volumeNok,[ser-
Data]
if serData <> QV_OK then volumeNok
debug "Volume = " ,HEX2 qvVolume,CR,CR
return


'==================================================

do_erase_mess

debug"Attempting to erase message",CR
serout TRANSMIT_LINE,NBAUD9600,[QV_ERASEMSG,Mess-
Num]
serin RECEIVE_LINE,NBAUD9600,1000,EraseNok,[ser-
Data]
if serData <> QV_OK then EraseNok
debug "Erased Message No:= " ,HEX2 MessNum,"  -OK-
",CR,CR

'==================================================
'==================================================
```

sent to the module. Sound will now be played back to the ANA_OUT pin, the auxiliary audio output.

With the connections as in the schematic diagram, the test-program can be downloaded into the Basic Stamp 2 and the module will record audio as directed by the user.

The test program

1) Resets the module
2) Formats the module
3) Queries the module for the next available message number
4) Queries the module Identification
5) Sets the playback volume
6) Queries the playback volume
7) Sets the Sampling Rate
8) Queries the Sampling Rate
9) Records a message with the No: 1, for 10 seconds
10) Plays back the message No: 1
11) Erases Message No: 1

The test program is structured so that the user can easily experiment by writing their own subroutines or by simply commenting out various segments. The test program only provides a simple starting point, multiple messages can be recorded, played successively, and deleted by calling the subroutines as required. Additional features of the module, not implemented in the test program, are described in the full user manual can be created by using these routines as a framework.

The test program is extensively commented to enable you to add your own extensions and features. The program is available from the Free Downloads section of the *Elektor Electronics* website at www.elektor-electronics.co.uk.

(024129-1)

Product availability

The QV381m0 Record/Playback Module is available from Quadravox Inc., 1701 N. Greenville Ave., Suite 608, Richardson, TX 75081, USA.
Tel. 1-800-779-1909 or 1-972-669-4002.
Fax: 1-972-437-6382.
Website: www.quadravox.com
Email: info@quadravox.com
James Evans Electronics can be contacted at Email: jamesevansjee@hotmail.com