BASIC (PART 3)

μP-controlled frequency meter
fourth in our series of measuring instruments

Portable guitar amplifier
two articles for guitar lovers

Cassette interface for VIC 20/64
overcomes one of Commodore's shortcomings

- **CMOS function generator** • **Simple transistor tester** • **Crystal time-base**

1.2 GHz
μP
COUNTER

POWER
ON
OFF

HOLD
RESET

MENU   YES   NO   LAST

TRIGGER

GATE

SENSITIVITY
MIN   MAX

HF

DIGITAL

LF

C

B

A

elektor

contents

tea leaves...

Looking in our tea cups, we did not feel like starting the new year with reflections on super-chips or solid-state audio, but decided to wait and see how last year's tea-leaf patterns in this respect will fare. So, what would we like to muse over? Well, the first point is the state of technological development. This is still accelerating at an increasing rate: the super-chip of today is the standard chip of tomorrow. We regularly receive letters with cries for help like: "Help! I stopped my subscription two years ago, but because of withdrawal symptoms I have recently recommenced it. But I don't understand the circuits any more!..."

If we give free rein to our gaze into the future, we must assume that intelligent machines, far-reaching miniaturization, and affordable prices will become the norm. The second point is: where will we find ourselves in 1985? One thing is certain: new devices and equipment so number and amaze us. The

housewife will have a microprocessor-controlled microwave oven which automatically weighs the food and sets the appropriate temperature. All she has to do is to indicate that she is putting in a chicken and the oven does the rest. Or does it? The first time the oven is being used, you hear cries from the kitchen "Why is that stupid thing not working? It is program three...chicken.... well then, why don't you start? Oh, the door was not closed properly. Sorry!" Click.

So, we are no longer overwhelmed with wonder, but we do remain critical. If a 'stupid' thing does not do what we expect it to do, it's no fun any longer.

Utensils and crazes

When something is new and not too expensive, it sells. With some luck (and a lot of publicity) it may become a craze. Or a fashion. Or a normal utensil for everyday use.

What's the difference? Why were newton's marbles, the hula hoop, and the rubik cube crazes? Why were the charleston, the mini skirt, and aluminium luggage racks in fashion? Why are telephones, automatic washing machines, and zips utensils? The difference appears to lie in the user-friendliness. Utensils are just, well, handy. They make it possible to do something you wanted to do in an easy manner. They have a purpose, they meet a need. Crazes and fashions are, initially, exciting, but after a while the novelty wears off — and all of a sudden they are no longer useful, or user-friendly; just plain dull.

Which brings us to the home computer. Craze or utensil? As they are now, we would say: craze. They are affordable, exciting — even addictive. If you manage to work with them you continue doing so. Sometimes until two or three in the morning. But why? And here's the crux of the matter. Because those

'stupid things' do not understand — and therefore don't do — what they want. They are not user-friendly. And that's why most computer owners get fed up with their machines. The standard games are boring and the 'family memory' was far more accessible when it was contained in the telephone book. Home programming is not so hot either. So?

In nine out of ten cases, the computer is put away in a cupboard. Nice when you have visitors, but for the rest... In the remaining one case, some interest remains. Elektor readers are prominent here. They want more. For them the computer is a hobby, not a utensil. For that, it is not yet sufficiently user-friendly.

The future

Back to the tea leaves. What can we expect? Which crazes will come and go? What utensils will become 'normal'?

As we said, the cheap home computer appears to have all the hallmarks of a craze. The new MSX sets promise to become one hell of a craze: they are almost 'real', there are promises of much software, and they are not too expensive. But...user-friendly? No, not really. Do they meet a real need? No, not really.

What about that pocket-diary-in-a-wristwatch recently launched by Seiko? There is some doubt here. An 'old-fashioned' diary is cheap and convenient, and you can enter or alter information in it without first having to read the operating instructions. And yet, a watch is always with you, while a diary may accidentally be left at home. What are our real needs; in what

direction should we look for future utensils? Let's see:

■ **Peace and quiet.** A noise eradicator would be very desirable. Built into your easy chair, or at the head-end of your bed. Technically, it is quite feasible, but as yet very expensive.

■ **Communication,** but selective. This is a wide field, but to name one aspect that is definitely technically-possible: a new British Telecom telephone service. John indicates on his phone that he would like to speak to his brother. All subscribers indicate when they don't want to be disturbed; for instance, by a switch on their phone. On the first occasion both John and his brother are free and available, the computer makes the connection.

■ **Information** Assuming that you're interested in international news, chess, cartoons, and technical news, which paper do you read? Why, your own, personalized one, of course — ordered from a fully automated publishing company. Each morning your very own paper rolls from your own printer — no search results, no stock exchange news, no gossip.

■ **Literature:** books and magazines, for learning. Currently, these have many advantages: they are easy to use, legible, and relatively cheap. A computer screen is definitely not an improvement. And yet, the electronic system has its good points: the information is distributed easily and cheaply and the 'covers' may be very compact — mini cassette or mini floppy. So, what do we need? A handy, not too expensive, battery-operated reading book with a clearly legible screen. Put in the cassette and on you go with your

novel. When you've finished it, you call for a new 'book' — by telephone, of course! Now, that will be user-friendly!

■ **Physical aids,** for healthy as well as for (lightly) handicapped people. Better hearing aids, for example, but also night glasses (with light amplifier), which are much more convenient and economical than torches. Or a muscle-enervating massage belt to enable you to lose those pounds during breakfast — now, that could become a craze (and, like its current mechanical counterpart, it would be quite useless!).

■ **Elektor** What sort of articles will we publish in *Elektor* in the 1990s? Using the same criteria as today — practice-oriented, using modern technology, broadly dealing with all aspects of electronics — the contents are bound to be quite different. Or are they?

We will publish audio preamplifiers, but with a 32-way input bus for those new audio ROMs.

And we will combine those new Mbyte memory ICs with those ridiculously cheap A/D and D/A converters to make a guaranteed-reproducible echo unit.

We will make a de luxe home telephone exchange for connection to the normal telephone network; this will be allowed by then.

We will make a 'music searcher' which tunes the radio (yes, that will still exist) to a transmitter that broadcasts the music of our liking and no advertising!

We will ... well, we'll see ... In any case, we'll make it interesting! ◾

selektor

computers of the future

It is not all that long ago that computers worked only from a bunch of punched cards which could not be produced at home. Nowadays, computers are generally operated from a keyboard but this is still a long way from the desired goal of voice-actuated machines. Keyboards are highly restrictive because they mean that if a user wants to communicate with a computer he or she has to learn to write programs. As the main uses of computers will become cen-

tred more and more on office automation, data communications, and home work stations, it is extremely unlikely that more than a small portion of users will be prepared to learn how to program — and why should they?

No, computers will have to come into much closer contact with the user to become a real tool of human thought. This means that computers have to become much cleverer than they are.

In essence, a computer is nothing

but an array of interfacing units: at the centre the microprocessor, swathed in software — machine code, compilers, assemblers, operating systems, languages, application programs, printers and/or screens. This is, of course, an unfortunate arrangement because any change at the user's end will normally affect all these peripherals, and therefore the very design of the computer.

Of course, even users of personal computers now have other devices at

their disposal for communicating with the computer. For example, the mouse is a small pointing device which when rolled along a desktop causes a pointer to move across the screen in a corresponding direction. Originally the mouse was operated through a pair of wires, but modern versions are cordless and communicate with the computer by infrared signals. The mouse is also used to select icons, which are small graphic representations displayed on the screen to illustrate a particular function.

These are but first steps towards personalisation of the computer, that is, giving the user facilities for moulding the operations more to his own requirements. The next steps will probably have to wait until the 32-bit microprocessors (such as Motorole's 68020S, Intel's 80386, Zilog's Z8000, or National Semiconductor's NS32032) and very large, inexpensive memories have become available in the course of the next few years. It is true, of course, that there are 32-bit computers on the market already, such as Apple's Lisa and Macintosh, both of which use Motorola's 68000 microprocessor. Unfortunately, the 68000 has only a 16-bit bus, so to the user these computers do not seem much faster than IBM's 16-bit PC. True 32-bit chips with a full-size bus will enable designers of PCs to give users much more flexibility: they can execute between two and three million instructions per second, that is as fast as many main-frame computers, and can process software programs more than three times as fast as the 68000.

Power requirements of these new chips are the same as that of the 68000 although they contain three times as many transistors. This low power consumption, which minimizes heat problems, and the high speed coupled with a large memory mean that very small but very powerful computers can be built. As a result, highly complex mathematical and engineering problems can be executed by a desktop computer and up to twenty people can work on the same computer file simultaneously.

Personalized computers have recently become available in the USA from Metaphor. These machines are customized for groups of users, such as accountants. Metaphor expects users, untrained in programming, to take it from there and write further applications for themselves. So, here is already an example of what can nowadays be done by the user

himself that had to be done by specialized programmers only a few years ago.

The two important developments are, however, the use of natural instead of man-made language (such as BASIC or FORTH) and graphics. The ability to speak into a computer instead of having to sit down at a keyboard will increase the uses of computers vastly. Basically, what is required here is an interface between the user and the computer: a device such as a voice-actuated typewriter (vet) that can transcribe speech without having to understand it.

As the problems facing designers of automatic speech recognition (ASR) systems are enormous, large programmes, many government sponsored, have been initiated in Britain (the Alvey programme, named after John Alvey who chaired the committee that established the programme in 1982); in Europe (the European Strategic Programme on Research in Information Technology — ESPRIT); in the USA (Defence Advanced Research Projects Agency — DARPA; — Microelectronics and Computer Technology Corporation — MCC — the industrial counterpart of DARPA; and Semiconductor Research Cooperative — SRC); and in Japan (Institute for new-generation Computer Technology — ICOT — and the National Superspeed Computer Project).

Several of the world's big computer companies have been working on automatic speech recognition (ASR) for some time. A few months ago, IBM announced that it had produced an experimental system with a ninety-five per cent recognition rate for a 5000-word business vocabulary. There are two approaches to ASR, of which one relies on a template recognition system. Templates are digital patterns and by a recognition algorithm to identify words. The other is based on pattern analysis of the forty phonemes that make up spoken English, and the 1600 transitions possible between each adjoining pair of them.

All the signs are that by the year 2001 we will be able to emulate HAL (but only as far as its positive aspects!) and command computers through natural language, because that is, and will remain, the most efficient and simple, though complicated, instrument for human communication.

Visual representation is virtually useless to convey certain kinds of information. But when it comes to show the relation between, say,

forecast and actual sales, pictures come into their own.

Unfortunately, many companies have been frightened by the experience of Mindset. This Silicon Valley company makes an IBM compatible PC that produces high-quality graphics. After this set had been hailed by the press at its launch early last year, it flopped. Although there were market reasons for this, the main disappointment was that the idea that good graphics would create its own market among general business users was wrong. None the less, most consultants are forecasting that the computer graphics industry is heading for a huge boom, with a growth rate of more than forty per cent per year for the remainder of the 1980s.

Good graphics will be possible because PCs are becoming powerful enough to use bit mapping, a technique in which each pixel can be controlled individually. The most impressive PC with 8 bit-mapped high-resolution screen is Apple's Macintosh, while in the business market it is IBM's 3270PC/G (which, incidentally, also has a mouse). Finally, two new printing technologies are becoming economically available. Up till now, there have been three basic types of printer: daisywheel, producing good quality print slowly and loudly; dot matrix, which uses a row of fine needles shot forward by electrical pulses to form the shape of letters end figures and is faster but less clear than the daisywheel; and heat transfer, which also uses needles but, instead of moving, these heat up momentarily.

But the future will almost certainly belong to laser and LCD (liquid crystal display) printers, both of which are fast (6. . .7 pages of A4 per minute), noiseless, and give a print quality where you do not notice any dots (pixels). Printers using these techniques are still very expensive, although market forecasts are that there will be a laser printer on the market for under £1000 in about a year's time. ◼
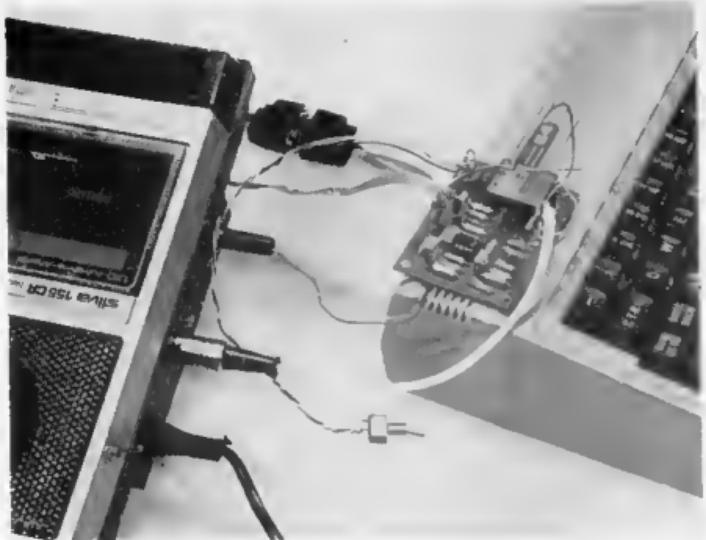
Commodore cassette interface

Necessity being the mother of invention this circuit was destined to be designed. The 'necessity' here is Commodore's insistence that the VIC 20 and C64 computers can only be used with a special cassette recorder supplied by ... Commodore. One of our designers beulked at the idea of buying e cassette recorder only for his computer when he eIready had e perfectly good eudio recorder. Some kindly soul directed his outraged energy towerds the neerest soldering iron and so wes born THE Commodore cassette interface by Elektor.

As in many home computers, the data output by the VIC 20 and C64 via the cassette interface is in the form of a square wave signal with an amplitude of 5 Vpp. The information input to the computer has the same form. The cassette connector also contains a so-called 'sense input' that enables the computer to check if the recorder's PLAY button is pressed. The Commodore will only activate its motor output if this is the case. The computer itself switches the recorder motor on and off and we will see later how this is actually done. First, however, we must see what happens if the computer wants to save a file on cassette.

THE interface in detail

The program to be saved appears in the form of pulses with an amplitude of 5 Vpp on the write output of the connector. This amplitude is, of course, much too large to be saved directly on the tape so the signal is first reduced to about 300 mV by voltage divider R13/R14. This signal is then suitable for recording on the tape. The procedure for loading a program is somewhat more involved. The signal supplied by the recorder via a DIN or loudspeaker output socket is nowhere near square in shape and its amplitude is also much too small, at about 200 to 300 mV. The signal must therefore be amplified and its shape improved. A single op-amp (A1) increases the signal's amplitude by about six times. The offset of A1, and consequently of A2, is set to half the supply voltage by means of R3 and R4. The second op-amp is set up as a schmitt trigger which takes the signal from A1 and forms it into a clean square wave signal with an amplitude of 5 Vpp. The computer can now load the program via the connec-
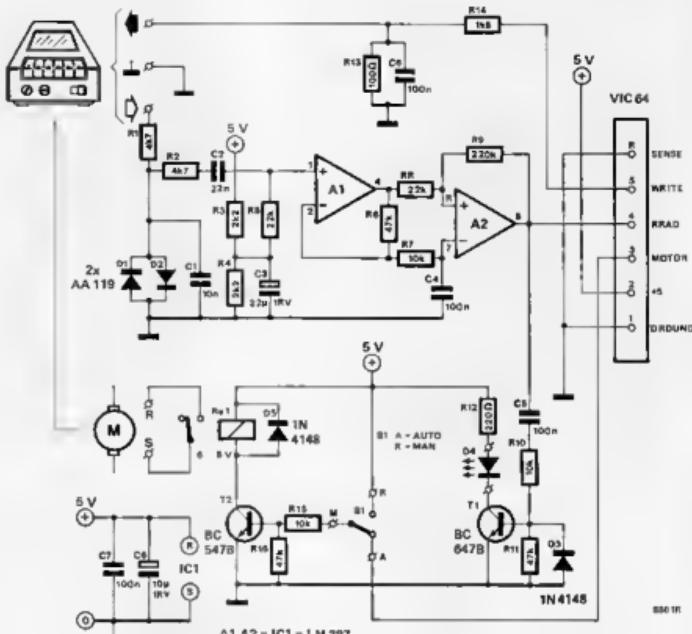
Figure 1. Looking at how simple this cassette interface is, you may wonder why Commodore found it necessary to insist on a special recorder. We did!
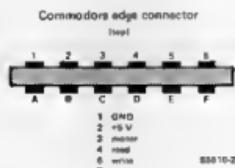
Figure 2. This is the layout used by the six-way edge connector on both C64 and VIC 20 computers.

tor's read input. A LED (D4) has been included in the circuit to show that this is actually happening. It will only light when transistor T1 is caused to conduct by a '1' appearing on its base but the logic levels switch far too fast to be seen with the naked eye. When data is being transferred the LED appears to be continually on. One obvious advantage of this LED is that it simplifies finding the right section of a program. The motor in the cassette recorder must be switched on and off at the right times by the computer. Naturally enough this is done via a relay (Re1) instead of directly. When pin 3 of the connector goes high transistor T2 is switched on and causes

the relay to operate. First of all, however, the computer must be made to think that the PLAY button is pressed. This condition is simulated by connecting the sense input to ground, which is exactly what happens in the Commodore data recorder when this button is pressed. If the sense input is connected straight to ground, as we have done, we can then simply forget about it. The power supply for the interface is very kindly provided by the computer. As figures 1 and 2 show, pins 1 and 2 of the cassette connector are GND and +5 V respectively. This makes a separate supply for the circuit unnecessary.

The interconnections

Before any data can be transferred all the electronics must be constructed and all interconnections must be made. A total of six links are needed at the computer side and the most important point here is to ensure that none of these are interchanged. The computer would not be very pleased with this so have a look at figures 1 and 2 to see where everything belongs before soldering any wires. The cassette recorder is more tolerant of wrong connections but it is far better if

the circuit works correctly first time. Again make sure to solder the right wires in the appropriate places. The layout usually used for a cassette recorder's DIN socket is illustrated in figure 3. The two terminals for the relay contacts (points R and S in figure 1) are linked to the recorder's remote input via a jack plug. If the tape recorder in question does not have any remote input this is no reason for panic; simply connect R and S in series in one of the voltage supply lines to the motor.

Installation and use

All the components must be fitted to the printed circuit board shown in figure 4 and when this is done a suitable case must be found for the circuit. Alternatively, it may be possible to include it within the cassette recorder case. Whichever of these is chosen there is one point to bear in mind: the connecting wires must not be made too long. A special connector is needed to link the interface to the computer's cassette input/output lines. This is a six-way printed circuit board connector with a spacing of 3.96 mm (0.156") between the pins. The wires can also be soldered directly onto the printed circuit board. We will be very brief in our instructions

on using the circuit; refer to page 18 of the C64 user's manual

The function of S1 in the circuit is clear. It is used to switch the motor on and off, which is very handy for winding the tape. If an error message is generated during loading the volume control on the cassette recorder is probably set incorrectly and when reading tapes that you have not recorded yourself it may be necessary to re-align the record/playback head. If our experiences with this interface are anything to go by, however, errors will rarely occur, even when 'turbo' loading.

DIN socket

playback   ground   record

85070-3

Figure 3. Here is the arrangement most commonly used when a cassette recorder has a DIN socket for input/output.
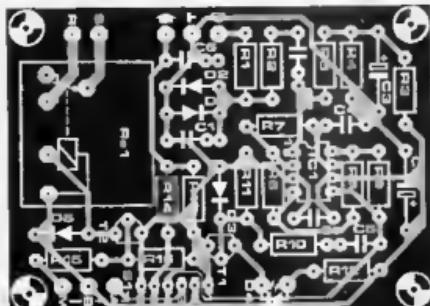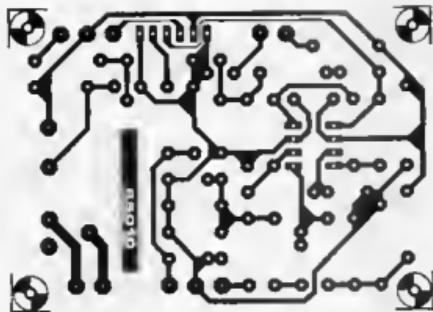
Parts list

Resistors:
R1, R2 = 4k7
R3, R4 = 2k2
R5, R8 = 22 k
R6, R11, R16 = 47 k
R7, R10, R15 = 10 k
R9 = 220 k
R12 = 220 Ω
R13 = 100 Ω
R14 = 1k5

Capacitors:
C1 = 10 n
C2 = 22 n
C3 = 22 µ/16 V
C4   C7 = 100 n
C8 = 10 µ/16 V

Semiconductors:
D1, D2 = AA119
D3, D5 = 1N4148
D4 = LED
T1, T2 = BC547B
IC1 = LM387

Miscellaneous:
Re1 = relay, 6 V PCB-mounting type (MS Components — part no. 9055)
S1 = single-pole toggle switch
6-way PCB edge connector with pin spacing of 3.96 mm (0.156") — Maplin, order no. FG24B)

Figure 4. Constructing the circuit is simply a matter of assembling the components on the printed circuit board. Make sure no wires are switched when making the interconnections.

Computers, video games, video cameras, games computers; all of these produce video signals that must be displayed via a television set. If the TV receiver in question does not have a video input and its owner is reluctant to vandalise it in order to fit one then this sort of modulator is the obvious solution. It is a simple circuit that processes video signals to enable them to be fed straight into the TV set's aerial input.

VHF/UHF TV modulator

for any TV set without a video input

A 'TV modulator' is really no more than a transmitter. It is a very simple transmitter, admittedly, but none the less that is what it is. What does a modulator actually do? In general — and this design is no exception to the rule — it is a simple oscillator that generates a frequency somewhere in the VHF or UHF region. The oscillator is modulated with the video signal and the modulated carrier wave thus generated is fed into the TV set's aerial input via a cable. Then all that remains to do is tune the TV to the correct frequency.

The layout

The whole business is not quite as simple as we have just suggested, of course, as the mini transmitter must meet certain requirements. The frequency stability must be very good as, indeed, must be the quality of the display. The required frequency stability is achieved by the use of a crystal oscillator. A well thought out
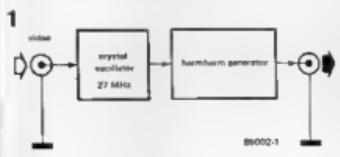
choice of component values takes care of the display quality: the modulator allows a resolution of 80 characters per line, as this is a value that is often needed.

A very important feature of the circuit that must be decided is the transmission frequency. If this is only a single channel, as suggested above, it gives rise to some practical problems. Different users will want different channels, the carrier wave can become somewhat difficult to locate, and unless the frequency is exactly spot on no signal will be received. A much better idea is to ensure that the HF signal contains a large number of different frequencies. This makes it much easier to tune the TV set to one of the channels as there will surely be one to suit every user.

The block diagram of figure 1 shows how this is achieved. The TV modulator is made up of two parts, namely a modulatable crystal oscillator and a harmonics generator. The oscillator operates at a frequency of 27 MHz, which is quite low as inexpensive crystals are readily available. The harmonics generator converts the oscillator signal into a sort of frequency spectrum containing all the multiples of 27 MHz to about 1800 MHz. The TV modulator's output signal is made up of a large number of little peaks, each of which is a complete transmitter signal. At least one of these will always be in band I (VHF channels 2 . . . 4), one in band III (VHF channels 5 . . . 12) and many of them will be in bands IV and V (UHF channels 21 . . . 69).

The circuit diagram

Like the block diagram, the circuit (shown in figure 2) is very straightforward. The crystal oscillator is based on a very fast HF transistor, T1 (BFR91), which performs the amplitude modulation. Apart from this there is little to be said about the oscillator except, perhaps, that it is essential to use the correct values for the components surrounding T1. This is, of course, simply common sense in this sort of HF circuit. The harmonics generator is formed by two Schottky diodes, D1 and D2. These diodes must switch very quickly in time with the 27 MHz signal so they provide strong harmonics up into the gigahertz range. The modulation depth can be set with P1, while the oscillator's d.c. value can be varied by means of P2. The combination of these two presets enables either positive or negative amplitude modulation to be selected. This is essential as the harmonics produced vary in this respect. We will discuss the calibration of P1 and P2 later in this article.

