EMERSON
ELP TO RE-FORM?

THE DIRTY
DOZEN
ANSWER 12
QUESTIONS -
WIN AN AKAI AX80

REVIEWS:
LINN 9000 DRUM MACHINE
            + DIGITAL
            RECORDER
        SIEL DK80 POLYSYNTH
        ZLATNA PANEGA SAMPLER
        PEARL ELECTRONIC DRUMS
        YAMAHA TX7 VOICE EXPANDER
            + QX7 SEQUENCER
            DATEL SOUND SAMPLER

CONTENTS

COMMENT

Still Going Strong

It didn't strike me until this issue was already in a fairly advanced state of production that all three of this month's featured artists have appeared within E&MM's pages before.

Of the three, Tim Souster is the elder statesman, having graced the magazine's pages as far back as May 1981. E&MM was in its infancy then, and that particular month, came to its readers with a small piece of vero-board stuck to its front cover, together with six short features inside to give people ideas as to what to do with it. Times have certainly changed.

Keith Emerson appeared on E&MM's cover exactly two years later, and waffled on at such enormous length that his interview took up no fewer than nine pages of editorial. As for China Crisis, they've spent hardly a year away from us: they shared the front cover position with rock keyboardist Don Airey in February 84.

But quite apart from being featured in E&MM for the second time around, those three artists have something else in common – perseverance. They've all struggled at some length to achieve the elevated positions they now enjoy in their respective musical

fields, and they've all experienced times when self-doubt nearly got the better of artistic determination and caused them to jack it in.

At the time China Crisis were striving to gain recognition, the world was full of young hopefuls all attempting to get their message across, but the problems facing Souster and Emerson were of an entirely opposite nature. Both of them spent their musical apprenticeship trying to do what almost nobody else was doing, and both were forced to take what can only be described as circuitous routes to their eventual goals.

Souster spent years sponging off educational authorities around the world in order to pursue his musical ideals, for the simple reason that they didn't fit in with the accepted standards of the day. It wasn't until the late seventies that the composer could confidently free himself from the bonds of academic life and go it alone, and even then, living entirely from his music proved a struggle.

Keith Emerson had a similar predicament. He was a keyboard player at a time when the world only wanted to listen to guitarists, and was forced to over-emphasise his own virtuosity (to some extent