The power for the circuit can be provided by either an unstabilized 8...30 V or a stabilized 5 V. The latter could be taken from a computer's power supply and in this case IC1 is not needed.

Construction

The tiny printed circuit board designed for this circuit is shown in figure 3. It is

Figure 1. A TV modulator is, in fact, a small TV transmitter. In this case the transmitter consists of a modulatable (AM) oscillator followed by a harmonics generator.

1

video

crystal oscillator 27 MHz → harmonics generator

86002-1

Figure 2. In this circuit diagram we see that the 27 MHz oscillator is based on transistor T1 and the harmonics are generated by Schottky diodes D1 and D2

Parts list

Resistors:
R1, R2 = 4k7
R3, R4 = 56 Ω
P1 = 100 Ω preset
P2 = 500 Ω preset

Capacitors:
C1 = 4µ7/16 V
C2 = 10 p
C3 = 220 p
C4 = 47 p
C5 = 47 n, ceramic
C6 = 100 n*
C7 = 330 n*

Inductors:
L1, L2 = 3½ turns of
0.2 mm (SWG 35 or 36)
CuL on a ferrite bead of
about 3.5 × 3.5 mm
L3 = 1 µH
L4 = 1 turn of 0.8...1 mm
(SWG 19...21) CuL, air-
wound with a diameter of
8 mm

Semiconductors:
D1, D2 = 1N6263
(Ambit/Cirkit)
D3 = 1N4148
T1 = BFR91 (Ambit/Cirkit)
IC1 = 7805*

Miscellaneous*
X1 = crystal, 27 MHz (3rd
overtone) or other 3rd
overtone crystal between
25 and 30 MHz

* = not needed if the
circuit is powered from a
stabilised 5 V supply

not double-sided as this was found to be
unnecessary. Construction is therefore
simplified and readers who do not want to
etch the board through our EPS service (tut-tut)
will find it easier to make themselves.
Building the circuit is simply a matter of
fitting the components onto the printed
circuit board. The coils, often a source of
much teeth-gnashing and hair-pulling, will
not be a problem in this case. Two of
them, L1 and L2, are made by winding 3½
turns of enamelled copper wire (about
0.2 mm thick) on a 3.5 mm ferrite bead.
Another, L4, is just one turn of copper
wire (0.8...1 mm thick) air-wound with a
diameter of 8 mm. The fourth inductor, L3,
can simply be bought.
Any third overtone crystal with a fre-
quency of between 25 and 30 MHz will
work in this circuit. A number of suitable
values are advertised in this issue.
The only parts that might prove difficult to
find are diodes D1 and D2. The ones
stated in the parts list are available at the
moment but do not give up hope if your
corner shop does not have them. The only
important thing is that they must be UHF
Schottky diodes; the actual type number
is of little consequence.

Calibration

Calibrating the modulator calls for a cer-
tain degree of care as it involves more
than just 'set the presets to mid position'.
The setting depends, in fact, on the har-
monic to which the circuit is tuned. Cali-
bration should be carried out as follows:
■ Set the TV receiver to maximum
brightness and contrast.
■ Feed a video signal into the modulator
(a video recording of a test card, or a
link to a computer's 'TV' socket, could
be used) and connect the circuit's output
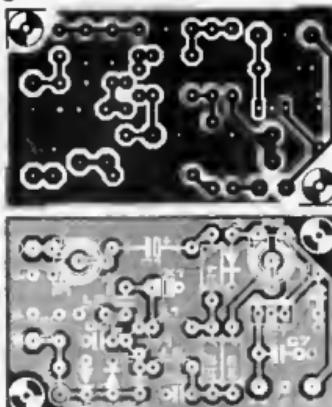
to the TV's aerial input.
■ Set P2 to mid-position and P1 to
minimum resistance (fully anti-
clockwise).
■ Tune the TV receiver to a harmonic,
preferably one of the VHF bands (chan-
nels 2...12). The tuning is correct when
the 'snow' on the screen disappears
and/or the screen becomes dark.
■ Turn P1 very slightly until 'something'
becomes visible.
■ Calibrate P2 to give the best possible
quality image. If the result is not very
good the wiper of P1 can be moved a bit
more and P2 again trimmed to give a bet-
ter image.
■ If this still fails to provide an acceptable
result tune the TV to the next harmonic.
This must give a decent image.   ∎

Figure 3. Fortunately the
printed circuit board for
the modulator is only
single-sided. The large
copper surface acts as a
ground plane.

portable guitar amplifier

with a host of facilities

Apart from output power and the choice of valves or transistors as active elements, there is not much to distinguish the multitude of commercial guitar amplifiers on the market. All that most of them offer is a three-channel aqualizer for bass, mid-range, and high frequencies, and a built-in reverberation spring: this does not give the musician much scope for experimentation. Yet, even for only a modest outlay is it possible to bring some colour to the monotony of the basic sound of the strings. This is made possible by a voltage-controlled filter — surely no stranger to our musical readers.

The controls for bass, mid-range, and high frequencies on most commercial guitar amplifiers may be compared with those on a hi-fi installation. Unfortunately, they do not change the basic character of the sounds produced by a guitar string: to do so, many more additional units are required: phaser, chorus, flanger, fuzz box, and so on. In many cases, these units really do mellow the harsh notes produced by the guitar. The amplifier described here does not, and is not intended to, replace such additional units entirely. However, several of the add-on units are based on similar principles and can be imitated fairly easily, and well, by the voltage-controlled filter on which the design of the present amplifier is based. Mixing the outputs of a voltage-controlled filter (VCF) allows the continuous transition from all pass to notch mode. Furthermore, the signal from the integral fuzz circuit may be added to the outputs of the VCF, so that with only four potentiometers a whole spectrum of tone colour variations becomes available.

Voltage-controlled filter (VCF)

The VCF is a simplified version of that used in the Formant synthesizer (see *Elektor (UK)* December 1977, page 12-27). It is built from opamps A2, A3 and A4 as shown in figure 1, which function as high pass, band pass, and low pass filter respectively. Stereo potentiometer P3 enables the setting of a specific turnover point, that is, the centre frequency of the band pass filter.

Overdrive (fuzz) circuit

Either the direct or the filtered signal from the guitar may be fed to opamp A5 by switch S1. The gain of the amplifier can be set within wide limits by P2: this is vital because the stage following A5 needs a threshold to function as limiter and so provide the required amount of distortion. The use of three pairs of diodes results in a smoother onset of limiting, that is, a progressive increase in distortion, which provides a tone reminiscent of valve amplifiers. The operation of the overdrive circuit is clarified in figure 2.

Mixing

Potentiometers P6...P9 allow the mixing of the outputs of the filter and the distortion generator: the wipers of all four are connected to the inverting input of opamp A7. Preset P10 enables the setting of the required feedback factor: the higher its value, the greater the gain of A7.

Reverb(eration) unit

Reverberation springs have been a welcome addition to guitar amplifiers for a long time. Although they do not perform miracles, they add to the fullness of the sound.
The principle on which they work is that the sounds to be reverberated are magnetically coupled to one or two metal

1

A8 A4 = IC1 - TL 074
A5 ... A8 = IC2 - TL 076

Figure 1. Circuit diagram of the pre-amplifier.

springs. Because of the elasticity of the springs, the mechanical waves to which the signals from the guitar have been converted take a certain time to reach the other side of the spring where they are recovered to electrical signals. Integrated a.f. amplifier IC3 drives the input coil: a standard opamp would not be able to provide sufficient energy, that is, output current.

Opamp A8 is a mixer in which the direct and the reverberated signals are added together. Potentiometer P12 enables continuous adjustment of the level of the echo. The relatively low value of R17, compared with that of feedback resistor R19, provides a fiftyfold gain of the reverberated signal.

Calibration

The potentiometers are all brought out to the front panel and their setting will be discussed under 'operation and setting up'.

Depending on the type of pick-up used, the peak value of the signals produced by an electric guitar vary widely. Preset P1 determines the gain of input amplifier A1. To start with, set the wiper of this control to the centre of its travel. Then connect the output of A1 (pin 7) to the output amplifier, and connect an electric guitar to the input of the circuit. If the 30-watt a.f. output stage described elsewhere in this issue is used, set the potentiometer, P13, between A1 and that amplifier to maximum. Also, set the volume control on the guitar to maximum.

The maximum required volume can now be set with P10, but only after P6...P8 have been turned open (wipers at output of opamps A4, A3, and A2 respectively), P3 and P4 have been set to the centre of their travel, and P9 and P12 so that their wipers are at earth potential. The setting of P2 is immaterial. When all this is done, the basic sound of the guitar, without distortion and reverberation, should be heard in the loudspeaker. If the sound is distorted, it probably means that A1 is overloaded. The remedy is to reduce its gain with P1. If you have an oscilloscope, check that the signals at the output (pin 7) of A1 are not being limited.

It may also be that the output stage is overloaded; in that case, reduce its gain with P13 until the sound is fairly soft.

If there is still distortion, A7 or A8 may be overloaded, the remedy for this is to

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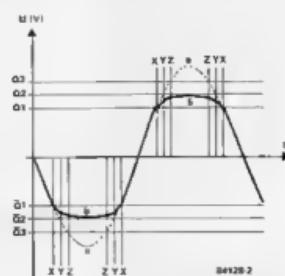
Figure 2. A clean sine wave (a) at the input of the overdrive (fuzz) circuit appears flattened (b) at the output if its peak value is equal to, or exceeds, the thresholds. Q1...Q3 of the three pairs of limiting diodes. At x the first diode pair starts to conduct, at y, the second also, and at z all diodes conduct.

reduce the gain of A7 with P10. Again, if you have an oscilloscope, the output of A7 (pin 8) or of A8 (pin 14) may be checked for clipping. It is, however, very unlikely that there is overloading of these stages.

Setting the reverberation

The level of the signal applied to the reverberation exciter coil is determined by preset P11. To find the correct level, first set P12 to maximum. Next, move the wiper of P11 slowly from earth potential and simultaneously pluck one of the guitar strings. If everything is all right, the echo

Figure 3. If a composite signal is distorted through limiting, the higher frequencies may be chopped (b, 2) or not be affected at all (b, 1). A more pleasant sound ensues if first only the high frequencies are limited and then superimposed onto the lower ones (c).

Figure 4. Suggested construction for the housing for the complete amplifier. Note the chassis to which the front panel is fixed; the whole may be slid into and out of the case

should become clearer and clearer. However, at a certain point, that is, when the output level of IC3 becomes too high, the echo becomes muffled and suffers from frequency-dependent distortion. To avoid this happening even when the guitar is plucked vigorously, carry out this test with firm plucking of the string. If the echo is too weak with P11 and P12 at maximum, it may be enhanced with P10, even though this also increases the level of the direct signal.

Operation and setting up

Playing without overdrive (fuzz): set P9 to zero (wiper at earth) and with P6...P8 choose the desired bass, mid-range, and treble response. It is important that the setting of each of these controls is compatible with that of the others. When all three are set for identical gain, the output of the guitar sounds virtually natural. Potentiometer P4 is of considerable importance to the operation of the VCF: it determines the quality, Q, that is, the slope of the pass band of the filter. When the Q is high (steep slope) it is possible to produce artificial resonance peaks in the band-pass characteristic which give the sound a distinct colouring. This is also affected to some extent by the setting of P3.

Varying P3 (imagine this control fitted in a foot-operated swell) with Q high and only the low-pass filter section operating causes a wa-wa effect. With a low Q and the mid-range frequencies attenuated, slowly altering the crossover frequency gives rise to a phasing effect.

When the wiper of P4 is at earth potential, the VCF functions as an oscillator and it is therefore necessary that P5 is adjusted so that with maximum Q the filter just does not oscillate.

Playing with overdrive (fuzz): when switch S1 is in position 1, the entire sound from the guitar becomes overdriven. Unlike many other guitar amplifiers, the present one allows the continuous mixing of the original and overdriven sounds: this may be arranged by a foot switch connected to jack socket S (see figure 1). If only the overdriven sound is wanted, potentiometers P6...P8 must be turned off completely (wipers at earth).

The degree of overdrive may be set with P2 to individual taste.

When S1 is in position 2, only those frequencies that lie above the crossover point set by P3 will be overdriven. If you mix the fuzzy high frequencies with the original low ones, a very pleasant, hoarse sound ensues that cannot be produced by a traditional fuzz box (see figure 3). Because of space considerations, we cannot describe all possible sound variations, but hope that the examples given will spur you on to further experimentation.

Power output stage

A low-weight, portable guitar amplifier of modest dimensions requires a power output stage that is small, reliable, efficient,
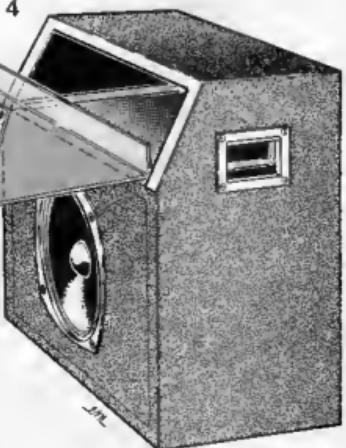
and yet provides sufficient power to prevent the audience reaching for their hearing aids! To cut a long story short, we have opted for the 30-watt output stage described elsewhere in this issue. The complete set-up gave very satisfactory results during tests and operation in small to medium-sized halls; it would probably not be quite suitable for use in larger spaces

Power supply

Power for the pre-amplifier is derived from the supply of the 30-watt output stage: the printed circuit of that stage is already fitted with appropriate take-off terminals. If you do not use the 30-watt output stage, you need a supply that is capable of providing $\pm 18 ... \pm 35$ V at 35 mA (positive line) and 22 mA (negative line).

Figure 5. View of the top of the chassis and the rear of the front panel. It is important to adhere to the wiring layout to prevent earth loops. Do not forget to insulate all sockets from the front panel!

All dashed lines indicate screened cables. Screens must only be connected to earth where shown or at the front panel
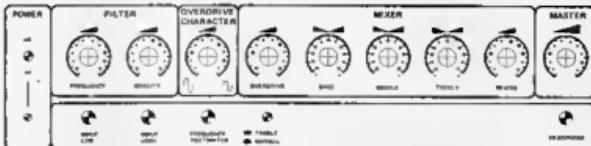
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Loudspeaker

Never use a loudspeaker designed for use in hi-fi installations for the following reasons. The amplitude of a vibrating guitar string is not particularly large: in the case of the top strings, it is hardly possible to see with the naked eye whether they are vibrating, especially if plucked only lightly. The amplification consequently required to make them audible is quite considerable.

If, however, a string is plucked vigorously, because of the type of music or the temperament of the player, it is deflected quite a distance from its rest position. The instantaneous voltage then induced in the pick-up coil reaches a very high peak and, in modern hi-fi equipment, this is faithfully transferred to the loudspeaker which in consequence may easily be damaged or destroyed. Fortunately, there are loudspeakers available which have been specially designed for use with electric guitars. They are characterized by a very rigid suspension of the cone and their ability to cope with the wide dynamic range of electric musical instruments.

Basically, the loudspeaker should be able to handle not less than 50 watts (sine wave) and have an input impedance of 8 ohms or 4 ohms. Note that mid-range and treble speakers are not just superfluous but unwanted! After all, we are not looking for linear transfer of the guitar sounds: it is the unfaithful reproduction that makes the music of electric guitars so popular!

Construction

The complete amplifier, including loudspeaker, may be housed in a case as suggested in figure 4. The shape corresponds closely to current market trends. Dimensions are primarily dependent upon the loudspeaker: unfortunately, we have not been able to find a suitable speaker that would make it possible to house the amplifier in a truly portable case.

The top compartment of the wooden case offers ample space for the pre-amplifier, output stage, and power supply.

The best material to use is 19 mm (¾ inch) chipboard. The various panels should be fastened together with dowels and wood glue: if you must, chipboard screws may also be used.

The completed case may then be covered in, say, black leatherette and the corners protected by suitable metal corner caps to give the whole a near-professional appearance.

The electronic circuits are best mounted onto a chassis which is fastened to the front panel so that the whole may be slid in and out of the upper compartment. The front panel should be fastened to the case with suitable chipboard screws. It is wise to fit two runners at the underside of the chassis so that it does not lie direct on the wood: this has the advantage that any fixing screws for the transformers, PCBs, and so on may protrude from the underside of the chassis without causing any problems.

Some useful hints

If the wiring to the potentiometers, the power lines, and the wires connecting the pre-amplifier to the output stage are not placed with careful thought, it may easily happen that hum is audible from the speaker even when all potentiometers are turned off. One of the prime causes of this is that the earth connections to various parts are not radial but form a loop. If such a loop lies in a stray field, a voltage may be induced in it which is then superimposed onto the signal. Care should also be taken that signal paths and wires (including PCB tracks) carrying unsmoothed alternating currents never have a common return. It is for these reasons strongly recommended to adhere to the interconnection wiring layout in figure 5.

When fitting the reverb unit, care should be taken that the output coil is not too close to the mains transformer. The screened housing of the unit does not offer all that much protection against the strong electro-magnetic field existing in the immediate vicinity of the transformer. It is also best not to fit the unit rigidly to the chassis to prevent any mechanical vibrations from the loudspeaker and mains transformer being transferred to it. This is easily accomplished with a felt washer fitted between the chassis and the unit housing: it is wise to glue the washer in place.

portable guitar amplifier
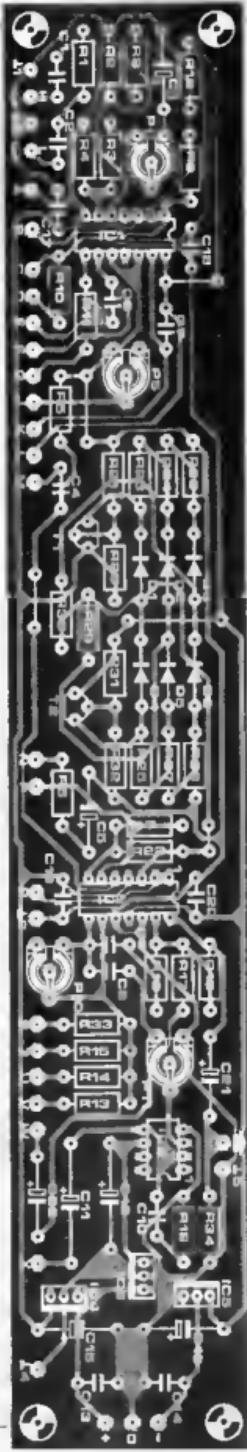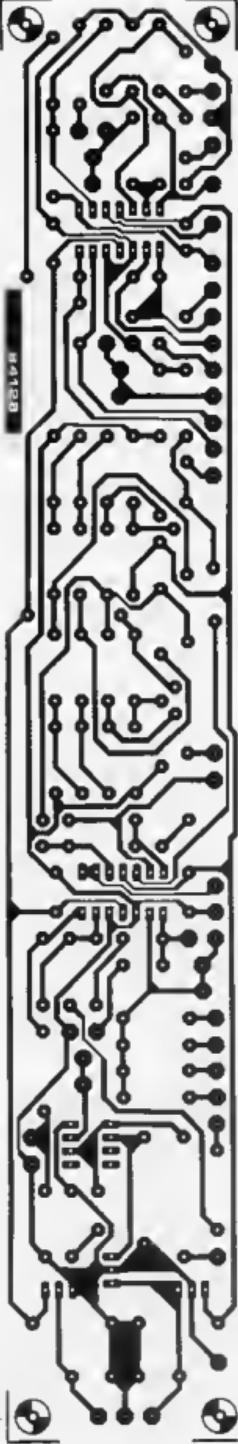
Parts list

Resistors:
R1 = 470 k
R2,R5,R33 = 1 M
R3,R17,R22,R29 = 2k2
R4,R7,R12,R13,R18,R19, R34 = 100 k
R6,R25,R28 = 220 Ω
R8,R9 = 47 k
R10,R11,R20,R31 = 10 k
R14,R15 = 22 k
R16 = 10 Ω
R21,R32 = 3k3
R23,R26 = 150 Ω
R24,R27 = 270 Ω
R30 = 8k2
P1 = 50 k preset, PCB type
P2 = 10 k potentiometer, log.
P3 = 100 k potentiometer, stereo, lin.
P4 = 100 k potentiometer, lin.
P5 = 25 k preset, PCB type
P6,P7,P8,P9,P12 = 100 k potentiometer, log.
P10 = 500 k preset, PCB type
P11 = 10 k preset, PCB type

Capacitors:
C1,C2,C4,C8,C10 = 47 n
C3,C5,C15,C16, C21 = 10 μ/35 V
C6,C7 = 3n3
C9,C17,C18,C19, C20 = 100 n
C11 = 220 μ/25 V
C12 = 47 μ/25 V
C13,C14 = 330 n
C22 = 2μ2/10 V

Semiconductors:
D1...D6 = 1N4148
T1 = BC 547
T2 = BC 557
IC1,IC2 = TL 074 (TL 084)
IC3 = LM 386
IC4,IC6 = 7815
IC5 = 7915

Miscellaneous:
S1 = SP5T switch
2 jack socket, 6.3 mm, mono, without switch
1 jack socket, 6.3 mm, mono, with switch
printed circuit board 84128

Figure 7. The printed circuit of the pre-amplifier.

The input of the device is formed by differential amplifier T2/T3 whose emitter resistance consists of current source T1/D1/D2. The output of the amplifier is fed to driver T6 whose collector resistance is formed by current source T4. Transistor T5 provides a constant base voltage for the output transistors; the bias allows a quiescent current of about 50 mA through these transistors. The output transistors are arranged as quasi-complementary darlingtons: T7 and T9 form an n-p-n darlington; T8 and T10 are the complementary pair. Any asymmetry of T8/T10 is negated by diode D5.

Circuit description

The amplifier circuit, shown in figure 2, is designed for operation from a ± 25 V symmetrical power supply. The full 50 V, decoupled by C4 and C8, is applied to the output transistors, pins 6 and 8, in the STK077. The supply for the input and driver stages is decoupled by R3/C3 and R5/C7 respectively and applied to pins 10 and 4. Feedback is arranged by connecting the output, pin 7, to the inverting input of the differential amplifier, pin 3, via R6.

The gain, A, is determined by the feedback factor, that is, the ratio R6:R4 —

$$A = (R6 + R4)/R4$$

When planning the power output stage for the guitar amplifier described elsewhere in this issue, we came across an interesting hybrid IC made by Sanyo: the STK077. In contrast to a monolithic IC where all the components are manufactured into or on top of a single chip of silicon, a hybrid IC consists of several separate component parts, attached to a ceramic substrate, that are interconnected by an appropriate metallization pattern or by wire bonds. Hybrid ICs are frequently encountered in medium power (30...60 W) hi-fi equipment. They are, however, also eminently suitable for use in home-constructed amplifiers, because they are more reliable and smaller than circuits built up from discrete components, and are not so vulnerable and susceptible to oscillation as monolithic IC stages. Hybrid ICs are generally available with power ratings up to 70 W and are normally reasonably priced.

30 watt a.f. output stage

based on a new hybrid IC

The STK077 is a 30 watt hybrid IC that is ideal for use in small mono a.f. amplifiers or medium power stereo equipment. It is also of interest to those who want a small amplifier for fitting into an active loudspeaker box.

The innards of the STK077 form a fairly conventional a.f. amplifier circuit as can be seen from figure 1. Note that the power transistors are mounted direct on the cooling area of the device to ensure better heat dissipation.

With values shown this amounts to nearly 27 dB.

The input signal is applied to the non-inverting input (pin 1) of the differential amplifier. DC zero potential is ensured by resistor R2, which carries the base current for T2. Pin 2 is the earth terminal which is internally connected to the metal base of the STK077.

Stability of the output transistors is arranged by various methods. On board there is a Miller capacitor between base

and collector of driver T6, while externally capacitor C5 is connected between pins 3 and 8. Capacitor C9 and resistor R7 at the output, pin 7, ensure a defined load at high frequencies and this enhances the stability under no-load conditions. Finally, the input has been provided with an RC low-pass filter (R1/C1) which increases the rise time of the input signal and so reduces the transient intermodulation distortion (TIM).

The power supply, with the exception of the mains transformer, is housed on the same printed circuit as the amplifier. It consists of an unregulated circuit: four standard diode rectifiers, and two electrolytic smoothing capacitors each shunted by a foil capacitor.

When a mains transformer with 2 x 18 V secondary is used, the direct output voltage under no-load conditions is of the order of 25 V, falling to about 22 V when a normal load is connected. A 1 A transformer can be used for output powers up to 20 W.

More power

With a ± 20 V power supply, the STK077 provides 20 watts into 8 ohms, or 30 watts into 4 ohms. In the latter case, both the third harmonic distortion and the current consumption are somewhat higher than in the former. If you want higher power, one of the series STK078...STK083 may be used: the printed circuit board remains unchanged, but it is, of course, necessary to use an appropriate mains transformer and higher rated electrolytic capacitors. Data for these are given in table 2.

Practical tips

The mains transformer may have a single centre-tapped secondary or two separate windings. In the latter case, proceed as follows: connect one of the terminals of secondary 1 with one of secondary 2 and measure the a.c. voltage between the two free terminals. If this is 0 V, the terminals of ONE of the secondaries must be changed over so that across the free terminals an a.c. voltage of twice the rating of one of the secondaries is measured. The interconnected terminals become the equivalent of the centre tap which is connected to earth as shown in figure 2.

STK077
POWER AMPLIFIER

1

30 watt a f output stage
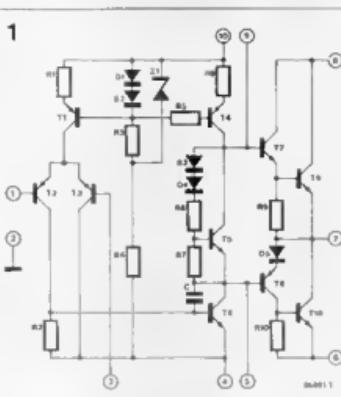
Figure 1. The circuit of the STK077. Note that the power transistors are connected in a quasi-complementary darlington arrangement.

2

STK-077

1 2 3 4 5 6 7 8 9 10
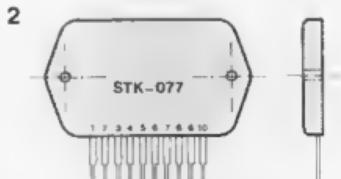
Figure 2. Circuit diagram of the amplifier and its unregulated, symmetrical power supply.
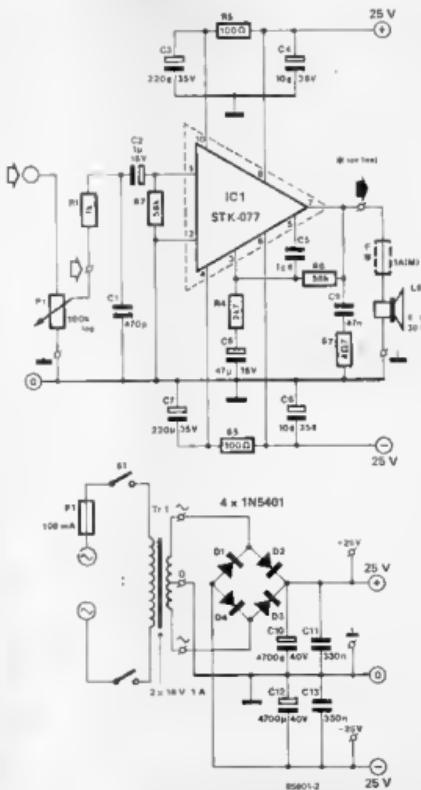
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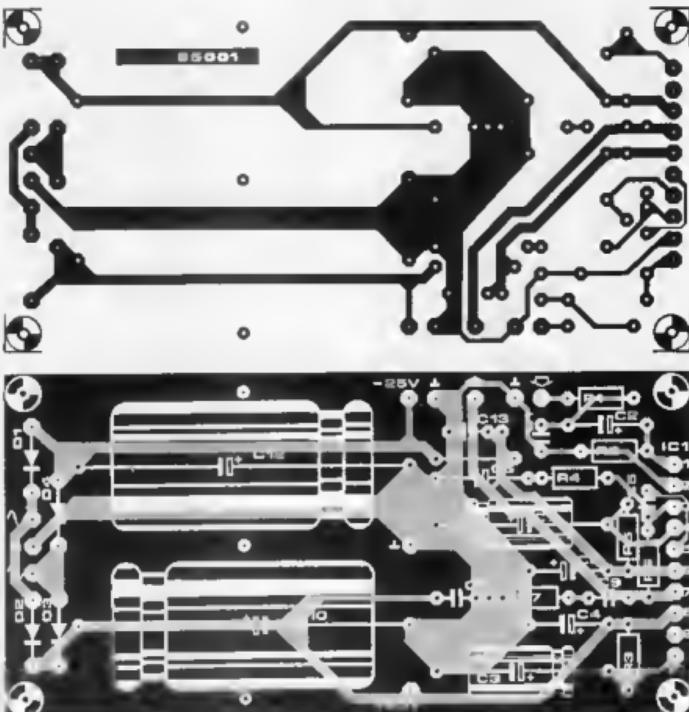
Parts list

Resistors:
R1 = 1 k
R2,R6 = 56 Ω
R3,R5 = 100 Ω
R4 = 2k7
R7 = 497
P1 = 100 k logarithmic potentiometer

Capacitors:
C1 = 470 p
C2 = 1 μ/16 V
C3,C7 = 220 μ/35 V
C4,C8 = 10 μ/35 V
C5 = 1p8
C6 = 47 μ/16 V
C9 = 47 n
C10,C12 = 4700 μ/40 V
C11,C13 = 330 n

Semiconductors:
D1 . D4 = 1N5401
IC1 = STK077 (or STK078 STK083)

Miscellaneous:
S1 = DPST mains switch
Tr1 = mains transformer, secondary 2 × 18 V/1 A
F1 = fuse, 100 mA, slow blow
F2 = 1 A (8 Ω loudspeaker) or 1.6 A (4 Ω loudspeaker) slow blow (2 seconds)
Heat sink temperature rise 1.7 K/W
Printed circuit board 85001

Figure 3. The printed circuit board is not only for use with the STK077 but also with other members of the family, STK078 STK083, which give output powers of not less than 24 . .40 watt into 8 ohms, depending on which member is used.

Failure of one of the power lines during operation of the amplifier would destroy the IC. It is, therefore, vital to ensure that both power lines are connected properly at all times. Furthermore, under no circumstances should either the positive or the negative line be protected by a fuse. It is, of course, also important that the

voltages across the two secondaries are identical.
The value of the thermal rating of the heat sink stated in the parts list and table 2 applies to the amplifier being driven hard.
If the amplifier is intended for domestic (music) use only, the value may be

Table 1

Supply voltage, U_b — maximum	± 32 volts
— recommended	± 22 volts
Case temperature — maximum	85°C
Short-circuit duration — maximum	2 seconds
Load resistance — recommended	8 ohms
— minimum	4 ohms
Quiescent current — maximum	100 mA
— typical	50 mA
Power into 8 ohms — at 1 W into 8 Ω	20 watts
Bandwidth — at 1 W into 8 Ω	10 Hz . . 100 kHz
— at 20 W into 8 Ω	10 Hz . . 30 kHz
Output direct voltage — maximum	± 70 mV
Input voltage (rms) — for 20 W into 8 Ω	800 mV
— for 20 W into 4 Ω	500 mV
Input impedance	50 kilohms
Current consumption — at 20 W into 8 Ω	1 A
— at 30 W into 4 Ω	1.5 A

* In range 20 Hz . .20 kHz, THD = 0.3%, U_b = ± 22 V

Table 1. Brief technical characteristics of the STK077.

somewhat lower. It may be useful to drill out the fixing holes in the heat sink slightly to avoid mechanical stress during its installation. The use of silicone heat transferring grease is strongly recommended.

If the amplifier is used for mono applications, a fuse may be fitted in the loudspeaker lead: this is shown in dashed lines in figure 2. The fuse should be of the medium-slow (about 2 seconds) type.

Values for use with other ICs in the series are given in table 2.

The heat sink and printed circuit board should be fixed to a chassis with the aid of an aluminium angle piece as shown in figure 4, since the terminals of the IC cannot support the PCB.

As always in a.f. amplifiers, the wiring layout should be thought out carefully. On the premiss that any wires may cause problems, we reduced the amount of wiring by designing the power supply and the amplifier on one PCB. The only wires consequently required are three to the mains transformer, two to the loudspeaker, and a screened one for the input signal.

If two amplifiers are built for stereo applications, it is possible to use one mains transformer of twice the rating stated in the parts list. Separate power lines should then be connected to each of the PCBs. Separate earth return lines for each loudspeaker are also required: this means that each PCB is connected to the appropriate loudspeaker by a two-core cable.

If you want to use a 4-ohm loudspeaker, the mains transformer should be capable of providing a secondary current of 1.5 A; alternatively, a transformer with lower secondary voltage (2 × 15 . . . 16 V) may be used. The heat sink should also be adapted to the higher dissipation, for instance, 1.5 K/W instead of 1.7 K/W The only other difference between the 4-ohm and 8-ohm versions is that the former has a slightly higher third harmonic distortion (THD) factor than the latter, as shown in figure 5.

Table 2

		STK078	STK080	STK082	STK083
Supply voltage — max.	V	± 35	± 39	± 43	± 46
— recommended	V	± 25	± 27	± 30	± 32
Load resistance — recommended	Ω	8	8	8	8
Output power into 8 Ω — minimum	W	24	30	35	40
Secondary transformer voltage — recommended	V	2 × 20	2 × 22	2 × 25	2 × 27
Transformer rating	VA	≧50	≧60	≧75	≧90
Rating of electrolytic capacitors (C3,C4,C7,C8,C10,C12)	V	35/40	≧50	≧50	≧50
Fuse rating (8-ohm loudspeaker)	A	1.2	1.6	1.6	2
Heat sink — temperature rise	K/W	≤1.5	≤1.4	≤1.2	≤1

Table 2. Variations in ratings if higher power ICs from the STK series are used.
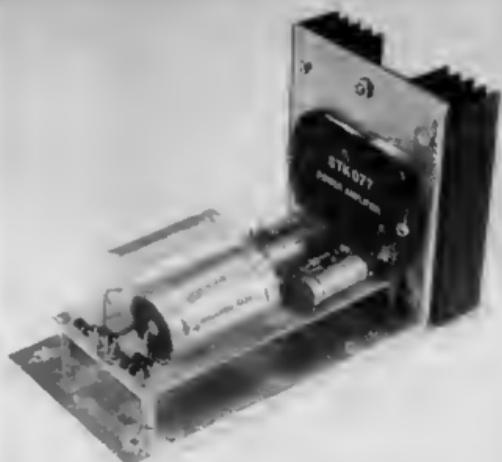
Figure 4. To avoid mechanical strain on the IC terminals, the printed circuit board, heat sink, and IC should be fitted together as shown.
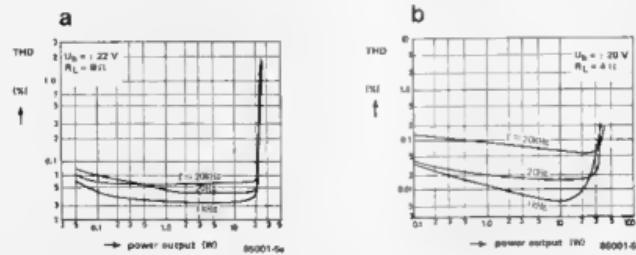
Figure 5. Third harmonic distortion vs output power in the STK077: in 5a with 8-ohm load and ± 22 V supply; in 5b with 4-ohm load and ±20 V supply.

Whenever two (or more) large programs must exist at the same time in a 6502's memory there is bound to be a conflict as regards page 0 and the stack (page 1). The situation could arise if a BASIC interpreter and a DOS (disk operating system), or both of these and a video handler, are being used simultaneously. One of the accepted methods of solving this problem involves reserving two areas in random access memory where these pages are 'duplicated', giving, for example, E000...E0FF for page 0 and E100...E1FF for the stack. Every time the computer changes from one program to the other the contents of these areas of RAM are swapped with the appropriate contents of pages 0 and 1. This removes any possibility of corrupting the pointers on page zero or the contents of the stack.

JSR SWAP

software to interchange pages 0 and 1 in a 6502-based system

Table 1. The program leaves the SWAP routine (which it entered using JSRI) not by RTS but by the jump instruction identified by the label JMPINS. The return address is 'stacked' at the start of the routine, then corrected (return address = address left + 1) and then placed after the jump instruction identified by the label JMPINS.

One of the notable characteristics of the 6502 microprocessor is the way it uses pages 0 and 1. The 256 bytes from 0000HEX to 00FFHEX can be addressed using commands specific to this zone. This is known as page zero addressing; the most significant address byte is not specified as it is implicit in the operation code. These same 256 bytes can be used as 16-bit pointers for indirect indexed addressing of the rest of the memory. The 256 bytes from 0100HEX to 01FFHEX form the 6502's stack. This is a register generated by the processor itself to enable it to store certain information. It operates on the principle of 'last in first out' so the processor can only work with the last item stored on the stack. An internal stack pointer continually indicates the address of this last item.

It is obvious that the slightest careless change of the parameters saved on these two pages will upset the operation of a program that is being run — usually with no possibility of correcting the error. When two programs are run in parallel it is essential that they do not destroy each other's page 0 and 1 parameters. This is an extra worry for the programmer, particularly as it can be an insoluble problem. As soon as the programs that are being run reach a certain size it is better to find a way to break away from the shackles governing the use of pages 0 and 1.

The routine proposed here is used to move the contents of page zero and page one to another area of RAM where they can be changed at will. At the same time the contents of the RAM area in question is transferred to pages 0 and 1. The name of the routine is SWAP, for obvious reasons.

Using this routine means that the programmer no longer has to worry about the contents of pages zero and one when leaving one program to carry on another. All he has to do is run the SWAP routine. Page zero (0000HEX.. 00FFHEX) and page one (0100HEX.. 01FFHEX) of the first program are saved at E000HEX..E1FFHEX, and the contents of pages 0 and 1 for the second program, which had been stored at E000HEX..E1FFHEX, are transferred to 0000HEX..01FFHEX. When returning from the second to the first program the SWAP routine is again executed and the same procedure is carried out again to return the two pairs of pages zero and one to their original locations. The locations we have used to store pages 0 and 1 (E000HEX...E1FFHEX) can, of course, be changed to suit the system with which the SWAP routine is used, provided the area reserved is in random access memory. Similarly the SWAP routine itself must be run in RAM. A look at the last line of the listing will show why this is necessary. Indexing and swapping are the two processes that make this routine possible so bear this in mind.

```
0010:    ;*******
0020:    ;*SWAP*
0030:    ;*******
0040:
0050:
0060:
0070:    ;   SWAP PAGE 0 AND PAGE 1 WITH E000 AND E100
0080:
0090:
0100:
0110:
0120: E200             PZ      =   0000    PAGE ZERO
0130: E200             STACK   =   0100    STACK AREA
0140: E200             SPZ     =   E000    SWAPPED PAGE ZERO
0150: E200             SSTACK  =   E100    SWAPPED STACK AREA
0160:
0170:
0180:
0200: E200  18         SWAP    CLC
0210: E201  68                 PLA
0220: E202  69 01               ADDIM #01    PUT RETURN ADDRESS
0230: E204  8D 2C E2            STA  JMPINS +01  JUST BEHIND A JUMP-OF-CODE
0250: E207  68                 PLA
0260: E208  AA                 TAX
0270: E209  90 01              BCC  SW
0280: E20B  E8                 INX
0290:
0291: E20C  8E 2D E2  SW       STX  JMPINS +02
0300: E20F  A2 00             LDXIM #00    RESET INDEX
0310:
0320: E211  BD 00 01  SWAPST   LDAX STACK   GET BYTE FROM PAGE 1
0330: E214  BC 00 E1           LDYX SSTACK  GET BYTE FROM SWAP AREA
0340: E217  9D 00 E1           STAX SSTACK  SAVE BYTE FROM PAGE 1 IN SWAP AREA
0350: E21A  98                 TYA
0360: E21B  9D 00 01           STAX STACK   SAVE BYTE FROM SWAP AREA IN PAGE 1
0370: E21E  BD 00 00           LDAX PZ      GET BYTE FROM PAGE 0
0380: E221  BC 00 E0           LDYX SPZ     GET BYTE FROM SWAP AREA
0390: E224  9D 00 E0           STAX SPZ     SAVE BYTE FROM PAGE 0 IN SWAP AREA
0400: E227  98                 TYA
0410: E228  9D 00 00           STAX PZ      SAVE BYTE FROM SWAP AREA IN PAGE 0
0420: E22B  E8                 INX          BUMP THE OFFSET
0430: E22C  D0 E6              BNE  SWAPST  NOT DONE, KEEP ON
0440: E22E  4C FF FF  JMPINS   JMP  $FFFF   SELF MODIFYING CODE'''
0450:
```

BASIC (PART 3)

Most of the principles involved in programming in BASIC have already been discussed. Most of the remaining BASIC statements will now be explained.
Having studied this third part of the series, it should be possible to write even fairly complicated programs; the last part of the series will deal with trouble-shooting in programs ('de-bugging').

There are several ways to enter data into a computer. If something is to be calculated, the 'data' usually consists of numbers. In Part 2, these were entered by assigning values to variables or by entering the numbers as part of the program. However, a different approach will often prove more useful in practice, using either of two further BASIC statements: INPUT and READ . . . DATA . . .

INPUT

By using the INPUT statement, data can be entered while the program is running. Or, to be more precise: when the computer finds an INPUT statement in the program, it stops and waits for the data to be entered before continuing with the program.
The complete statement consists of 'INPUT' followed by the name of a variable. For example:

```
> 10 INPUT A
> 20 PRINT A
> 30 END
> RUN
? 256
256
BRK AT 30
>
```
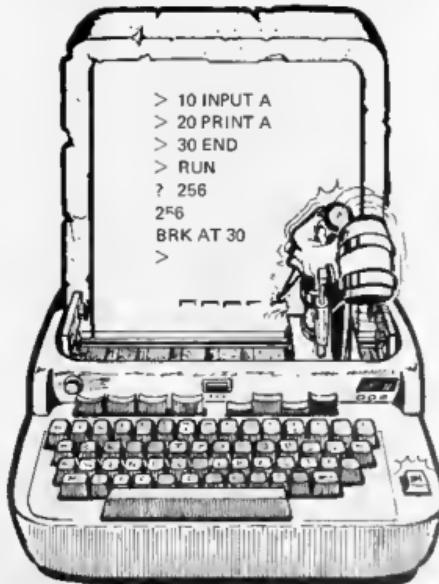
When running the program, the computer prints a question mark as soon as it finds the INPUT statement on line 10. It then waits until a number is entered, followed by the CR key. As soon as the number '256' was entered, this value was assigned to the variable A — after which the rest of the program could be carried out.
The same statement can be used to assign values to several variables at the same time: 'INPUT A, B, C, . . .'. When the question mark is printed, all corresponding numbers must be entered: '? 123, 62, 23, . . . '. The INPUT statement can also be used to enter text variables, as will be explained in Part 4. The advantage of using the INPUT statement is that it opens the possibility of dialogue with the computer. Depending on intermediate results, for instance, the program can be re-run with new values until a desired final result is achieved. For instance, let us assume that we wish to know the returns after a certain number of years (N) from a personal investment (I) at various interest percentages (P). The final value (F) is equal to:

$$F = I \times (1 + \frac{P}{100})^N$$

In BASIC, this becomes:
$F = I * (1 + P/100) \uparrow N.$

A suitable program is therefore as follows:

```
> 10 PRINT "ENTER INITIAL INVESTMENT,"
> 11 PRINT "INTEREST RATE"
> 12 PRINT "AND NUMBER OF YEARS"
> 20 INPUT I, P, N
> 30 LET F = I * (1 + P/100) ↑ N
> 40 PRINT "THE FINAL VALUE IS", F
> 50 END
> RUN
ENTER INITIAL INVESTMENT,
INTEREST RATE
AND NUMBER OF YEARS
? 1000, 9, 10
THE FINAL VALUE IS 2367
>
```

In other words, with an initial investment of £ 1000 and a 9% interest rate, the final value after 10 years will be £ 2367.

READ . . . DATA . . .

Another way of entering data is the use of so-called 'data blocks'. A data block is a group of numbers, preceded by the DATA statement; the various numbers and/or texts are separated by comma's. A

BASIC (PART 3)

data block is usually located at the end of a program; it is entered prior to running the program. In the main program, READ statements are used to recall the data as required; each new READ statement causes the next number, text or group of data to be recalled. For example:

```
> 10 READ A, B, C
> 20 D = A • B • C
> 30 READ E
> 40 F = D/E
> 50 PRINT D, F
> 60 DATA 1, 2, 3, 3
> 70 END
> RUN
6  2
BRK AT 70
>
```

In line 10, the first three data are read from the data block on line 60 and assigned to the variables A, B and C. In other words, A becomes 1, B becomes 2 and C becomes 3. Then D is calculated (line 20); the next number is recalled from the data block (line 30: E becomes 3); and so on.
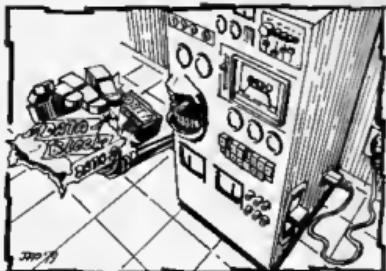
As can be seen, several READ statements can be used — the data will be read out consecutively. Similarly, several DATA statements can be used; however, there is little point in this — they are simply used in consecutive order — and it makes it more difficult to locate and modify the data at a later date.

Obviously, it is important that enough data is stored in the data block. If, after reading the last of the data, the program encounters a further

READ statement, it will print some comment on the lines of 'OUT OF DATA IN xxx' (where 'xxx' is the line number of the READ statement where the data block proved to be empty).

In some BASIC dialects, the same data block contents can be re-used several times: the RESTORE statement causes the computer to start again at the beginning of the block.

The READ . . . DATA . . . statements are not known in NIBL.

REM

The REM statement (for 'REMark') is used to add explanatory text to the program, as an aid to the programmer. The text is simply entered after the REM statement; it will be ignored by the interpreter, but will reappear when a LIST command is given.

This statement will prove its value when a program has not been used for some time: it serves as a quick reminder of the meaning of variables, the importance of sections of program, etc. For example, in the 'personal investment' program given above:

```
> 1 REM THIS PROGRAM CALCULATES
> 2 REM THE FINAL VALUE F
> 3 REM OF AN INITIAL INVESTMENT I
> 4 REM AFTER N YEARS
> 5 REM AT AN INTEREST RATE OF P PERCENT
> 10 PRINT "ENTER INITIAL INVESTMENT:
> 11 PRINT "INTEREST RATE"
> 12 PRINT "AND NUMBER OF YEARS"
> 20 INPUT I, P, N
> 30 LET F = I • (1 + P/100) ↑ N
> 40 PRINT "THE FINAL VALUE IS", F
> 50 END

(etc.)
```

The program itself is not affected in any way by the REM statements; they only reappear when a program listing is requested. They do, of course, use up some memory space — but there will normally be sufficient memory available.

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Standard functions

To simplify programming in BASIC, 10 standard mathematical functions are available:

standard function	description
SIN(X)	sine x
COS(X)	cosine x }x is the angle in radials.
TAN(X)	tangent x
ATN(X)	arctangent x.
EXP(X)	e^x (e = 2.71828 ...).
ABS(X)	|x|, the absolute value of x.
LOG(X)	^{e}log |x| = ln |x|, the natural logarithm.
SQR(X)	$\sqrt{|x|}$, the square root of |x|.
INT(X)	the integer of x.
SGN(X)	the 'sign' of x: 1 if x is positive, −1 if x is negative, 0 if x is zero.

Note that for the sine, cosine and tangent functions the angle must be expressed in radials (1 rad. = 57.2958°, or 1° = 0.017483 rad.); for the arctangent function, the result will be expressed in radials. The LOG (natural logarithm) and SQR (square root) functions automatically take the absolute value of x, without giving an error indication. For instance: SQR (−4) = 2.

The INT (integer) function sometimes causes confusion. It produces the largest whole number that is smaller than or equal to x. For positive numbers, this simply means omitting the decimal fraction: INT (2.78) = 2. For negative numbers, however, the result is one less than one might assume: INT (−2.78) = −3!

As can be seen from the examples given above, the number ('X') must always be included in brackets. Usually, it is also permissible to use a variable or even an algebraic expression here. A few examples are given in the following program:

```
> 10 REM EXAMPLES OF STANDARD FUNCTIONS
> 20 PRINT "ENTER ANGLE IN DEGREES"
> 30 INPUT A
> 40 LET 8 = A/57.2958
> 50 REM 8 IS ANGLE IN RADIALS
> 60 PRINT "THE SINE OF"; A; "DEGREES IS"; SIN (8)
> 70 PRINT "INT (5.3) = "; INT (5.3)
> 80 PRINT "INT(−8.5) = "; INT(−8.5)
> 90 PRINT "INT(7)="; INT(7)
> 100 END
> RUN

ENTER ANGLE IN DEGREES
? 30
THE SINE OF 30 DEGREES IS 0.5
INT(5.3) = 5
INT(−8.5) = −9
INT(7) = 7
BRK AT 100
>
```

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It shouldn't come as a surprise that there are exceptions to the rules given above. Some BASIC dialects don't use the absolute value of x in the LOG and SQR functions if a negative value is entered for x — they print an error indication.

In all program examples so far, the programs were executed in a fixed (numerical) order. The statement with the lowest line number was carried out first, and so on. Where a different order is required, 'jump statements' can be used: 'GOTO', 'IF . . . THEN . . .' and so on.

In Tiny BASIC (and, therefore, in NIBL) these standard functions are unknown. For that matter, the INT function would be pointless, since Tiny BASIC only recognises whole numbers in the first place; NIBL does recognise certain other functions; these will be discussed in part 4.

... The computer will continue to run around this loop ...

Jump statements

In all program examples given so far, the programs were executed in a fixed (numerical) order. The statement with the lowest line number was carried out first, and so on. Where a different order is required, 'jump statements' can be used: 'GOTO', 'IF . . . THEN . . .' and so on.

GOTO

The GOTO statement is used when a 'jump' to a specified line number is required in the program. Since no initial check is required (to see whether certain specified conditions are met), GOTO is known as an 'unconditional statement'. An example:

```
> 5 N = 0
> 10 PRINT N
> 20 N = N + 1
> 30 GOTO 10
> 40 END
> RUN
0
1
2
3
(etc.)
```

In this program, N is first made equal to zero and this value is printed. In line 20, the value is increased by 1 ('incremented'); line 30 initiates a jump back to line 10, where the value (1) is printed; and so on. The computer will continue to run around this 'loop', printing all numbers from 0 up — until it either runs out of paper or reaches its maximum count.
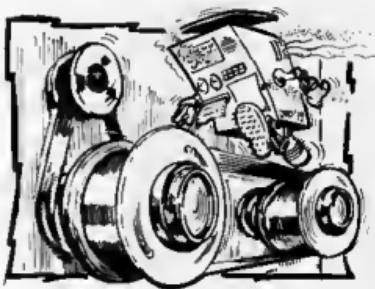
Obviously, if something like this occurs when running a program (due to a programming error), there must be some way to stop the computer. There is: the 'BREAK' key. As soon as this key is operated, the computer will stop the program and print out 'BRK AT 20' or something similar.

IF . . . THEN . . .

Often, a jump to a new line number is only required if certain conditions are fulfilled. The 'conditional statement' offers the possibility of executing different sections of program, depending on intermediate results. In general, the statement will be entered as

IF (relational expression) THEN (line number).

The 'relational expression' will normally be some kind of comparison, for instance 'X = 10' or 'A > B'. If the result of the comparison is 'true', the computer will jump to the specified line number; if not, it will simply proceed to the next program line.

As an example, let us assume that a program is required that will print out the multiplication table of any number ('X'). The flow chart for a suitable program would be as shown in figure 1. The sequence of operations is as follows: after the 'initialisation' procedure — in this case, making N = 1 — the computer will request the value of X (step 2). In steps 3 and 4, 1 × X is calculated and printed, after which N is incremented by 1 (step 5). Then, in step 6, the value of N is checked: if it is smaller than or equal to 10, the computer must jump back to step 3 for the next calculation. After calculating and printing all values up to 10 × X, N will become 11 in step 5, the result of the comparison in step 6 will be 'false' and the program will have reached the END.

1
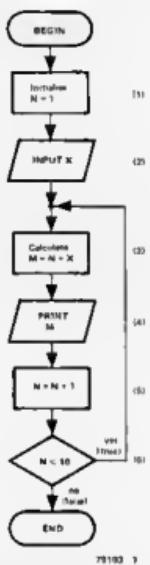
The corresponding program is as follows:

```
> 10 REM THIS PROGRAM CALCULATES
> 15 REM AND PRINTS 1xX THROUGH 10xX
> 20 LET N = 1
> 30 PRINT "ENTER X"
> 40 INPUT X
> 50 LET M = N x X
> 60 PRINT N; "x"; X; "="; M
> 70 LET N= N+1
> 80 IF N < = 10 THEN 50
> 90 END
> RUN
ENTER X
? 12
1 x 12 = 12
2 x 12 = 24
: (etc.)
10 x 12 = 120
BRK AT 90
>
```

In some BASIC dialects (including NIBL), it is possible to enter a statement after THEN, instead of a line number. In that case, if the result of the comparison is true the statement will be executed; otherwise it will be ignored.

In NIBL, if a jump to a line number is required 'GOTO' must be used instead of 'THEN'. The two possibilities are therefore as follows in NIBL:

IF (relational expression) GOTO (line number), and
IF (relational expression) THEN (statement).

FOR . . . NEXT . . .

Another way to run through the same section of program several times in succession is by means of the FOR . . . NEXT . . . statement. This really consists of two statements that must be entered on different program lines, as illustrated in the following example:

```
> 10 FOR A= 1 TO 5
> 20 PRINT A↑2
> 30 NEXT A
> 40 END
> RUN
1
4
9
16
25
BRK AT 40
>
```

In the FOR statement, a variable (A), an initial value (1) and a final value (5) are specified. The section of program between the FOR and the NEXT statements is executed several times, starting with the initial value for A and then incrementing it by 1 until the final value is reached.

The initial and final values need not be given as numbers. Both variables and expressions can also be used (for instance: 'For A = N TO 50 * N'). The final value should, of course, be greater than (or at least equal to) the initial value.

It is not a good idea to use the 'running variable' (A in the example given above) at any other point in the program — except for other FOR statements. The statements between FOR and NEXT are often referred to as the 'FOR-NEXT' block. Before executing a FOR-NEXT block, the computer first calculates the initial and final values for the running variable. If these values depend on another variable, its value is taken at that moment; once calculated, the initial and final values remain unchanged as the block is executed, so that even if the value of the variable is then altered this will have no effect on the final value.

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A more 'practical' example of the FOR...NEXT... statements is a program for the calculation of A!. (A! = 1 * 2 * 3 * ...... * (A-2) * (A-1) * A. For example: 3! = 1 * 2 * 3 = 6. By definition, 0! = 1.) A flow chart for a suitable program is given in figure 2.

2

```
> 10 REM CALCULATION OF A!
> 20 LET N = 1
> 30 PRINT "ENTER A"
> 40 INPUT A
> 50 IF A < = 1 THEN 90
> 60 FOR X = 1 TO A
> 70 LET N = N * X
> 80 NEXT X
> 90 PRINT A; "! ="; N
> 100 END
> RUN
ENTER A
? 3
3! = 6
BRK AT 100
>
```

After writing a program like this, it must be checked with several values for the variables which should give known results (in this example, say, 3 and 5); special attention should be payed to values that require a different procedure (0 and 1 in this program).
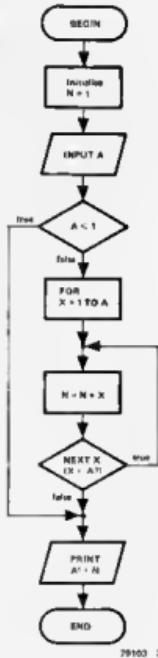
After the initialisation procedure, the value for A is entered (a positive number). If A = 0 or 1, the result will be 1 and there is no point in running through the whole program — in fact for A = 0 it wouldn't work, since the final value in the FOR statement would be less than the initial value — so the result can be printed immediately. For all other values of A, the calculation described above is performed in the FOR-NEXT block and the result is then printed. The corresponding program is as follows:

FOR ... TO ... STEP ...
In the FOR ... NEXT ... statements as described so far, the running variable is increased by 1 each time round the loop. This is not always desirable: an increase by a larger or smaller amount may be required. In that case the STEP statement can be added, as follows:

FOR I= –90 TO 90 STEP 15

The running variable will now be increased in steps of 15. Other steps can be specified, and even negative steps are permitted — in other words, the running variable is decreased step-by-step. For example:

FOR I = 0 TO –90 STEP –10

A certain amount of care must be taken to ensure that the running variable can reach its final value with the specified steps. It would be asking for trouble, for example, to specify:

FOR I = 0 TO 90 STEP –10

DO ... UNTIL ...
The DO ... UNTIL ... statement is only known in relatively few BASIC dialects. The reason for

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discussing it here is that this statement is also known in NIBL. As an example of its use, let us consider the following program section:

```
> 170 N = 0
> 180 DO
> 190   N = N+1
> 200   PRINT "EXAMPLE OF A DO — UNTIL LOOP"
> 210 UNTIL N = 5
> 220 ...
```

The statements between DO and UNTIL are repeated for as long as the expression after UNTIL remains 'false'. As soon as it becomes 'true' the computer moves out of the program 'loop' and proceeds with the following statement. In the example given above, the text will be printed 5 times.

Subroutines, GOSUB . . . RETURN

When writing large programs, 'subroutines' can be extremely useful. It is often the case that a certain section of the program is required several times — for example, a complicated input/output routine or the AI calculation described earlier. This program section is then entered after the main program, as a so-called subroutine. When it is required in the course of the main program a GOSUB statement is used, followed by the first line number of the subroutine. The subroutine itself will normally be concluded with a RETURN instruction, causing the computer to jump back to the line number immediately following the GOSUB statement in the main program.

As an example, let us assume that the computer is being used as an aid to circuit design, and that we will regularly be using two resistors in parallel to approximate a desired resistance value. A section of the main program and the subroutine might then be as follows:

```
> 110 PRINT "SELECT TWO RESISTANCE VALUES FOR R21"
> 120 GOSUB 500
> 130 A = P
    :
> 500 REM SUBROUTINE FOR CALCULATING PARALLEL RESISTANCE"
> 510 PRINT "ENTER VALUE FIRST RESISTOR"
> 520 INPUT X
> 530 PRINT "ENTER VALUE SECOND RESISTOR"
> 540 INPUT Y
> 550 P = X*Y/(X+Y)
> 560 PRINT "THE PARALLEL RESISTANCE IS", P
> 570 PRINT "IS THIS VALUE SATISFACTORY?";
> 580 PRINT "IF SO, ENTER 1; IF NOT, ENTER 0"
> 590 INPUT Z
> 600 IF Z=0 THEN 510
> 610 RETURN
```

When the computer reaches the point in the main program where it needs the value for R21 — which will be approximated by connecting two resistors in parallel — it prints out the corresponding request and then jumps to the subroutine. Once there, the request is made more specific by asking for the value of the first resistor. Having then received the second value, the parallel resistance is calculated and printed. This is followed by the query whether or not the value proves satisfactory. Depending on the answer, the computer will either repeat the subroutine (asking for new values) or else jump back to the main program and assign the calculated value to the corresponding variable (A) in the main program.

3

79163 3
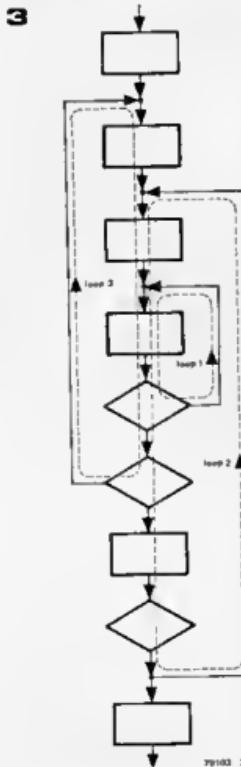
Program loops

Several statements have now been discussed that can be used when program loops are required.

Loops within loops are also possible — in fact the program given above is an example: the subroutine, effectively, is a loop; within the subroutine, an IF . . . THEN . . . statement is used to create a further loop.

Figure 3 illustrates 'loops within loops' in a flow chart. In this case, however, loop 3 is 'dangerous'. If loop 2 is either a FOR-NEXT or a DO-UNTIL loop, a jump to a line number outside this loop is not permitted: the computer must first be given the chance to complete its countdown (final value in the FOR-NEXT statement) or fulfil the condition specified (in the DO-UNTIL statement). Furthermore, a program should not loop-the-loop ad infinitum. Each loop within another loop is referred to as 'one (further) level down' and when 'nesting' loops in this way there is a maximum number of levels (depending on the BASIC dialect) that should not be exceeded. In NIBL, for instance, the maximum nesting depths for the various types of loop are as follows:

GOSUB-RETURN	B levels
FOR-NEXT	4 levels
DO-UNTIL	B levels

If the computer detects loop programming errors, it will indicate this by printing out some suitable comment. In NIBL, for instance, the indications are as follows:

Print-out	Meaning
NEST ERROR	there are too many loops within loops
FOR ERROR	FOR is not followed by NEXT
NEXT ERROR	NEXT was not preceded by FOR
NOGO ERROR	the line number indicated in a GOTO or GOSUB statement does not exist
RTRN ERROR	RETURN was not preceded by GOSUB
UNTL ERROR	UNTIL was not preceded by DO

A program example: Pay as you earn

To illustrate the possibilities of the BASIC theory discussed so far, let us consider a program that can calculate the costs involved in hire-purchase or pay-as-you-earn. In this case we will assume that:

- A fixed sum is repaid each month, which includes interest: the monthly payment M. This sum is to be calculated.
- The cost price of the item that we wish to buy is C.
- A starting capital, S, is available.
- A loan, L, is therefore required: $L = C - S$.
- The interest rate is P (%).
- The loan is to be repaid within N years.

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In the course of the program, we will also calculate:

- The amount already repayed (R) after X years.
- The amount still to be repayed (O, for Outstanding) after X years.
- The total interest, I, paid in N years.
- How many times the cost of the item has been payed after N years (F).

To sum it up, the following inputs are required:
the cost price, C;
the starting capital, S;
the interest rate, P;
the total number of years, N;
if required, an intermediate number of years, X.
Based on this data, the following will be calculated:

- The loan,
  L = C − S
- The monthly payment,

$$M = \frac{1}{12} \cdot L \cdot \frac{P}{100} \cdot \frac{\left(1 + \frac{P}{100}\right)^N}{\left(1 + \frac{P}{100}\right)^N - 1};$$

- The amount repayed after X years,

$$R = L \cdot \frac{\left(1 + \frac{P}{100}\right)^X - 1}{\left(1 + \frac{P}{100}\right)^N - 1};$$

- The amount outstanding after X years,
  O = L − R;
- The total interest payed in N years,
  I = M · 12 · N − L;
- How many times the cost has been payed,

$$F = \frac{I + L}{L}.$$

In these calculations, $\left(1 + \frac{P}{100}\right)^N$ appears several times. It is a good idea to calculate it at the beginning of the program and assign the value found to a further variable, V. The various formulae will then appear in BASIC as follows:

```
L = C − S
V = (1 + P/100)↑N
M = L * (P / 1200) * V / (V−1)
R = L * ((1 + P/100) ↑ X − 1) / (V − 1)
O = L − R
I = M * 12 * N − L
F = (I + L)/L
```

Which brings us to the flow chart and program:
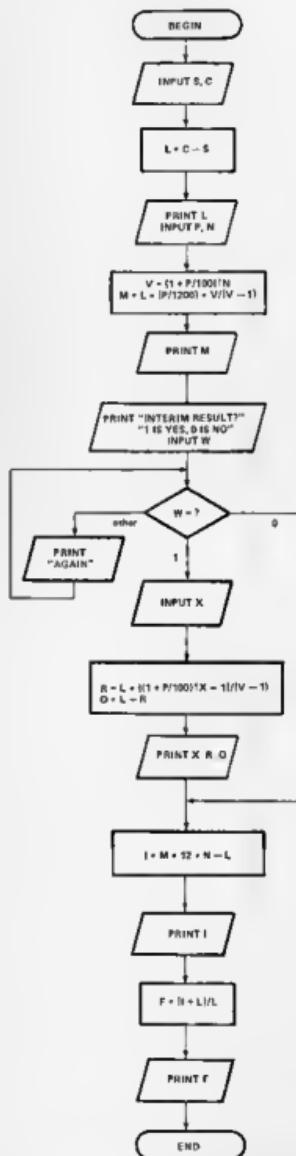
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```
> 10 PRINT "ELEKTOR SOFTWARE SERVICE"
> 20 PRINT "LOAN REPAYMENT"
> 30 PRINT "WHAT STARTING CAPITAL IS AVAILABLE?"
> 40 INPUT S
> 50 PRINT "WHAT IS THE COST PRICE?"
> 60 INPUT C
> 70 LET L = C - S
> 80 PRINT "THE NECESSARY LOAN IS"; L; "POUNDS"
> 90 PRINT "WHAT IS THE INTEREST RATE?"
> 100 INPUT P
> 110 PRINT "IN HOW MANY YEARS MUST THE LOAN BE REPAID?"
> 120 INPUT N
> 130 LET V = (1 + P/100) ↑N
> 140 LET M = L * (P/1200) * V / (V - 1)
> 150 PRINT "THE MONTHLY PAYMENT IS"; M; "POUNDS"
> 160 PRINT "OO YOU WISH TO KNOW THE AMOUNTS ALREAOY REPAYED"
> 170 PRINT "AND STILL OUTSTANOING AFTER A NUMBER OF YEARS?"
> 180 PRINT "PLEASE ENTER 1 FOR YES, OR 0 FOR NO"
> 190 INPUT W
> 200 IF W=0 THEN 320
> 210 IF W=1 THEN 250
> 220 PRINT "PLEASE STUDY THE KEYBOARO CAREFULLY, THEN"
> 230 PRINT "ENTER 1 FOR YES, OR 0 FOR NO. TRY AGAIN"
> 240 GOTO 190
> 250 PRINT "AFTER HOW MANY YEARS?"
> 260 INPUT X
> 270 LET R = L * ((1 + P/100)↑X - 1) / (V - 1)
> 280 LET 0 = L - R
> 290 PRINT "AFTER"; X; "YEARS"
> 300 PRINT "YOU WILL HAVE REPAYEO"; R; "POUNDS"
> 310 PRINT "AND"; 0; "POUNOS WILL REMAIN OUTSTANOING"
> 320 LET I = M * 12 * N - L
> 330 PRINT "AFTER"; N; "YEARS, YOU WILL HAVE PAYED"
> 340 PRINT "A TOTAL OF"; I; "POUNOS INTEREST"
> 350 LET F = (I + L)/L
> 360 PRINT "YOU WILL HAVE PAYED"; F; "TIMES THE ORIGINAL LOAN"
> 370 END
```

Lines 220, 230 and 240 are included in the program in case an 'impossible' answer is given ('5', for instance) instead of 0 or 1. Since neither of the conditions given in lines 200 and 210 are then

fulfilled, the computer will warn the operator and jump back to line 190 for another try.
Having entered the program, it can be tried out:

```
> RUN
ELEKTOR SOFTWARE SERVICE
LOAN REPAYMENT
WHAT STARTING CAPITAL IS AVAILABLE?
? 0
WHAT IS THE COST PRICE?
? 10000
THE NECESSARY LOAN IS 10000 POUNDS
WHAT IS THE INTEREST RATE?
? 9

IN HOW MANY YEARS MUST THE LOAN BE REPAYED?
? 30
THE MONTHLY PAYMENT IS 81 POUNDS
DO YOU WISH TO KNOW THE AMOUNTS ALREADY REPAYED
AND STILL OUTSTANDING AFTER A NUMBER OF YEARS?
PLEASE ENTER 1 FOR YES, OR 0 FOR NO
? 2

PLEASE STUDY THE KEYBOARD CAREFULLY, THEN
ENTER 1 FOR YES, OR 0 FOR NO. TRY AGAIN
? 1

AFTER HOW MANY YEARS?
? 10
AFTER 10 YEARS
YOU WILL HAVE REPAYED 1114 POUNDS
AND 8886 POUNDS WILL REMAIN OUTSTANDING
AFTER 30 YEARS, YOU WILL HAVE PAYED
A TOTAL OF 19160 POUNDS INTEREST
YOU WILL HAVE PAYED 2.916 TIMES THE ORIGINAL LOAN
```

BASIC (PART 3)

QUESTIONS

1. Is it permissible to enter more information in a 'data block' than is actually required in the program? And what about storing less data than required?
2. What use is the REM statement?
3. Can extensive use of the REM statement cause problems?
4. What effect will 'jump' statements have on the time it takes the computer to run the program (the 'execution time')?
5. When using the FOR . . . TO . . . STEP . . . statement, should the final value always be larger than the initial value?
6. What is the advantage of using subroutines?
7. When using FOR-NEXT and DO-UNTIL loops, why is it not permitted to 'jump out of the loop'?

ANSWERS TO QUESTIONS IN PART 2.

1. If an interpreter program were stored in RAM, it would be lost when the computer is switched off. It would then have to be re-entered before running even the shortest of programs. For this reason, it is normally stored in ROM.
2. The effect of the SCRATCH command is to erase the current program and the display.
3. The CLEAR command is used to reset the variables to 0. This command is often given before the RUN command; in fact, in some BASIC dialects it is effectively included in the RUN command.
4. The errors in the program lines are as follows:
   a) 150 LI ST 5: a space in the 'key word' (LIST) is forbidden.
   b) 1 0 PRINT 1B: a space within the line number is forbidden.
   c) 160 PRINT CHAIR: quotation marks should be included ("CHAIR").
   d) 170 PRINT 1253 14: a space within a number is not permitted.
   e) 190 LET A = 0.31: in most BASIC dialects that recognise decimal fractions, '.31' should be written instead of '0.31'.
   f) 200 PRINT 4.35E1.2: the number following the E must be a whole number — the decimal point is never permitted here.
5. a) $3 * 2 + 8 + 15/3 = 19$.
   b) $11 - 24/3/2 = 13$.
6. In BASIC, a variable should consist of one letter followed by not more than one digit, so 'A15' is not permitted.

Summary of statements and commands used in part 3.

INPUT *variable(s)*	This statement causes the computer to request keyboard entry of value(s) that must be assigned to the specified *variable(s)*.
READ *variable(s)* READ *variable(s)* . .	The *variable(s)* listed after the READ statement(s) are assigned the value(s)
DATA *data, data*...	given after the DATA statement.
RESTORE	This statement causes the data block to be re-used from the beginning.
REM *text*	The specified *text* appears in a listing, but has no effect on the program.
GOTO *line number*	This causes a jump to the specified *line number*.
BREAK	A key on the terminal that is used to stop the program.
IF *comp.* . . . THEN *line number* . . . THEN *statement* . . . GOTO *line number*	If the result of the *comparison* after IF is true, the computer 'jumps' to the specified *line number*; otherwise the program is continued on the next line in NIBL. A *statement* can be given instead of a *line number*; if a jump is required 'GOTO' must be used instead of 'THEN'.
FOR . . . TO . . . STEP . . . NEXT . . .	A 'running variable' is assigned an 'initial value', both as specified after FOR (e.g. FOR *A=1*). The statements between FOR and NEXT (the 'FOR-NEXT' block) are carried out; the running variable is increased by the specified step (e.g. STEP*15*), after which the FOR-NEXT block is repeated; and so on until the 'final value' specified after TO (e.g. TO *90*) is reached or exceeded. If no STEP is specified, the step is automatically taken as +1.
DO . . . UNTIL *comp.*	This 'loop' is known in NIBL. The statements between DO and UNTIL are repeated until the *comparison* specified after UNTIL becomes 'true'.
GOSUB *line number*	This causes a jump to the subroutine that starts at the specified *line number*.
RETURN	Last statement in a subroutine. It causes a jump back to the main program.

basic 2B elektor india

A very large low-frequency range is not necessarily a good thing. Often when a record is played at a reasonable volume the woofers occasionally 'strike' far further than the normal (acceptable) distance. This is usually due to undesirable distortion in the range of about 1 to 10 Hz. This phenomenon is limited only to record players; it is not noticed with compact disc players, radio tuners or tape players. The reason is that these last three have a high pass filter of about 20 Hz fitted somewhere in the electronics or in the signal source. As far as this circuit is concerned, therefore, we are only interested in the record player.

For most people a record player is still the best source of high-quality sound reproduction. (The CD — compact disc — system may be technically better but is far less popular.) A lot of attention is paid to the MM (moving magnet) and MC (moving coil) inputs in 'reasonable-quality' amplifiers as the best results are possible by making use of these inputs. A good amplifier must nowadays have a frequency range right down to (practically) d.c. values — the merit of which we will not discuss — but when combined with a turntable this can make life difficult for loudspeakers. In order to remove the problem, however, we must first of all know how it arises.

Resonance and warped records

Most subsonic problems are caused by
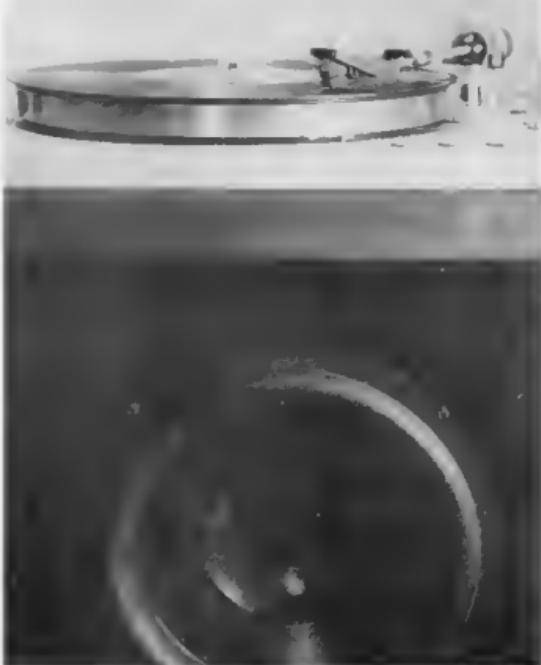
rumble detector

Manufacturers of audio equipment are constantly competing against each other by improving the quality of their products. As a result even inexpensive amplifiers now have an extremely good frequency range, especially at low frequency. A lower cut-off point of a few hertz is certainly no exception and some actually go right down to d.c. This is all to the good as far as the quality of sound reproduction is concerned but it is also disadvantageous to a certain extent. Subsonic noise can be passed on to the loudspeakers and could even damage them. Murphy's Law of Acoustics comes into play here, of course: as this noise is subsonic it cannot be heard so how do you know that it is there? That can be very difficult ... unless you build this rumble detector, which indicates, by means of a LED, when the subsonic part of an output signal becomes too great.

resonance in the turntable and pick-up arm. It all starts at the point where the arm pivots, its fulcrum in other words. The arm always pivots on some sort of elastic structure to allow the needle to move freely in the record grooves. Because there is a reaction between the mass of the arm and the elasticity of the mounting the assembly will have a resonant point. The frequency at which resonation occurs depends on the mass of the arm and the type of material on which it pivots. The

extent of the resonance depends mainly on the inherent damping of the fulcrum and any damping intentionally built in. When choosing a cartridge to fit to an arm, or vice versa, care must be taken to be sure that the resonant frequency of the combination is not in the audible range. By the same token the resonant point must not be so low that it is triggered by warps in the record or by a vibrating floor. A resonant frequency of about 10 Hz is generally considered to be optimal, although

traces ultra-low frequency signals
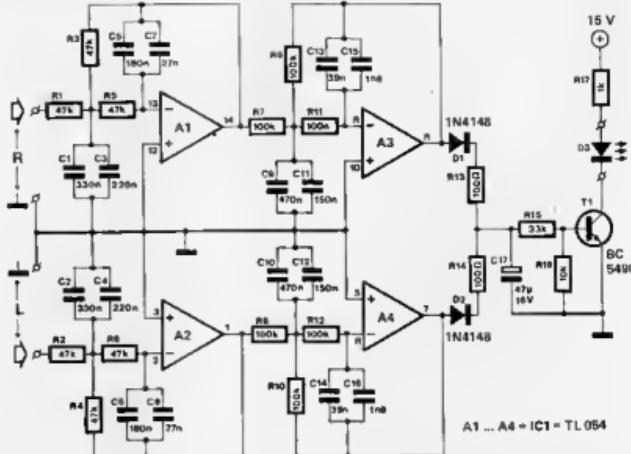
rumble detector

Figures 1. The filters in the rumble detector only pass frequencies below about 10 Hz. When this sort of subsonic noise is detected LED D1 lights to show that the signal is not very healthy for the woofers.

A1 ... A4 = IC1 = TL 054
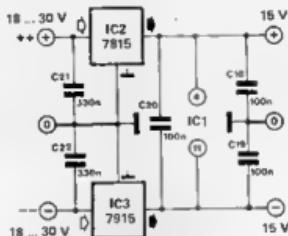
in practice a value between 5 and 15 Hz is acceptable. Unfortunately it is usually impossible to predict beforehand what the resonant frequency will be if a new arm or a new cartridge is bought. The only thing to do is to put your faith in an experienced audio shop. Most of us already have a certain combination of arm and cartridge and unless this gives terrible results we are not likely to change it. The next problem is that of resonance in the chassis. Like the arm, the chassis will probably be mounted on springs or some other elastic assembly. Understandably, this also has a resonant point. In a well-designed turntable the resonant frequency will be about 2 to 4 Hz so that it cannot influence the resonance of the arm. The turntable is then relatively immune to sounds in the room (such as somebody walking around) and the arm can cope with warped records. If the chassis is not mounted on an elastic assembly there is one less resonant point to worry about, but this sort of turntable is more sensitive to outside interference, like those same heavy footsteps and feedback from the loudspeakers.

If a record that is being played has a bump in it or is warped there is quite a

likelihood that the pick-up arm will start to resonate. The result is a subsonic 'spike' that will be strengthened by the amplifier and fed through to the loudspeakers. Even if the resonant point is well damped this will still happen. When somebody forgets to tread softly their heavy footfalls can cause the chassis to resonate, with the same result. The loudspeaker cones again thump out a note that nobody hears. Clearly subsonic noise can very easily be produced by a turntable but it is not so easy to get rid of it. A good steep subsonic filter with a cut-off frequency of 18 to 20 Hz is ideal for this but such a filter is rarely included in an amplifier. If a filter is provided it is usually a rather imprecise affair that also removes some of the low frequency sounds which should be heard. That does not seem to leave any options open, but the situation is not by any means hopeless. Provided the loudspeakers do not deflect too much the subsonic signals themselves will cause no damage. That idea was the inspiration for this circuit, which monitors the subsonic content of the signal at an amplifier's loudspeaker outputs and lights a LED to signal when it becomes too great. The listener can then take appropriate measures. This may involve reducing the volume setting, examining the record for bumps or warps or mounting the turntable more firmly, for example. Whatever measures are needed the rumble detector will not effect the quality of the hi-fi system as it does not involve connecting anything in the signal line.

The circuit

The circuit has a very simple layout, consisting of only a low-pass filter for each channel (left and right) followed by a common LED for the indication. The filter roll-
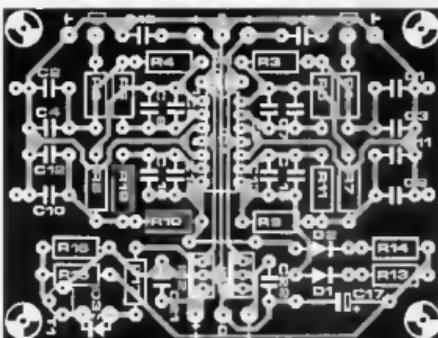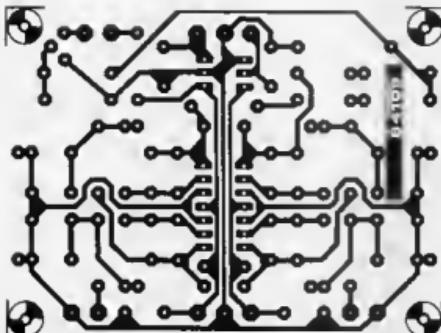
2

Figure 2. Voltage
regulators are among the
components found on the
printed circuit board. If a
+ and −15 V supply can
be tapped off the ampli-
fier these two stabilisers
are not needed.

off characteristics must be very steep to
prevent the circuit from reacting to bass
frequencies that are acceptable for the
loudspeakers. This is the reason for using
24 dB/octave fourth-order Butterworth
filters. (For detailed information about this
and other types of filters refer to the
article 'active crossover filter' published in
the October 1984 issue of Elektor.) The
actual multiple-feedback arrangement
used for the filters makes them very stable
and that is more than enough justification
for their relative complexity. Each filter
consists of two 12 dB/octave sections —
A1/A3 and A2/A4. The component values
chosen set the cut-off frequency to about
10 Hz so signals up to about 12 Hz can be
detected. The output signals from the
filters are half-wave rectified and then
added together. The resultant signal
charges capacitor C17 quickly. When the
voltage on this capacitor reaches about
2.5 V transistor T1 conducts and causes
the LED to light. After the subsonic signal
has passed the LED remains lit for a cer-
tain time, dependent upon the d.c. voltage
across C17.

Power for the circuit is provided via two
voltage regulators, IC2 and IC3. The maxi-
mum input voltage to this power supply
section is 30 V. This can be supplied by a
small transformer (2 x 15 V/50 mA, for
example) in combination with a bridge
rectifier and two electrolytic capacitors
(such as 470 μ/25 V). If the amplifier has a
suitable symmetrical 15 V from which a
tap-off can be taken the components
marked with an asterisk in the parts list
can be left out. The current consumption
is a mere 20 mA so this will not prove an
excessive burden to an amplifier.
The last thing to be done is to connect
the circuit in parallel with the amplifier's
loudspeaker outputs and the detector is
ready for use.

Filter component values:

Butterworth 4th order
(24 dB/octave)

C5 + C7 = C6 + C8 =
1.85/6 x fR

C1 + C3 = C2 + C4 =
3/3.7 x fR

where R = R1 = R2 =
R3 = R4 = R5 = R6

C13 + C15 = C14 +
C16 = 0.77/6 x fR

C9 + C11 = C10 + C12 =
3/1.54 x fR

where R = R7 = R8 =
R9 = R10 = R11 = R12

(All values are given in Ω,
Hz and F)

microprocessor-controlled frequency counter

accurate, fast, user-friendly and practical

Superlatives simply have to be used when describing this frequency counter, but we will try not to let it get out of hand. We think it is only natural to feel a certain amount of pride, however. Here, at last, is a do-it-yourself frequency meter whose capabilities, features and ease of use are comparable to much more expensive ready-made (professional) equipment. This is a tribute to the Elektor designers who continually burned the midnight oil working on the design. Their advice to anybody who is thinking of building or buying a new frequency counter is to read this article and study the circuit diagram first. The chances are that this will be your next frequency meter.

Technical specifications:
- Frequency range:
  0.01 Hz ... 1.2 GHz
- Period time measurement. 10 ns ... 100 s
- Pulse time measurement: 0.1 μs ... 100 s
- Events counter 0 ... 10⁹ pulses
- User-selectable accuracy to 6 or 7 digits
- Measuring time for frequency measurements:
  <0.2 s (6 digits) at f>20 Hz
  <2 s (7 digits) at f>2.5 Hz
- Auto-ranging in all ranges
- 16-digit alphanumeric display with interactive operation
- Sensitivity:
  input A. 10 mV/rms (2 MΩ)
  input B. TTL, CMOS level (R≥5 kΩ)
  input C. 10 mV/rms (50 Ω)
  with prescaler
  (f>100 MHz)
  100 mV/rms (50 Ω)

Many of our readers regard microprocessors as a scourge and something that has no place in a respectable 'electronic' circuit. Even having read the title on this page we hope that you have none the less read this far because we are now going to entreat you to bear with us and read the remainder of this article. This frequency counter does contain a microprocessor but for all intents and purposes it can be considered as just another 'black box'. Its use does, however, allow the design to be made more versatile and totally removes the need for a user's manual. Surely this is enough reason to turn a blind eye to the fact that this test instrument is 'cursed' with a microprocessor.

Let us look at the frequency counter's features one by one, starting with the auto-ranging possibility. In most meters this would have required a handful of components but in this design the 'calculator' does all the work.

The second point is the measuring method used, which was mentioned in the introductory article published in last month's issue. This so-called reciprocal measuring is only possible with the aid of a microprocessor as there are a lot of calculations to be made. The advantages of this method are maximum accuracy with a very short measuring time. It is

used both for frequency and period measurements. A secondary benefit is that multi-position rotary switches are unnecessary as simple push buttons are sufficient.

If the circuit has a microprocessor anyway why not use an alphanumeric display? This, again, is easier to use. The frequency meter plays a game of question and answer with the user, who then makes a choice of the function desired by pressing a YES or a NO button. The display always shows what is being measured and what the units are: such as 'FREQ. 1.234567 KHZ' or 'PER. 8.69089 MSEC'. The price of the alphanumeric display and the associated controller IC is comparable to that of a set of good 7-segment displays plus drivers so cost does not enter into the equation. Extensions and modifications can easily be included in this frequency counter; usually this is a matter of changing part of the software. This line of thought could lead to such things as an IEEE output or feeding in offset frequencies.

Finally, we draw your attention to the front panel foil with built-in low-pressure membrane switches. This gives the project a very professional appearance and simplifies fitting it into the case.

The measuring principle

If the frequency counter electronics are considered in the form of blocks a first rough sub-division leaves us with the three parts shown in figure 1: the microprocessor, the counter hardware and the display controller. The microprocessor section consists of a standard 6502 system with RAM (6116), EPROM (2732) and PIA (6821). The microprocessor works with the program stored in the EPROM and uses the RAM memory as a notepad and to store data. Communication with the other hardware is via the PIA (peripheral interface adapter). The display controller will be described in next month's issue so we will not deal with it now. The counter section makes up most of the circuit so we will look at that in detail, using block diagrams and timing charts, to give a better understanding of the operation of the whole circuit.

All the hardware is controlled by the microprocessor, with the aid of a pair of multiplexers to select the various measuring procedures. Each of the possibilities will be discussed in detail on the basis of a timing chart and a block diagram containing only the components appropriate to a particular function. The multiplexers are considered as straight-through links. First, however, we must see how all the measurements are made.

A lot of calculation

The article in the last month's issue (called 'part 0' because it was only an introduction, intended to whet your appetite) skimmed over the frequency counter's *modus operandi*. The microprocessor starts by setting the programmable divider to its lowest value. This means that the divider stops the measurement after a single period of the input signal. This is a test to prepare for the actual measurement so that the right division factor can be selected. The period time is worked out from the test and the processor then calculates which multiple of this period

can be stored in the counter within the desired measuring time. Based on this the programmable divider factor, which is always a power of 2, is set. The measuring time is then defined by: $2^n \times T$, where T is the period time measured and n is the number selected for the divider. The number of period times that are counted is always a power of 2 so the measuring time for different frequencies can also vary by a factor of 2. As an example, assume an accuracy of 6 figures is selected. The gate time (measuring time) for this accuracy is at least 0.1 s. If the frequency to be measured is 5 kHz its period time will be $T = 200 \mu s$. After the test measurement the microprocessor quickly makes the following calculations:

$2^n \times 200 \times 10^{-6} \geq 0.1$, so $2^n \geq 500$, giving $n = 9$ and $2^n = 512$.

The last number is the factor for the programmable divider. The gate time is then:
$512 \times 200 \times 10^{-6} = 0.1024$ s.

If the frequency were 2.6 kHz the period would be: $T = 385 \mu s$.
$2^n \times 385 \times 10^{-6} \geq 0.1$, whereby $2^n \geq 260$, which gives $n = 9$ and $2^n = 512$.

The same division factor is selected but the gate time is now:
$512 \times 385 \times 10^{-6} = 0.197$ s.

At 2.5 kHz the gate time does become less as the division factor is then $2^8 = 256$. The measuring time is given by:
$2^8 \times 400 \times 10^{-6} = 0.1024$ s.

When the 'real' measurement has been made the contents of the counter (X) is read out by the processor. The result is the gate time multiplied by 10^7 (the reference frequency of 10 MHz). The gate time is $2^n \times T$, so $X = 10^7 \times 2^n \times T$. Taking the example of f = 2.6 kHz again we see that $T = 1/2600$ s. This gives
$X = 10^7 \times 512 \times 1/2600 = 1969230$. Based on this figure and the division factor the microprocessor can calculate the frequency or period time, whichever the user has chosen.
$f = (10^7 \times 2^n)/X = 2.60000$ kHz

1

84087-1
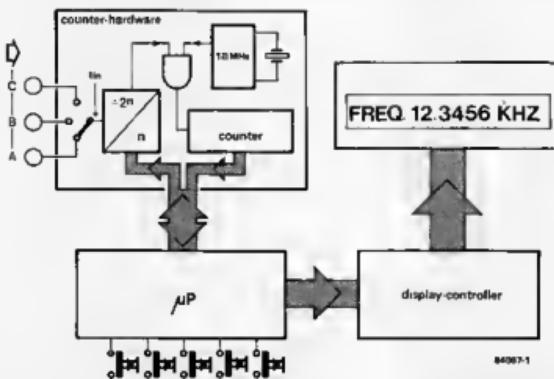
Figure 1. The block diagram for the frequency counter in vastly simplified form. The microprocessor controls virtually all the rest of the electronics.

microprocessor-controlled
frequency counter
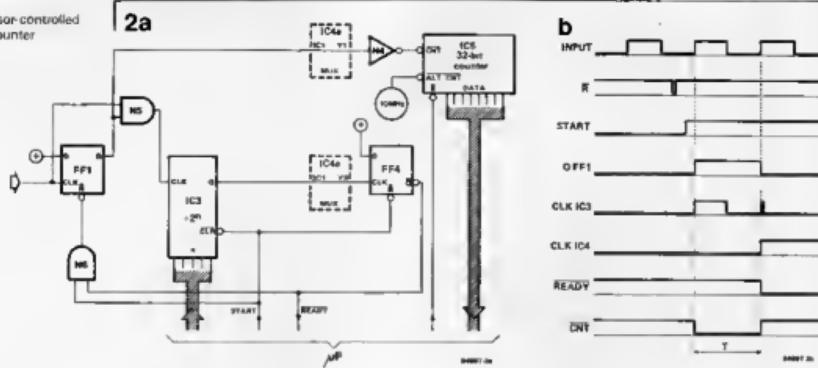
Figure 2. These are the components that play a part in test measurements for the frequency or period functions. The timing chart shows what happens to the relevant signals.

$$T = X/(10^7 \times 2^0) = 384.615 \ \mu s$$

That, basically is how the frequency and period time are calculated. We will now look at the electronics needed to achieve this.

Frequency and period measurements

The set-up used for the frequency and period 'test' measurements is shown in figure 2a. The output signal of FF1 is fed to N4 and the Q output of IC3 is connected to FF4's clock input (via a multiplexer in each case). The sequence of events is shown in the timing chart (figure 2a). First the 32 bit counter is reset and the hardware is enabled by means of the START signal. The lowest possible division factor, 2^1, is selected for IC3. The next rising edge of the input signal makes FF1's Q output high so N5 can then feed the input signal to IC3. At the same time the $\overline{CNT}$ input of IC5 goes low so it starts counting 10 MHz pulses. The second rising edge of the input signal causes the Q output of IC3 to go high. (This output becomes '1' after the 2^{n-1}th rising edge if a division factor of 2^n is selected.) The

Figure 3. This arrangement is used for actual frequency or period measurements.

READY line then becomes zero and resets FF1. This flip-flop's Q output goes low with the result that IC5 stops counting. Even though IC3's division factor was set to 4 the output becomes high after a single input-signal period. The contents of IC5 is the result of counting 10 MHz pulses for one period and this figure is used to decide what division factor has to be set for the actual measurement.
For the actual measurement FF3's Q output is connected to FF4's clock input (see figure 3a). The microprocessor then sets IC3 to the division factor that it has selected. The first rising edge after the START signal takes FF1's Q output high so N5 can then pass the input signal on to IC3. After 2^{n-1} rising edges the output of IC3 becomes '1'; this signal clocks FF3 and only then can IC5 start counting 10 MHz pulses. The next clock pulse for FF3 comes after exactly 2^n periods and FF3 then flips and stops counter IC5. At the same moment the READY line is taken low by FF4 to inform the processor that the measurement is completed. The counter's contents and IC3's division factor are then used to calculate the frequency or period. When

2.44 elektor india february 1985

4a

b

this is done the result is passed to the display and the circuit is then ready to start the next measurement.

Pulse time

Pulse time is measured on the basis of the circuit seen in figure 4. The signal is input via an EXOR gate (N1) to enable the microprocessor to define whether the '1' or the '0' time is measured. The timing chart of figure 4b assumes that N1 does not invert the signal so the '1' time is measured. Once the hardware is STARTed the Q output of FF1 goes high with the input signal's first rising edge. The 32-bit counter is triggered via N5 and stops again at the input's falling edge. At the falling edge a clock pulse is fed to FF4 via FF2 and N2. The READY line (which goes low) then informs the processor that the measurement is complete and FF1 is also prevented from reacting to any more input signals. The contents of the counter is then equal to the pulse time in tenths of microseconds (as N1 pulses were counted). This makes further calculations unnecessary.

The width of the pulses that are counted ($1/10^7 = 0.1\ \mu s$) means that the resolution

for pulse-time measurements is also $0.1\ \mu s$ and the result displayed takes this into account. A number shown in microseconds on the display will never have more than one figure after the decimal point. The shorter the time measured the fewer figures will appear on the display.

Counting pulses

This is the easiest function so its block diagram (figure 5a) is also the most straightforward. Again N1 enables the microprocessor to choose whether the counter reacts to a rising or falling edge at the input. The input signal is then used to clock counter IC5 directly (so the 10 MHz reference frequency is no longer connected to the counter's ALTCNT input). The microprocessor regularly examines the contents of the counter and outputs the result to the display.

The actual components used for each of the frequency counter's functions were given in each of the block diagrams so they can be more or less ignored in the actual circuit diagram. This is a distinct advantage as the circuit's complexity makes it difficult to see directly which parts relate to which function.

5a
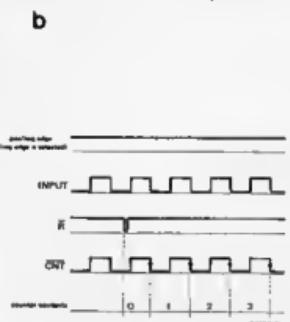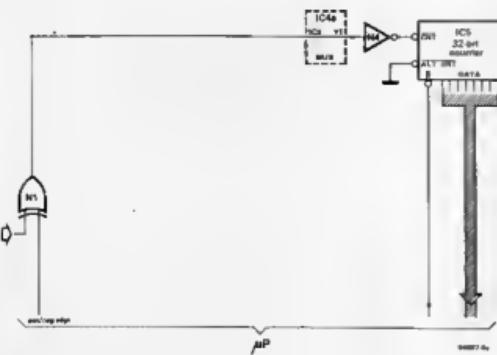
The frequency counter circuit

The circuit diagram is divided into two parts: the processor section (figure 6a) and the counter hardware section (figure 6b). The power supply has been included in the drawing of figure 6a.

There is little need to say anything about the 6502, 2732 and 6116 in the processor system, especially as they are being treated as 'black boxes' in this circuit. The address decoder, IC8, and PIA, IC7, also belong to the processor section but they have been drawn in figure 6b because of the large number of connections between these ICs and the hardware section.

A break-down of the addressing ranges is given in the margin here as we must say

something of the address decoding. The decoding method used is very wasteful of memory space but its great advantage is its simplicity as it requires only a single IC. Furthermore it works very well in this application, even leaving a block of 8 K free. The clock signal for the 6502 is taken from the 10 MHz reference via decade divider IC18. The circuit can cater for either a 2732 or 2764 EPROM (IC16) so there is room for software expansion. The system is reset on power up by means of C8, R50 and N13.

Many of the components found in figure 6b are already known from the block diagrams. The display and its controller

Memory map

Figure 6a. The processor section of the circuit comprises the microprocessor itself and two memory chips, a ROM (EPROM, actually) and a RAM. The power supply is also shown in this diagram.

6a
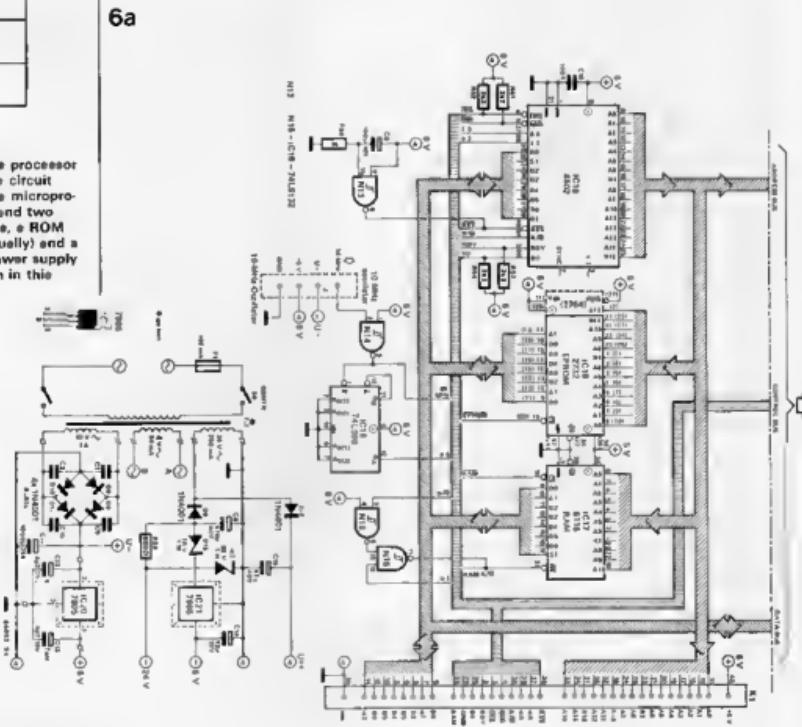
Table 1. These are the connection details for the three connectors in the frequency meter circuit.

6b

Figure 6b. The remainder of this circuit consists mainly of the counter hardware. Some of the components do not actually belong here: IC7 and IC8 are part of the processor section and the display with its sub-section is really a separate sub-section all of its own.

84287-6b

are also shown in this section but we will not deal with them until next month. All the 'boxed-in' parts of this diagram are found on the display printed circuit board. A number of points about this main circuit diagram must now be clarified.

At the input is the pair of four-bit multiplexers found in IC1. One of these multiplexers is used for choosing an input A, B or C/16. If a prescaler is included in the circuit (this will be published next month with the input amplifier) the C/512 input can also be used. The prescaler is designed to handle the frequency range from 100 MHz up to 1.2 GHz. If link PR is fitted the circuit 'knows' that the prescaler is not fitted and the menu published last month applies. If, on the other hand, the prescaler is included points P and R must be linked with a wire bridge. An extra choice is then included in the menu. Choosing FREQUENCY and C-INPUT would normally be followed by the question 6 DIG. PRECISION? Instead the query displayed is: FREQ.<100 MHZ? If the reply is NO the counter then asks FREQ.>100 MHZ? Depending on the answer to this question the input multiplexer chooses input C/16 or C/512. The other section of IC1 drives LEDs D14...D16 via N9 and N10.

The TRIGGER LED, D1, is driven by monostables MMV1 and MMV2. This LED always flashes if there are active edges at the input. If the input signal frequency is low, 2 Hz, for example, D1 flashes with each active edge. At higher frequencies the LED simply flashes at a constant easily visible rate but there is then no distinction made between a frequency of 100 Hz and one of 10 MHz. This does not matter, of course, as the only intention of the LED is

to show the user that the counter is being triggered.

The SCRL and CNTRES signals (scan reset/load and counter reset, both of which come from the address decoder) are synchronized to ø0 by means of FF5 and FF6 to remove any spikes that might be present. This brings us to the heart of the circuit, the LS7060, whose internal layout is shown in figure 7. This is a dedicated counter IC containing a 32-bit binary counter complete with latch, multiplexer and three-state data outputs, all of which is microprocessor-bus compatible. To achieve the same result in TTL would require about 15 ICs so IC7's price is not as exorbitant as it might seem. The contents of the 32-bit counter is stored in the latches as soon as the SCAN RESET/LOAD input goes from '0' to '1'. At the same time the scan counter is reset and the left-most multiplexer feeds the eight least significant bits to the data outputs. If an ENABLE pulse is now fed to the IC (CNTR goes low) these eight bits are placed on the data bus by the output drivers so that they can be read by the microprocessor. Should the E line return high the contents of the scan counter is increased so that the next eight bits are sent to the output drivers. The next E pulse places these eight bits on the data bus so after four enable (CNTR) pulses the processor has read all 32 bits. A logic zero of at least 1 µs on pin 37 of the IC RESETs the 32-bit counter. This is taken care of by the CNTRES line in the circuit. The microprocessor generates the signal by simply placing the address for the CNTRES block momentarily on the address bus. In this way output '4' of address decoder IC8 is activated and this

Figure 7. The LS7060 is a dedicated counter IC, which means that it is quite expensive, but not in terms of what it provides.

7

LS7060
BLOCK DIAGRAM

84067-7
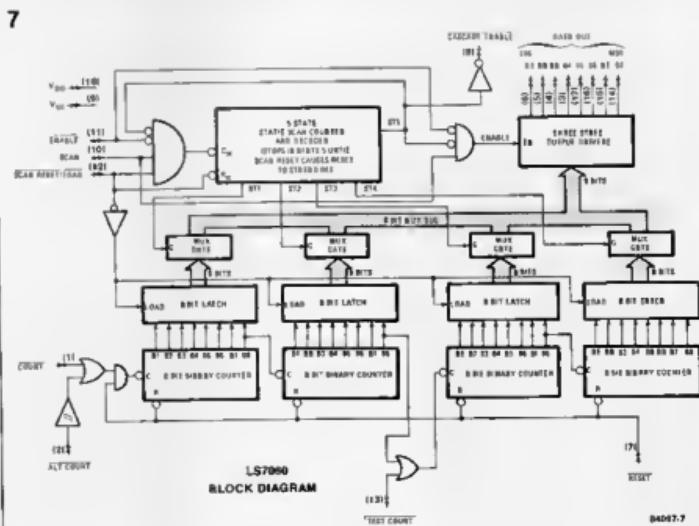
signal is passed via FF6 to IC5's $\overline{R}$ input. The LS7060 IC also has two clock inputs, COUNT and ALT COUNT. One of these is used as the clock input while the other serves as the enable input. We will not deal with the remaining pins of IC7 here as the TEST COUNT, SCAN and CASCADE ENABLE are not used in our frequency counter circuit.

The gate LED, D2, indicates the time during which a measurement is made, and will light if there is a '1' on line PB5 or on output Y1 of IC4. We have already seen from the block diagrams that the Y1 output defines the 32-bit counter's measuring time for frequency and period measurements. When the event counter mode is selected the input is connected to Y1 and IC5 counts the incoming pulses. The circuit is then measuring continuously, even if a lot of the time is spent waiting for a pulse, so the gate LED should light continuously. For this reason the processor places a '1' on the PB5 line. For pulse width measurements either the '1' or the '0' part of the signal is measured. A long pulse time (0.5 s, for example) can easily be seen on the LED but this is not so if the measured time is short (a repetition frequency of 1 kHz and a pulse of 200 µs, for instance). A delay built into the software takes care of the problem. After taking a measurement the processor waits for about 200 ms before starting to measure again. The LED is driven by PB5 for a few milliseconds so that it flashes visibly. The software delay also works on the display, and in this way the last digit does not keep changing from one value to another (if, for instance, the pulse time is between 200.0 µs and 200.1 µs).

The gate LED makes one short and one long flash during frequency and period measurements of signals below about 100...200 Hz. The short flash is the test, while the actual measurement is made during the long flash. The LED only flashes once when very low frequency signals are being measured as the maximum measuring time is then shorter than the signal's period. In this case the 'test' is the actual measurement.

Finally we have STPLED, D5, which indicates that the display is frozen and that no more measurements are being carried out. This LED is linked to output '5' of the address decoder and lights if the relevant memory block is addressed. The processor must then run through a loop in which the LED is constantly addressed. This is the only thing the processor has to do in the HOLD state. The display controller (IC6) ensures that the read-out remains fixed.

The display section is a completely independent part of the circuit. The ASCII data that must be displayed is sent to the controller IC by the processor. This information is then stored by IC6, which also takes care of driving the display. The processor only intervenes by transmitting new data if the displayed information has to be changed.
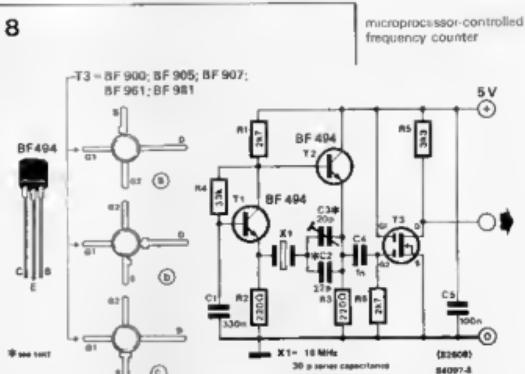
The power supply section seen in figure 6a shows that several different voltages are needed. A heating voltage of 6 V a.c. is fed to the fluorescent display from one of Tr1's windings. The same transformer provides a separate −24 V d.c. that is also used for the display. The remainder of the circuit is powered by the regulated 5 V line. Two more supply voltages (U++ and −5 V) are needed for the input stage. The diagram shows a single mains transformer with a total of three secondary windings but the same effect can be achieved with two or with three separate transformers.

The crystal oscillator

The crystal oscillator is a very important part of the frequency counter as it determines the accuracy to a large extent. The design used is shown in figure 8. The crystal is in series resonance as this guarantees quite good stability. Both sides of the crystal are terminated into a low impedance in order to prevent its Q factor from being unnecessarily reduced. The MOSFET at the output acts as a buffer between oscillator and frequency meter. This oscillator circuit has been fitted to a separate small printed circuit board, and there is a good reason for this.

The oscillator shown is quite acceptable but a very precise, temperature-stable crystal oscillator is needed to achieve an accuracy of six or seven digits. A cautious estimate gives the oscillator of figure 8 a stability of at most 10 ppm (parts per million) in a temperature range of 15 to 35 °C. This means that the accuracy is far from optimal but it is good enough for the average hobbyist especially as the temperature within the case will settle down to a constant value within quite a short time. Some people will, of course, demand that measurements be accurate to six or seven digits. This can be achieved by substituting a special temperature-compensated oscillator in place of the one shown here. Such an oscillator will be expensive but it can easily be incorporated into this frequency counter.

Figure 8. The circuit for the crystal oscillator is straightforward. Two transistors, T1 and T2, look after the low-impedance terminations for the crystal and a third one, T3, acts as a buffer

A self-testing circuit

The frequency meter is made up of three
printed circuit boards:
— the main board, which is double sided
  with through-plated holes
— the display board, and
— the crystal oscillator.

The first to be built is the crystal oscillator
(if it is used). The series capacitance
indicated in figure 8 is needed if the
crystal has a capacitance of 30 pF. The
capacitor and trimmer will have to be
changed if different crystals are used.
(Keep the trimmer as small as possible to
prevent deterioration caused by its poor
temperature stability). The circuit is con-
nected to a regulated supply and tested to
see if the oscillator is supplying a 10 MHz
signal. The output is probably distorted
but this does not matter as it is caused by
the capacitive load of the probe. A 10 MHz
oscilloscope is needed during construc-
tion of the frequency counter so this is
something you will have to get your hands
on.

The next part to be tackled is the main
printed circuit board, starting with the
discrete components. Some of the
capacitors and resistors (and one diode)
must be mounted vertically. Sockets
(preferably low-profile types) should be
fitted for all ICs, which must not be fit-
ted yet. The eight lines indicated as K3
can be fitted with soldering pins. One of
the voltage regulators, IC21, is connected
to the board by means of three thick flex-
ible leads about 10 or 11 inches long
(roughly 25 cm). A pair of capacitors, C12
and C13, solder directly to the IC's pins.
This regulator is not fitted with a heat sink
but rather mounts straight onto the back
panel of the Verobox. The heat generated
is conducted through the metal plate and
to aid this process silicone grease should
be applied between voltage regulator and
back panel. The photographs show where
and how this IC must be fitted.
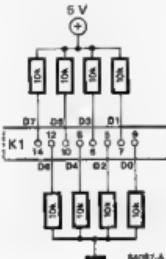
The mains transformer possibilities have
already been mentioned but one point to
note is that the case recommended does
not leave a lot of room to play with. The
fewer and the smaller the transformers the
better they can be fitted into the available
space. The 10 V winding is first soldered
to the board and the output of IC20 is
measured. This should be 5 V. The 24 V
winding is now connected and the
appropriate voltages measured: −5 V at
pin 7 of K3 and −24 V at pin 19 of K2.
The oscillator can now be mounted
on the main board. Be particularly careful
to connect the U~ connection of the 5 V
copper track and the ground connection
to the oscillator board. The U~ connec-
tion is not used by the Elektor oscillator. It
provides an unregulated 9…12 V d.c. for
crystal oscillators that need more than the
+5 V we have used.

Two of the ICs, IC18 and IC19, are now fit-
ted to their sockets. A square wave of
1 MHz should be present at pin 12 of IC18.
Hexadecimal number $AA must then be
set on the data bus by means of eight 10 k
resistors as shown in the margin here.

The resistors are temporarily soldered to
connector K1 and the 6502 can then be
inserted into the socket. When the power
is switched on again there should be a
square wave of 250 kHz on address line
A0, 125 kHz on A1, 62.5 kHz on A2, and so
on down to 7.6 Hz on A15. This is easily
checked with the oscilloscope as the
period doubles each time. All address
lines are available at connector K1. If the
frequencies are not correct or if they are
simply not present check pin 40 of IC16
(RES), which should be '1'. Similarly NMI
and IRQ should be '1'. On pin 39 (ø2) there
should be a (rounded) 1 MHz square wave.
If there are signals on the address bus,
but not the correct ones, it is possible that
the 10 k resistors are not properly con-
nected and that something other than $AA
(10101010) is present on the data bus. A
further, ever present, possibility is that an
address line may be shorted to another
line.

The resistors are left where they are and
IC8 and IC14 are fitted. Outputs '0'…'7' of
IC8 should then in turn become logic
zero for 16.4 ms. Pin 9 of IC14 must
become '0' at the same time as pin 4 of
IC8 and the same applies for pin 5 of IC14
and pin 5 of IC8. If this checks out the
10 k resistors can be removed.

Now we move on to the display section
but first the heating voltage for the fila-
ment must be measured. Connect 330 Ω
across the transformer's 5 V winding and
measure the a.c. voltage across the
resistor. It will usually be a little higher
than 6 V. The series resistance needed to
give a voltage drop of 5.8 V across the
330 Ω resistor can be calculated from the
formula: R = u × 57 − 330, where u is the
voltage measured. Divide the result by
two and use the nearest E12 or E24 value
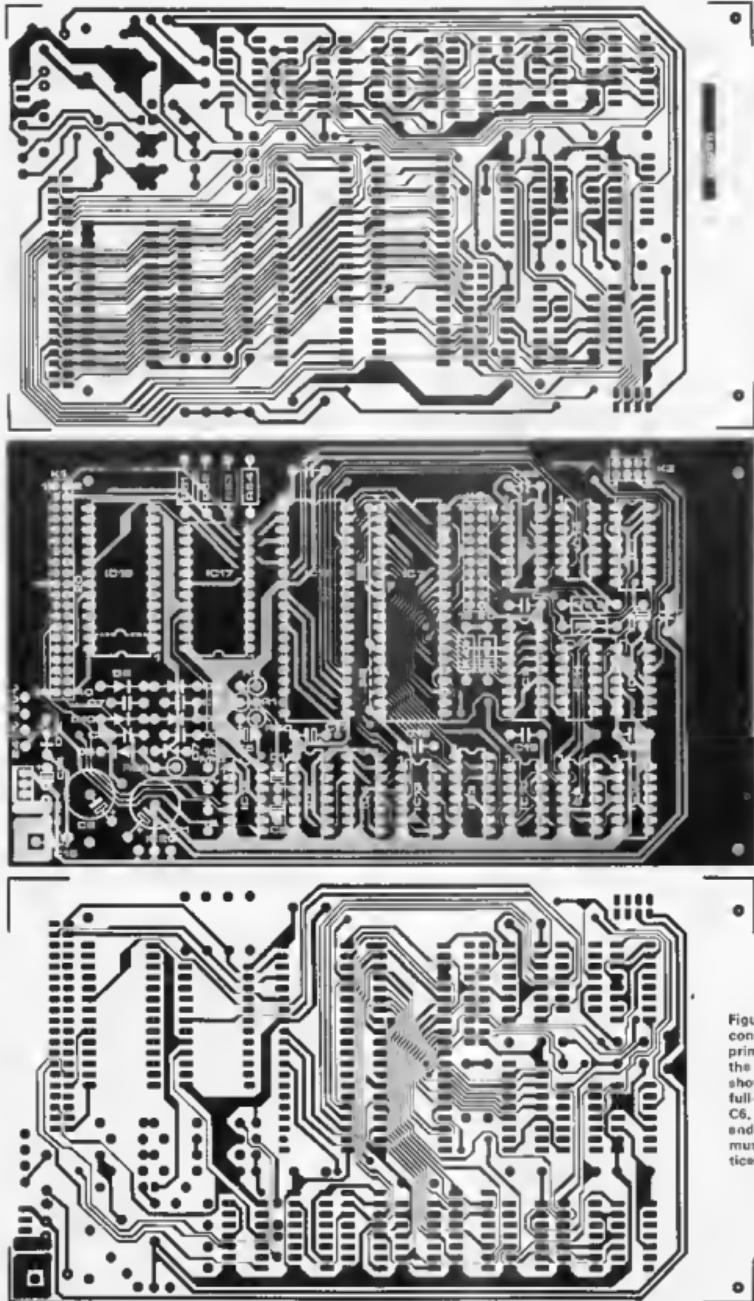
Figure 9 Due to space
considerations the main
printed circuit board for
the frequency counter is
shown here at 70% of
full-size. Note that C3, C5,
C6, C11, C14, C15, D13
and some of the resistors
must be mounted ver-
tically.

for R9 and R42 on the display board. The display board can now be assembled except that the display itself and IC6 should not yet be fitted. Do not use a socket for IC6. Note that the tops of the LEDs should extend 12 mm above the board and that transistor T2 is laid flat on top of the board. Use as thin a capacitor as possible for C4 or fit it on the reverse side of the printed circuit board. The
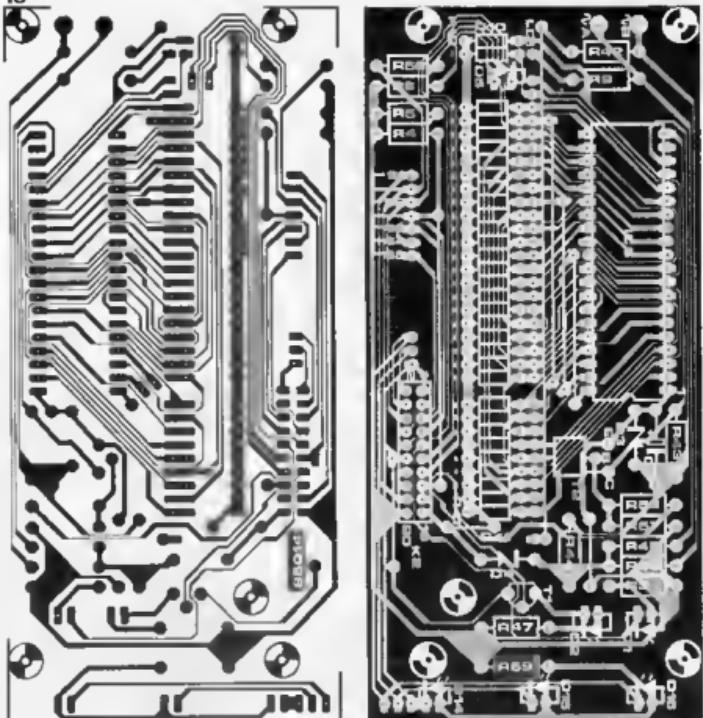
Figure 10. The display section is fitted on this printed circuit board. The 16-digit alphanumeric display is soldered straight onto the board and only bent back after testing is complete.

Parts list

— Main board and display board

Resistors:
R1, R2 = 100 k
R3, R47, R59 = 390 Ω
R4, R5, R51 . . . R54,
R57 = 2k2
R6 = 5k6
R7, R48 = 220 Ω
R9, R42 = *
R10 . . . R41, R55 = 100 k, 1/8 W
R43 = 22 k
R44, R46, R58 = 1 k
R48, R49, R56, R58 = 10 k
R55 = 680 Ω

Capacitors:
C1, C2 = 0.47 μ/10 V Ta
C3, C14 = 47 μ/10 V
C4 = 10 μ/25 V
C5 = 100 μ/10 V
C6 = 470 μ/40 V
C7 . . . C10 = 47 n
C11 = 1000 μ/25 V
C12, C13 = 4.7 μ/25 V Ta
C15 = 47 μ/40 V
C16, C18 . . . C23 = 100 n
C17 = 10 μ/10 V Ta

Semiconductors
D1 = LED, 5 mm yellow
D2, D5, D14 . . . D16 = LED, 5 mm red
D3 = 15 V/400 mW zener
D4 = 5V6/400 mW zener
D6, D8 . . . D12 = 1N4001
D7 = 24 V/1 W zener
D13 = 10 V/1 W zener
T1 = BC547
T2 = BC557
IC1, IC4 = 74LS153
IC2 = 74LS221
IC3 = 74LS292
IC5 = LS7060 (LSI)
IC6 = 10937 SY (Rockwell — available from Regsbrook)
IC7 = 6821
IC8 = 74LS42
IC9 = 74LS86
IC10 = 74LS08
IC11 = 74LS132
IC12 . . . IC14 = 74LS74
IC15 = 6502
IC16 = 2732
IC17 = 6116
IC18 = 74LS90
IC19 = 74LS132
IC20 = 7805
IC21 = 7905

Miscellaneous
F1 = fuse, 100 mA slow blow
LD1 = Futaba 16-digit alphanumeric fluorescent display, type number 16-SY-03(2) (available from Regsbrook)
S1 . . . S5 = membrane switch in front panel foil
S6 = miniature double-pole mains switch
TR1* = mains transformer 6 V/≥50 mA, 10 12 V/≥800 mA) and 24 V/≥150 mA
heatsink for IC21
connector for cable from front panel foil (7 way right angled, 2.54 mm/0.1" centres)
Verobox no 075-01411D (dimensions 205 × 140 × 75 mm)
front panel no. 84097 F

* = see text

cable from the membrane switches in the front panel chips into the connector between R4 and K2. A length of 20-way ribbon cable about 7 inches (17 cm) long is soldered directly into the holes for connector K2 on the reverse side of the display board. The other end of the cable is then soldered directly to the main board (do not use soldering pins at either end). The numbers linked by each wire in the cable must, of course, be the same on both boards.

With the power switched on the voltages on the display board can be checked. There should be +5 V at pin 1 of IC6 and −10 V on pin 18. The d.c. voltage on the common line for R9…R41 should be −24 V. Switch the power off and solder IC6 directly onto the board. The pins of the display are pushed through the board until they extend by about 1 mm and are then soldered. Do not bend the display parallel with the board yet; until testing is complete it is left standing straight out like this. Returning to the main board IC16, complete with its program, is inserted into its socket. Switch the power on again and then touch the NMI line (pin 4 of K1) to ground for an instant. The self-test program in the EPROM then starts.

The first function of this program is to test if it is itself correctly stored in IC6. If it is stop LED D5 lights for about 1 s to show that all is in order. If the LED remains on the EPROM is probably incorrectly programmed or one of its pins is not making proper contact. This assumes, of course, that the LED is fitted properly and that the wiring between the two boards is also correct.

After D5 has lit up the power can be switched off to enable IC7 to be fitted into its socket. The power is switched on again and the '0' pulse is again applied to the NMI line. Each of connections PA0…PA7 goes high for 1 s in turn so LEDs D3, D14, D15 and D16 start flashing (on for 1 s and off for 8 s). There is also a '1' running through lines PB0…PB6 (except for PB4). Provided the '1's keep running and the LEDs flashing the circuit is correct so far. Otherwise check IC7's socket to ensure that all the pins of the chip are inserted correctly. Next it is IC17's turn to be fitted. Substitute a wire bridge for S3 on the display board so that the circuit thinks that this push button is pressed continuously. Power is again applied to the circuit and the NMI line momentarily grounded. The gate LED should light for a short time. If this does not happen there is something wrong in (or with) the RAM. The gate LED may also fail to turn off and the stop LED may light and if this happens it indicates a problem, probably a programming error, in the EPROM itself.

It may be an advantage at this stage to see exactly what the test program does. If switch S3 is closed (or bridged) and NMI receives a short '0' pulse the program tests itself and lights the stop LED for about one second. The microprocessor then examines PB7. If this line is '1' the

PIA is then tested but if it is '0' (as it is when S3 is bridged) the RAM test procedure is started. This involves copying the contents of the EPROM to the RAM and then comparing what is stored in both. Assuming that this test gives a positive result the gate LED lights briefly. Immediately after this the EPROM is tested: all the bytes are summed and a checksum byte is added. The result must be $00. Any other result indicates one or more faults in the EPROM and the stop LED then lights. The power can now be switched off and the short circuit for S3 removed, as can the wire used to provide pulses to the NMI line. Connect the 6 V winding of the mains transformer to the display board. When the power is now switched on the display shows the text OVERFLOW, PLEASE RESET or FREQ. X.XXXXX HZ. This is an indication that the display board is correct. If nothing appears on the display the connections to the display board from the main board and from the transformer should be checked. Test also that the 6 V a.c. is present. Remove the power again and install the remaining ICs into the main printed circuit board. The cable from the front panel can also be plugged into its connector. (This cable is actually a thin plastic 'tail' with the wires from the front panel's switches embedded into it.) Treat the ICs carefully, especially the 74LS232 which is very sensitive. Once again the power is applied to the circuit. The display should now show FREQ. 0.000000 HZ. Press the menu button once and the NO button three times. The display is now EVENT COUNT? Press the YES button three times and the display will indicate TOTAL 0. Apply a square wave (about 1 Hz) at TTL level to pin 2 of K3 and measure if this signal is also present on pin 9 of IC1, pin 3 of IC3, pin 7 of IC4, pin 8 of IC9 and pin 1 of IC5. Both pin 6 of IC9 and pin 2 of IC5 should be zero.

If the circuit is working correctly each incoming pulse will be added to the total shown on the display. The trigger LED should also flash with each pulse and the gate LED should light continuously. If this does not happen the problem is almost certainly to be found in or around IC5. All the frequency meter's functions should now operate properly. Doubting Thomases can check this by comparing signals at various points in the circuit with the appropriate timing charts.

Casing the circuit

Before any part of the circuit can be fitted into the case the necessary ventilation holes must be drilled in both upper and lower sections. Assuming the recommended type of box is used the main printed circuit board will fit snugly inside parallel to the bottom and can be fixed to the mounting bosses with self-tapping screws. An insulating ring must be placed under the bolt beside K1. The mounting boss at the front left-hand corner of the

11
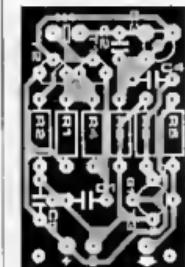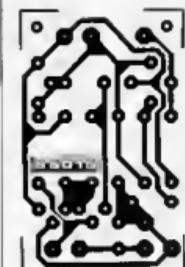
Figure 11. The small size of the printed circuit board for the crystal oscillator belies its importance. This is what makes the microprocessor tick.

Parts list

— Crystal oscillator

Resistors:
R1, R6 = 2k7
R2, R3 = 220 Ω
R4 = 33 k
R5 = 3k3

Capacitors:
C1 = 330 n
C2* = 22 p
C3* = 20 p trimmer
C4 = 1 n
C5 = 100 n

Semiconductors:
T1, T2 = BF494
T3 = BF900, BF905, BF907, BF961, BF981

Miscellaneous:
X1* = 10 MHz crystal with 30 p series capacitance

* = see text

top of the case must be removed so that the surface here is completely flush.

The attractive finished appearance and correct operation of the meter can only be assured if the front panel foil available through Elektor's EPS service is used. It is an essential part of the circuit as it contains the membrane switches used to control the frequency counter. This foil is quite thick so that the slots provided in the case to accommodate the metal panel supplied with the box will be too thin to accept both of these at the same time. For this reason a sheet of aluminium about 1 mm thick and with the same dimensions as the front panel supplied must be used instead. Use the template provided with the front panel foil to drill all the necessary holes in the new front plate. The display board can be fixed to the plate by means of four countersunk bolts or with four bolts stuck to the inside of the panel. The locations for these holes or bolts are indicated by the template and it is important that the bolts may not protrude by more than 16 mm. The holes for the BNC sockets are drilled undersize and then filed to the right shape and size. One side of each hole is flat so the sockets cannot turn. The slot for the front panel cable must also be cut and then a final check made to make certain that everything fits as it should. The backing paper can now be taken off the front panel foil. Pass the cable through the slot and then stick the foil carefully and accurately onto the aluminium front plate. Any excess foil should now be cut off with a sharp knife. There is a thin layer of protective plastic on the front of the foil that can be removed when the case is completely finished.

The wires to the mains switch can now be soldered in place. Note that they must rise straight up from the switch as otherwise they might foul the capacitors on the

display board. (The sketch in the margin here indicates what we have in mind.) Insulate the connections very well, with short lengths of heat-shrink tubing, for example.

All the components must now be fixed to the front panel. A 10 k potentiometer must be fitted to the hole left in the display printed circuit board. The type used should ideally have small dimensions and an insulating washer must be used at the copper side of the board. This potentiometer will be used later — when the input stage is added to the meter.

Bend the display carefully towards the board and over IC6. Do this correctly and the display will fit exactly in the window in the front panel. Fix the display board to the front panel by means of the bolts already discussed.

The mains transformer is now fixed to the rear panel of the case so that its mounting bracket is about 2 mm below the top of the box. Fit the transformer above IC15, IC7 and IC17 in order to leave room for the input stage next month. Also fitted to the back panel is IC20 and in this case some heat-conducting paste must be used between IC and metal. The mains cable, naturally enough, feeds through the back panel, preferably by means of the chassis plug and socket arrangement normally used for laboratory test instruments. Before screwing the top onto the case a heatsink must be fitted onto IC21.

The oscillator in the circuit can only be correctly calibrated with reference to another (accurate) frequency counter. Measure the frequency at pin 1 of IC18 and use the trimmer to set this to exactly 10 MHz. Put the top onto the case and leave the meter on for about half an hour. Measure the frequency again and if necessary re-trim C3.

The constructional details of this frequency counter have been dealt with in much more detail than is usual in our circuits. We felt that this was necessary as it is a very important test instrument and the instructions given should enable any hobbyist to build a working example. Next month we will describe the input amplifier with the prescaler. All of the counter's functions are self-explanatory and selection is simplicity itself but it is always useful to familiarise yourself with any new equipment by experimenting with it. ◼

84497 8

Table 2 This hexdump is the program that tests, drives and controls the frequency counter circuit.

CMOS function generator

The aim of this project was to produce a simple, cost-effective, general purpose audio generator, which was easy to build and use. This aim has certainly been achieved, since the circuit offers a choice of sine, square and triangle waveforms and a frequency range from about 12 Hz to 70 kHz, yet uses only one CMOS hex inverter IC and a few discrete components. Of course, the design does not offer the performance of more sophisticated circuits, particularly as regards waveform quality at higher frequencies, but it is nonetheless an extremely useful instrument for audio work.

Using only one inexpensive CMOS IC and a handful of discrete components, it is possible to build a versatile function generator that will provide a choice of three waveforms over the entire audio spectrum and beyond.

Block diagram

Figure 1 illustrates the operating principles of the circuit. The heart of the generator is a triangle/squarewave generator consisting of an integrator and a Schmitt trigger. When the output of the Schmitt trigger is high, the voltage fed back from the Schmitt output to the input of the integrator causes the integrator output to ramp negative until it reaches the lower trigger threshold of the Schmitt trigger. At this point the output of the Schmitt trigger goes low, and the low voltage fed back to the integrator input causes it to ramp positive until the upper trigger threshold of the Schmitt trigger is reached. The output of the Schmitt trigger again goes high, and the integrator output ramps negative again, and so on. The positive-going and negative-going sweeps of the integrator output make up a triangular waveform, whose amplitude is determined by the hysteresis of the Schmitt trigger (i.e. the difference between the upper and lower trigger thresholds). The output of the Schmitt trigger is, of course, a square wave consisting of alternate high and low output states.

The triangle output is fed through a buffer amplifier to a diode shaper, which 'rounds off' the peaks and troughs of the triangle to produce an approximation to a sinewave shape.

Any one of the three waveforms may then be selected by a three-position switch and fed to an output buffer amplifier. The frequency of all three

signals is varied by altering the integrator time constant, which changes the rate at which the integrator ramps, and hence the signal frequency.

Complete circuit

The practical circuit of the CMOS function generator is given in figure 2. The integrator is based on a CMOS inverter, N1, whilst the Schmitt trigger uses two inverters with positive feedback, N2 and N3.

The circuit functions as follows; assuming, for the moment, that the wiper of P2 is at its lowest position, when the output of N3 is high a current

$$\frac{U_b - U_t}{P_1 + R_1}$$

flows through R1 and P1, where U_b is the supply voltage and U_t is the threshold voltage of N1. Since this current cannot flow into the high impedance input of the inverter, it all flows into C1 or C2 (depending on which is selected by S1).

The voltage drop across C1 thus increases linearly, so the output voltage of N1 falls linearly until the lower threshold voltage of the Schmitt trigger is reached, when the output of the Schmitt trigger goes low. A current

$$\frac{-U_t}{P_1 + R_1}$$

now flows through R1 and P1. This current also flows into C1, so the output voltage of N1 rises linearly until the upper threshold voltage of the Schmitt trigger is reached, when the output of the Schmitt trigger goes high and the whole cycle repeats.

To ensure symmetry of the triangle waveform (i.e. the same slope on both positive-going and negative-going portions of the waveform) the charge and discharge currents of the capacitor must be equal, which means that $U_b - U_t$ must equal U_t. Unfortunately U_t is determined by the characteristics of the CMOS inverter and is typically 55% of supply voltage, so $U_b - U_t$ is about 2.7 V with a 6 V supply and U_t is about 3.3 V.

This difficulty is overcome by means of P2, which allows symmetry adjustment. Assume for the moment that R4 is connected to the positive supply rail (position A). Whatever the setting of P2, the high output voltage of the Schmitt trigger is always U_b. However, when the output of N3 is low, R4 and P2 form a potential divider so that a voltage from 0 V to 3 V can be fed back to P1, depending on the wiper setting of P2. This means that the voltage across R1 and P1 is no longer $-U_t$ but
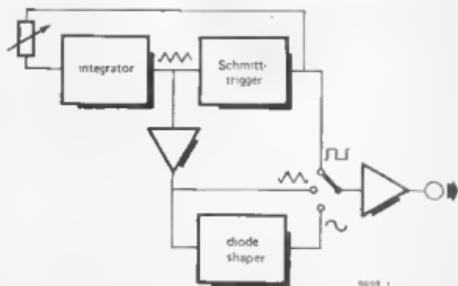
Figure 1. Block diagram of the CMOS function generator.
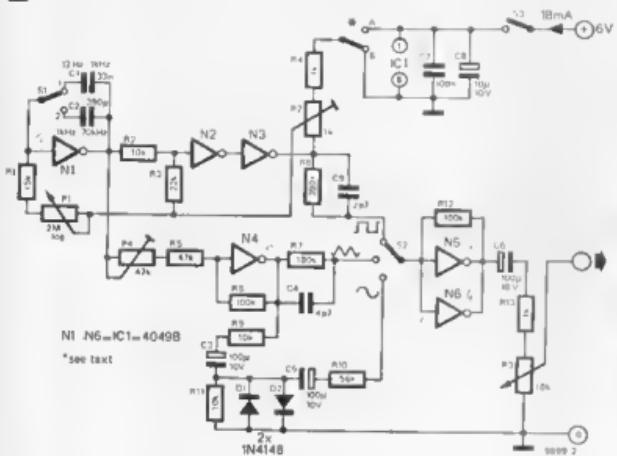
Figure 2. Complete circuit of the function generator.

Photos. The three output waveforms produced by the function generator.

N1...N6=IC1=4049B
*see text
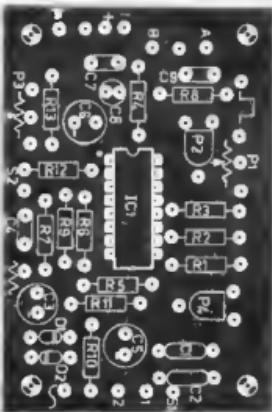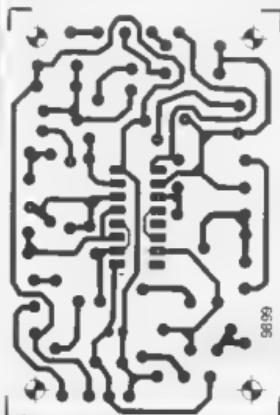
Up$_2$ – U$_t$. If the slider voltage of P2 is about 0.6 V then Up$_2$ – U$_t$ will be around –2.7 V, so the charge and discharge currents will be the same. Of course, the adjustment of P2 must be carried out to suit each individual function generator, owing to the tolerance in the value of U$_t$. In cases where U$_t$ is less than 50% of the supply voltage, it will be necessary to connect

the top of R4 to ground (position B). Two frequency ranges are provided, which are selected by means of S1; 12 Hz–1 kHz and 1 kHz to about 70 kHz. Fine frequency control is provided by P1 which varies the charge and discharge current of C1 or C2 and hence the rate at which the integrator ramps up and down.

The squarewave output from N3 is
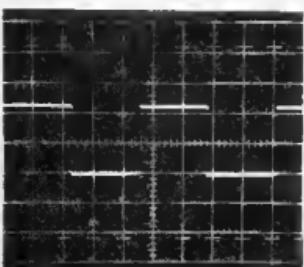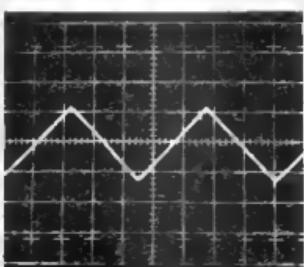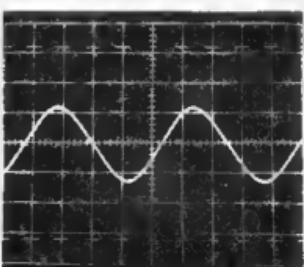
taken via a waveform selector switch, S2, to a buffer amplifier, which consists of two inverters (connected in parallel to boost their output current capability) biased as a linear amplifier. The triangle output is taken through a buffer amplifier N4, and thence through the selector switch to the output buffer amplifier.

The triangle output from N4 is also taken to the sine shaper, which consists of R9, R11, C3, D1 and D2. Up to about plus or minus 0.5 volts D1 and D2 draw little current, but above this voltage their dynamic resistance falls and they limit the peaks and troughs of the triangle signal logarithmically to produce an approximation to a sine wave. The sine output is fed via C5 and R10 to the output amplifier.

Sine purity is adjusted by P4, which varies the gain of N4 and thus the amplitude of the triangle signal fed to the sine shaper. Too low a signal level, and the triangle amplitude will be below the diode threshold voltage, so that it will pass without alteration, too high a signal level, and the peaks and troughs will be clipped severely, thus not giving a good sine wave.

The input resistors to the output buffer amplifier are chosen so that all three waveforms have a peak to peak output voltage of about 1.2 V maximum. The output level can be adjusted by P3.

Adjustment procedure

The adjustment procedure consists simply of adjusting the triangle symmetry and sine purity. Triangle symmetry is actually best adjusted by observing the squarewave signal, since a symmetrical triangle is obtained when the squarewave duty-cycle is 50% (1-1 mark-space ratio). P2 is adjusted to achieve this. In cases where the symmetry improves as the wiper of P2 is turned down towards the output of N3 but exact symmetry cannot be obtained, the top of R4 should be connected in the alternative position.

Sine purity is adjusted by varying P4 until the waveform 'looks right' or by adjusting for minimum distortion if a distortion meter is available. Since the supply voltage alters the output voltage of the various waveforms, and hence the sine purity, the circuit should be operated from a stable 6 V supply. If batteries are used they should never be allowed to run down too far.

CMOS ICs used as linear circuits draw more current than when used in the normal switching mode, and the supply voltage should not be greater than 6 V, otherwise the IC may overheat due to excessive power dissipation.

Performance

The quality of the waveforms can be judged from the oscilloscope photographs. In all three cases the vertical sensitivity is 500 mV/div and the timebase speed 200 μs/div. ■

simple transistor tester
—an adapter for your multimeter.
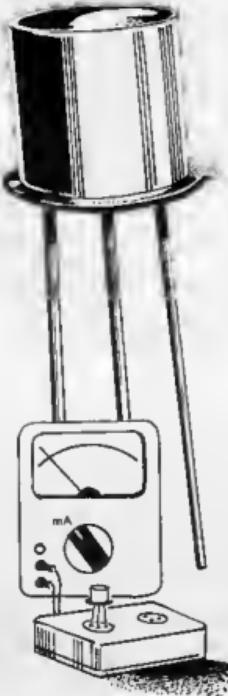
This simple circuit checks the functioning and measures the current gain (h_{FE}) of PNP or NPN bipolar transistors. It operates by feeding a known constant current into the base of the transistor and measuring the collector current. Since the collector current of a non-saturated transistor is h_{FE} times the base current (which is known) it is a simple matter to calculate the value of h_{FE}, and in fact the meter which measures the collector current can be calibrated directly in h_{FE}.

Since both PNP and NPN transistors must be tested, two constant current sources are required, to provide a negative base current for PNP transistors and a positive base current for NPN transistors. The voltage dropped across the LED causes a constant current to flow through the emitter resistor of the TUP and a corresponding constant collector current, which flows into the base of the NPN transistor under test. This current can be set to 10 μA by connecting a 50 μA meter between points B and E, and adjusting P1.

The lower LED and TUN constitute the negative current source. Here again, this may be set to 10 μA by connecting a microammeter between the lower points B and E, and adjusting P2.

When a transistor is plugged into the appropriate socket a current of 10 μA will thus flow into the base and a current of h_{FE} times this will flow through the milliammeter. The full-scale deflection of the milliammeter depends on the maximum h_{FE} to be measured. Since the collector current is h_{FE} times the base current (which is 10 μA) a reading of 1 mA corresponds to an h_{FE} of 100, so if a 5 mA meter is to hand it can be calibrated in h_{FE} values from 0 to
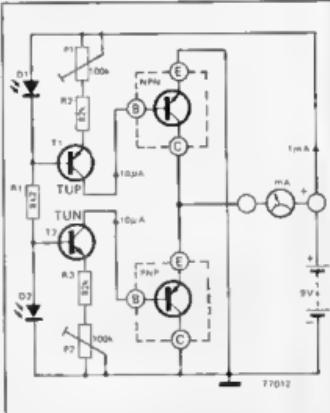
500, which should be adequate for most run-of-the-mill transistors. However, for testing 'C' versions of small-signal transistors, which can have gains up to 800, a 10 mA meter calibrated 0 to 1000 could be used, or a lower f.s.d. meter 'shunted' to read 8 mA and calibrated 0 to 800.

Readers may have noticed that it is actually the emitter current of the PNP transistor that is measured, which is of course 1 + h_{FE} times the base current. However, since few transistors have gains less than 50 the worst error introduced by this is less than 2%, which is probably less than the error of the milliammeter. ■

economical crystal time base

a 50 Hz 'bench mark'

This time base circuit is built using normal readily available CMOS ICs and a cheap crystal. The circuit gives the constructor the possibility of 50 Hz, 100 Hz or 200 Hz. The 50 Hz reference frequency is an ideal time base for the construction or calibration of electronic clocks, frequency meters and so on. Because of the flexible supply voltage requirement, it is also a good basis from which to build a digital clock for the car.

IC1 contains an oscillator and a 2^{14} divider. Providing the oscillator loop is correctly calibrated using C2, the output at pin 3 (Q14) will produce a 200 Hz square wave. With the help of the two flip-flops in IC2
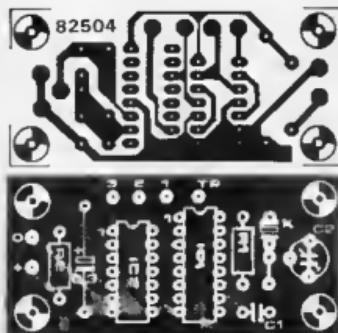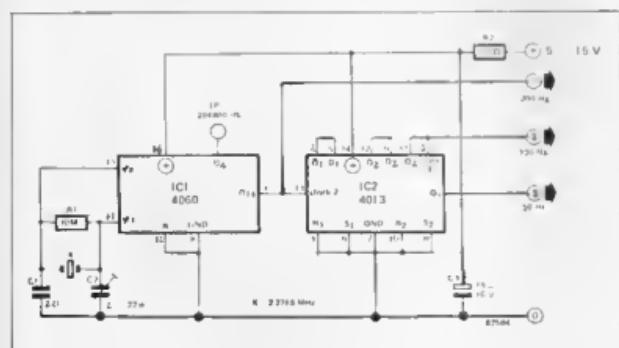
this square wave voltage is then divided by two and by four resulting in two further outputs of 100 Hz and 50 Hz, the latter from pin 1. Readers who have a frequency meter can calibrate the circuit by simply connecting the meter to pin 7 of IC1 (Q4) and adjusting C2 until a reading of 204,800 Hz is indicated. As a matter of interest, anyone without a frequency meter should not despair since setting trimmer C2 to about midway will provide sufficient accuracy for most applications.

The 100 Hz output is useful for the construction of digital counters. For this purpose we suggest that a 1 : 10 divider (like the 4518) is connected to the 100 Hz output pin. The power supply requirements are:
from 5 ... 15 V and 0.5 ... 2.5 mA.

---

low voltage stabiliser

battery powered voltage regulator

Depending on their condition 1.5 V batteries supply a voltage of 1.2 ... 1.7 V. This circuit can be very useful when a project has to be fed with a constant, low voltage. With an input voltage of 1.2 ... 1.8 V this stabiliser produces a relatively constant voltage of 1.15 V with a maximum load of 5 mA.
T2 cuts off at a minimum battery voltage of 1.2 V with a load of 5 mA. The output voltage tends to increase with a higher battery

voltage, causing T2 to conduct and reducing the base current of T1 and T3 (indirectly), so that the output voltage will remain 1.15 V.
The internal impedance of this low voltage supply is 1 to 2 Ω. The output voltage will only be reduced by 70 mV when changing the battery voltage from 1.8 V to 1.2 V.

(ITT application)

When deciding which capacitors to use, consideration should be given to reliability, the permissible range of operating conditions, size and so on. Size is important especially when building high density circuits, and last but not least price. Keep in mind, that, any need for special current limiting resistors is going to increase the overall cost of using tantalums. Even so, tantalum capacitors are used widely, where the operating characteristics of the capacitor is critical. Quite a few Elektor circuits specify the use of tantalums, and not just because they are small and good to look at. They have a stable capacitive value, and a long shelf life. The impedance is virtually unaffected by frequency changes. So, on the face of it tantalums are ideal. However, they do have one major drawback; price!

applications, and certainly fell down, for high frequency designs.

From the offset for many applications the tantalums did not have too much competition, and with the craze for miniaturisation they filled a need straight away. The biggest factor initially in favour of tantalums was this question of shelf life. Even after years of storage, their current leakage, and value remain unchanged. In fact their shelf stability is about 100 times better than the wet aluminium type.

Apart from everything so far explained the tantalums were not embarrassed by temperature. The wet electrolytics are affected quite considerably by temperature, causing large increases in the leakage current level. It is this failure to dissipate heat that causes their mediocre conductance.

electrolytics run dry

everything you wished to know!

So far for many industrial and professional applications wet electrolytics and tantalums have ruled the roost. With new innovations and technological advancements, it is now possible to use alternatives which can be cheaper and more reliable. Many factors need consideration when making the right choice, and a good knowledge of the merits and limitations of the different types available is useful.

In fact a comparison shows that the new solid aluminium electrolytics, can be used as alternatives for tantalums, and in many respects can be seen to be better.
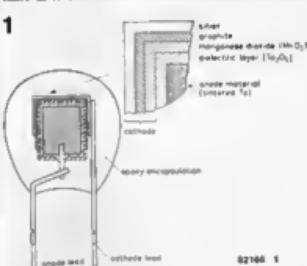

Figure 1. Diagram of the structure of tantalum electrolytic.

With such an incentive as this, some type of alternative was needed.

With the constant improvement in technology, coupled with energy saving and the conservation of natural resources, many experts started to question the sanity of using tantalum for capacitors. Tantalum is now in limited supply, as with everything else, and its price is increasing by leaps and bounds. So, why use tantalums?

We must first of all keep in mind that at the time that they emerged onto the market, the only other type, with which they could be compared, were the wet aluminium electrolytics. These were and still are inexpensive, but are generally larger than any solid counterpart and suffered from a relatively short shelf life. By this we mean, that long storage, their leakage current increases, and can only be restored to its original value by post-forming. Also, if they are used close to their maximum operating temperature, their life expectancy, which is normally far less than solid types, is curtailed even more. The wet type could not be used in certain new

By comparison the tantalums have a wide operating range as far as temperature is concerned, making them suitable for filters and oscillators. Hence the reason why they are widely used in Elektor designs.

Most of you by now must probably think that the writer must be completely sold on tantalums. Not so! They do have, what can be termed as inconveniences, rather than faults or disadvantages:

● The voltage level they can sustain when connected the wrong way round is extremely small, even for a very short interval. They breakdown rapidly and can explode easily.

● Their a.c. voltage performance is poor and further diminished at high frequencies and temperatures.

● The charge/discharge rate resistance is 3 Ω/V, making it necessary to use series resistors.

● A surcharge, whether it is of a thermo, current, or voltage nature, will cause immediate breakdown, short circuiting and a possible explosion.

● The price of each item is quickly approaching prohibitive levels.

All in all tantalums are certainly not perfect, mind you what is these days! Should series resistors not be used in order to limit the charge/discharge rate, then the results are always fatal. This is because field crystallisation will occur, causing short circuiting.

At the beginning of the article we explained all the advantages of using tantalums; impedance, heat dissipation, life span, high frequency performance and so on. But, it seems that it did not take us very long to arrive at the conclusion that even these are not as good as we would wish. As with many things, it is a fact of life that the more you use something the less appealing it becomes. Notice that we do not say everything,

since the good things in life are always welcome.

Luckily the capacitor manufacturers have not stood still and have come up with a relatively new development.

Using deeply-etched foil an axial-lead solid aluminium electrolytic has been created, achieving a high CV density making them less expensive replacements for tantalums. Although they are not going to depose the latter completely, they will be very widely used in a variety of industrial and professional equipment.

Solid aluminium capacitors

The solid aluminium electrolytic has a comparable performance with the tantalum type, but not only is it cheaper, but it does have a few advantages.

Figure 1 shows the different com-

Figure 3. Structure of a solid aluminium electrolytic.

pyrolitically - formed manganese - dioxide cathode

epoxy resin

anode lead

silver epoxy on graphite

etched end formed aluminium anode

'kink' to prevent coating running down lead

cathode lead

Figure 5. Stability of the main electrical parameters of the three types of electrolytic as a function of time. The curves are for a 0.1 μF item measure at 85°C.
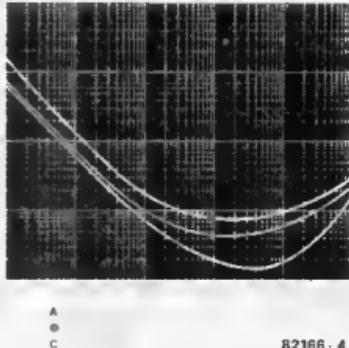
Figure 4. Impedance as a function of frequency with capacitor type as a parameter. All capacitors are 33 μF, 10 V.

82166 · 4

ponents which go to make a tantalum. There are a lot of similarities in construction with the solid aluminium type (SAL). Looking at figure 1, you will note that the former has layers of silver, graphite and manganese dioxide (MnO_2) which form the cathode. Then comes a dielectric layer and finally the anode made of tantalum. This is sintered to the cathode (inside dielectric layer).

Figure 3 shows the make up of a SAL. The real difference between the two lies in the fact that, the anode is composed of deeply etched aluminium and that the dielectric layer is aluminium oxide AL_2O_3. Hence the remarkable conductivity of the solid aluminium electrolytic!

These SALs, to coin a phrase, are very robust to say the least. They can operate near their maximum temperature ratings without shortening their life span, and do not have any catastrophic failure mechanism. In other words they are not going to blow-up in your face at the wrong moment. An added bonus is the fact that series resistors are not needed. The values already available are in the range 47...1000 μF and one major manufacturer has proposed to make smaller ones with 0.22...47 μF, but, it may take some time before these are available.

They are slightly larger than their equivalent counterparts, and although being less expensive than tantalums they are marginally more costly than the wet type.

Present applications include telecommunications, space programs, and power stations. Their small size and robustness make ideal for the automobile industry. Because they are being improved upon all the time, a rosy future lies ahead of them.

To summarise the main characteristics of the SAL are:
* Lower price.
* Their voltage rating remains unchanged throughout the operating range, even at high temperatures (−80 to 175°C).
* The allowed d.c. voltage (reversed) is around 33% of their rated voltage.
* Does not require current limiting.
* a.c. voltages (up to limits) can be handled and do not adversely affect their performance.
* Their impedance fall more steeply with increasing frequency than any other type.
* They can withstand 50/100 Hz a.c. voltages up to a level which is 80% of their d.c. rating.
* Temperature stable, and low failure rate coupled with longevity.

Output amplifier IC Type LM 1875

The LM 1875 from National Semiconductor is a monolithic output amplifier that can provide up to 20 watts audio power of excellent quality. It is housed in a TO-220 encapsulation in which you would normally expect no more than a small voltage regulator.
Maximum output power is 35 watts into 8 ohms, but the distortion at this level is no longer acceptable for hi-fi purposes. At 20 W, however, the harmonic distortion is only 0.05% at 1 kHz. The bandwidth is a very reasonable 70 kHz, while the slew rate of 8 V/µs is excellent for this

type of IC. Other plus points are the very good hum suppression of 94 dB, the thermal protection circuit, and the short-circuit protection.
Further characteristics are given in Table 1.
The LM 1875 has five connecting pins: two for the supply voltages (positive and negative), one for the output, and two for the inputs (inverting and non-inverting).
Two possible circuits are given in figure 1: one with a symmetrical supply (a) and the other for use with an asymmetrical power supply (b). Figure 1a is noteworthy for the minimal number of additional components required: two diodes, D1

and D2, for the protection of the output transistors, high-pass filter C5/R5, input filter C1/R1, negative-feedback network R2/R3/R4/C2, and decoupling capacitors C3 and C4. The gain, A, is determined by
$$A = 1 \div R4/R3.$$
The characteristics in figure 2 show the total harmonic distortion, THD, versus power output and frequency and the power output versus the supply voltage.
Because of its small dimensions, the quality of the output, and the small number of external components required, this IC is eminently suitable for use in active loudspeaker systems

Table 1

Absolute Maximum Ratings

Supply Voltage	± 30 V
Input Voltage	V_{EE} to V_{CC}
Operating Temperature	0°C to +70°C
Storage Temperature	−65°C to +150°C
Junction Temperature	150°C
Power Dissipation (Note)	30 W
Lead Temperature (Soldering, 10 seconds)	300°C

TO-220 Power Package (T)
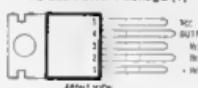
Electrical Characteristics

$V_{CC} = 30$ V, $-V_{EE} = -30$ V, $T_{TAB} = 25°C$, $R_L = 8 \Omega$, $A_V = 32$ (30 dB), $f_o = 1$ kHz, unless otherwise specified

Parameter	Conditions	Min	Typ	Max	Units
Supply Current	$P_{OUT} = 0$ W		80	100	mA
DC Output Level			0		V
Output Power	THD = 1%		30		W
THD	$P_{OUT} = 20$ W		0 05		%
	$P_{OUT} = 20$ W, $f_o = 20$ kHz		0 2	0 4	%
	$P_{OUT} = 30$ W		0.1		%
	$P_{OUT} = 30$ W, $R_L = 4 \Omega$		0.4	1.0	%
	$P_{OUT} = 20$ W, $R_L = 4 \Omega$		0 06		%
	$P_{OUT} = 20$ W, $R_L = 4 \Omega$, $f_o = 20$kHz		0.3	0.6	%
Offset Voltage		−30	± 5	30	mV
Input Bias Current		−5	−2	5	µA
Input Offset Current		−1.5		1.5	µA
Input Sensitivity	$P_{OUT} = 20$ W, $f_o = 20$ kHz		400	450	mVrms
Open Loop Gain			90		dB
PSRR	V_{CC}, 120 Hz, 1 Vrms	52	93		dB
	$-V_{EE}$, 120 Hz, 1 Vrms	52	95		dB
Max Slew Rate			8		V/µs
Current Limit		3	4		A
Equivalent Input Noise Voltage	$R_S = 600 \Omega$, CCIR		3		µVrms

Note Assumes T_{TAB} to 60°C max. For operation at higher tab temperatures and at ambient temperatures greater than 25°C, the LM 1875 must be derated based on a maximum 150°C junction temperature. Thermal resistance depends upon device mounting techniques.

Table 1. Absolute maximum ratings and electrical characteristics.
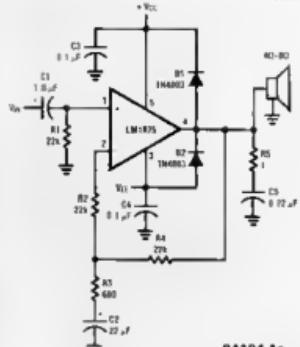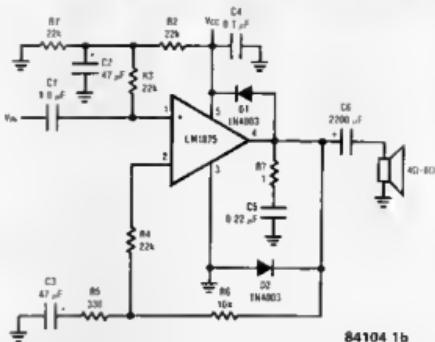
Figure 1. Two possible circuits for an output amplifier based on the LM 1875: one for use with a symmetrical power supply (a) and the other for an asymmetrical supply (b).
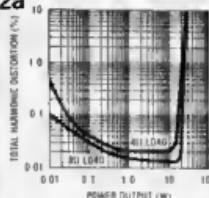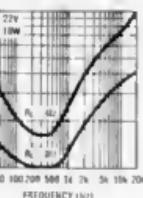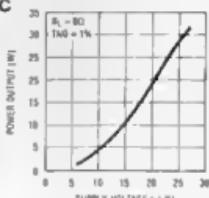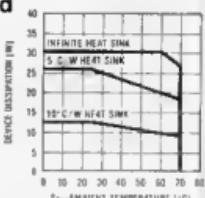

Figure 2. Total harmonic distortion versus power output (a); total harmonic distortion versus frequency (b); power output versus supply voltage (c); internal dissipation versus ambient temperature (d).
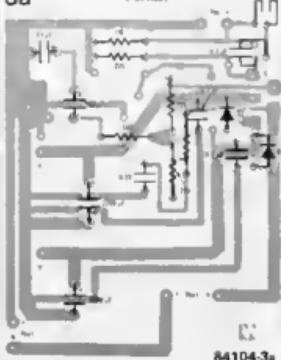

Figure 3. Printed-circuit layout of the two suggested circuits of figure 1: (a) with symmetrical and (b) with asymmetrical power supply.

Denmark:
Bianco Lunos Alle 1
1868 Copenhagen V
Phone: (01) 213211

Southern Europe:
Via Solferino 19
20121 Milano
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Phone: (02) 6596146

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c. Agustin de Foxa 27 (9 d.)
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España
Phone: (01) 7332954

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S-127 02 Skärholmen
Sverige
Phone: 08-970190

Outside Europe:
2900 Semiconductor Drive
Santa Clara
California 95051
USA
Phone: (408) 721-5000

Literature:
National Semiconductor:
Preliminary Application
Note on the LM 1875

National Semiconductor addresses:
United Kingdom:
301 Harpur Centre
Home Lane
Bedford
Phone: (0234) 47147

SOLDERING STATION

The SOLDEX soldering station has been developed specially for soldering micro electronic components like LSI chips, MOS devices and other sensitive components. The bit size range is 1.5, 2.5 and 3.5 mm diameter, for precision soldering work. The construction is slip in type and provides easy interchangeability.

The station contains a soldering stand and the bit is positively connected to the earth pin of the power plug to prevent static build up.

Nominal power consumption is 25 watts, but the circuit can draw upto 50 watts if necessary. The soldering iron weighs 95 gms.

For further information, write to Thermal Sensors 37, Electronics Complex, Kushaiguda, Hyderabad 500 762

MICRO ANALYSER

Electronumerica have introduced EN 85, a versatile analyser for software development, testing and trouble shooting of 8-bit systems based on 8085, Z-80 etc.

The EN 85 is a compact unit with stand alone capability and is useful for manufacturing industries as well as in field applications. This analyser system is used to examine and modify the memory locations, execute the programme upto a particular address and stop or step through a programme one instruction at a time or one machine cycle at a time.

For further information, write to: Electronumerica Kommaghandahalli, Opp. HMT Industrial Estate Jalahalli (West) Bangalore 560 015

DIGITAL MEGGER

Arun Electronics have developed a new insulation tester with digital readout. This fully solid state unit does not require any hand driven mechanism for generating test voltages. The instrument operates directly from the mains supply and has built-in DC to DC converters to generate the test voltages.

The measurement ranges available are 20, 2000, and 10000 Megaohms. Linearity is claimed to be 0.1% throughout the range.

For further information, write to: Arun Electronics Pvt. Ltd B—125, Ansa Industrial Estate, Saki Vihar Road, Saki Naka, Bombay 400 072

WIRE STRIPPER

Efficient Engineers have developed a thermal wire stripper which gives neat, clean and fast stripping of PVC and Malcon coated wires, flat cables and Taflon coated cables

This is an easy to use tool which does not require adjustments for varying wire diameter. It can strip wires with a wide range of diameters. A strip stop guide is provided to set the length of insulation to be stripped. The unit operates directly on mains supply

For further information, write to: SAI Electronics Thakore Estate, Kurla-Kirol Road, Vidyavihar (West), Bombay 400 086

STEPPING MOTOR DRIVES

Spectron offers 'Ameya' stepping motor drives in 3 models

The possible application areas are X-Y-Z positioning and movement, Rotary and linear Indexing, Feeders and length control, Flow control etc

The range covers stepping speeds upto 5,000 steps/sec driving upto 60 Kg-Cm torque rated stepping motors

Standard controls such as Start/Stop, Direction selection, Jog/Slew selection and Declaration Ramp and Linear step-rate control etc. are provided Optional features are Remote control, Single or Tandem drives, Indexing, Interface with microprocessor keyboard or Thumb wheel switches
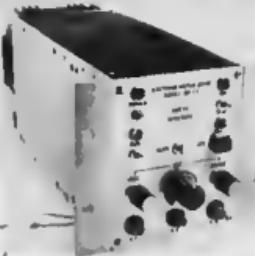
For further information, write to: Spectron Sales and Service Pvt Ltd 63, Bherakunj No. 2 Erandwane, Pune 411 038

RELAY SOCKETS

Essen Deinki now introduce two new plug-in type relay sockets The DS-08 is for 8 pin and DS-11 is for 11 pin configurations These sockets can be snap mounted on DIN rail having only 36 mm length of the rail. They can also be mounted on chassis using two screws The body is moulded in flame retardant glass filled thermoplastic resin. Screw terminals are provided with captive U shape terminals for secure wiring Snap on receptacles can also be connected to the spade terminations after removing screws and washers.
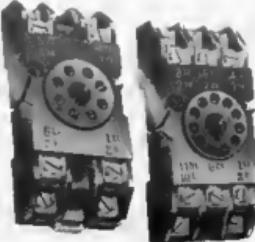
For further information, write to: Essen Deinki 386, Industrial Area, Chandigarh 160 002

DIGITAL MULTIMETERS

Digital Instruments Corporation have introduced three new models of digital multimeters, DM 520, DM 525, DM 555. All the models have three and a half digit, half inch LED/LCD display. The measurement ranges are AC/DC voltages from 200 mV to 600 V rms AC/1.2 kV DC, AC/DC currents from 200 µA to 2 A and resistances from 100 milliohms to 20 megaohms. Input impedance is claimed to be very high.

Model DM 525 has facility for optional external shunts. Models DM 520 and DM 525 have very high accuracy in AC measurement over the entire audio range of frequencies. All models are provided with autopolarity and decimal points, they are fully protected against overloads in all ranges.

For further information write to Digital Instruments Corporation Near Tower, Saipur Bogha Naroda Road, Ahmedabad-382 345

CONTACT CLEANER

Kli-Nit Contact cleaner CT-2 is now available as an extension for pin point application on the inaccessible electrical contacts. Kli-Nit contact cleaner CT-2 when sprayed, not only cleans the contact but also reduces arcing as well.

The product is now available in 250 ml and 350 ml aerosol containers to suit the requirements of different users.

For further information, write to: Hekotronics Pvt Ltd Dadoji Konddeo X Marg Bombay 400 027

CLIMATE ANALYSER

Indoor Climate Analyser Type 1213 is a handy, easy to operate, portable instrument for evaluating all of the basic parameters which influence the thermal environment and its effect on human life.

The analyser has a 20 character alpha numeric display. It has five transducers which can measure air temperature, surface temperature, radiant temperature assymetry, humidity and air velocity. The measurements are carried out and recorded automatically. Four different recording periods of 1, 6, 24, and 120 Hours are available. Measurements are spaced evenly throughout the recording period and the analyser can be left unattended.
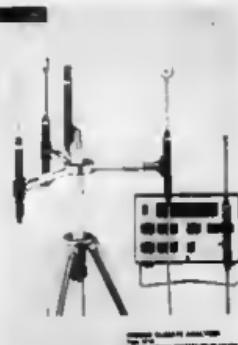
For further information, write to Jost's Engineering Co Ltd, 60, Sir Phirozeshah Mehta Road, Bombay 400 001

PRESS CLIPS

SEE press clips have unlimited uses in clamping applications. They offer a fast and easy way to hold cables and wires. The clips are provided with an adhesive backing for long lasting hold and are available in full or half styles. Material used for manufacture is claimed to be non conductive, non corrosive and resistant to most common solvents, alkalies, dilutes acids, petroleum oil and greases.

For further information, write to: Suresh Electrics and Electronics Manasarovar, 3 B, Camac Street, Calcutta 700 016

LINE FREQUENCY METER

The new line frequency meter from JIVAN Electro Instruments is based on the latest CMOS technology for high stability and noise immunity. The display is 4 digit, half inch seven segment LEDs.

The resolution is 0.01 Hz and the accuracy is claimed to be ±0.01% of the reading. The unit can be made available in two ranges of 99.99 Hz and 999.9 Hz. High and low alarm facility can also be provided.

For further information, write to: JIVAN Electro Instruments 394, G I D.C. Estate, Makarpura, Barode 390 010.

MINIATURE HIGH-V RELAYS

Kilovac Corporation have announced a complete new range of miniature High-Voltage relays. Available in SPDT, SPST, fail-safe and latching configurations, the range includes relays with rahngs upto 10 kV. The light weight miniature relays are offered in low cost commercial version and also in a military version. They are suitable for application in digital antenna couplers, laser systems, medical instruments and several other industrial applications.

For further information, write to Toshni-Tek International 267, Kilpauk Garden Road, Madras 600 010.

missing link

PARSER
(August/September 1984, page 8-94)

This program does not work on some computers but the situation can be improved by doing the following.

remove lines 160 and 1110
copy lines 1000...1100 between lines 150 and 170 with line numbers from 151 to 161
line 151 — ...THEN 1080 becomes ...THEN 159
line 120 — ...THEN 1090 becomes ...THEN 160
line 140 — ...THEN 1080 becomes ...THEN 159

capacitance meter
(March 1984, page 3-42)

The note in the margin on page 3-55 is incorrect. It should read

Meter board:
P1 sets the display to '0' in range a
P2 calibrates ranges a, b and c
P3 sets display to '0' in ranges b... f

funny bird
(summer circuits,
Page 6-82)

T1 is drawn erroneously as an NPN type, whereas it is, of course, a PNP type

FM pocket radio
(August/September 1984, page 8-53)

Transistor T8 must be BC 560C with the collector connected to +5 V.
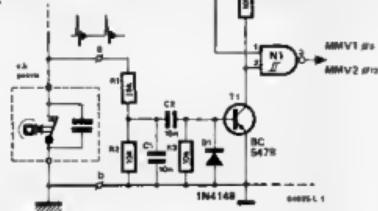
valve amplifier
(December 1984)

Apparently not all the cobwebs were blown away before we finalised this article. The test voltage of −10.4 V given at pin 2 of the EL84s should, in fact, be 10.4 — 10.4 V referred to in the last sentence of the difference between cathode and gnd rather than an absolute value.

burglar deterrent
(December 1984)

In figure 2 and the parts list R7 is given as 1 k/½ W. Depending on individual circumstances, it may be necessary to change this for a value of 1 k/1 W.

ZX81 cassette pulse cleaner
(November 1984)

The first paragraph of the text states that a logic one is represented by eight pulses. This is incorrect; it should, in fact, read nine pulses.

real time analyser (part 1)
(April 1984, page 4-740)

The voltage regulators should be type 7812 and 7912 respectively as shown in the circuit diagram. Where the text and the parts list read 7808 and 7909 these should be corrected to 7812 and 7912 respectively. Likewise, voltages of +8V and −8V should read +12V and −12 V.

how to recycle dry batteries
(November 1984, page 11-58)

In fig. 5 the resistor mentioned as 220 ohms should actually be 2 k 2 ohms in the last sentence of the article the mention of S1 is incorrect it should be S2

anodizing aluminium
(October 1984, Page 10-487)

We have been advised by a reader that caustic soda lye CAN be stored in glass bottles, provided they do not use a glass stopper fitted with a screw plastic cap or rubber stopper they are perfectly satisfactory If a glass stopper is used, carbon dioxide is gradually absorbed from the atmosphere (because the stopper is a poor seal) and the contents are eventually converted to sodium carbonate (washing soda). Furthermore, if caustic soda solution gets onto the ground glass of the glass stopper, this again is converted to sodium carbonate which can make the stopper quite unremovable in a matter of weeks

direct-coupled modem
(November 1984, Page 11-347)

It appeals that the missing link in our November issue is still not the complete solution to FF3 setting inadvertently on switch on A satisfactory solution is to connect a 470 pF capacitor between pins 7 and 13 of IC7 and to ensure that C21 in positions TELEPHONE and MODEM is connected to earth through one of the unused sections of switch S2

digital tachometer
(October 1984, page 10-36)

The formula given in the last part of the marginal note beside figure 2 (page 8-42) is not right. It should state:
$f = 1/(2.2 \times C4 \times (R5 \div P1))$.
The frequency range is then 686 .1194 Hz.

elabyrinth
April 1984, Page 4-307

The last address and data (800 FF) in the hexdump in table 1 should be deleted

petrol saver
(April 1984, page 4-15)

In some cars, it may be advantageous to replace the part of the circuit shown in 1 by that in 2.
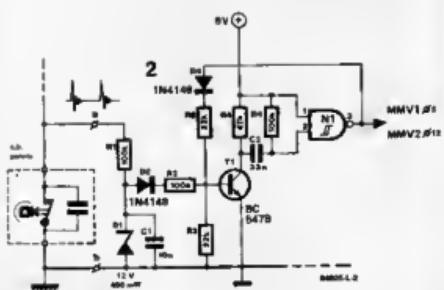
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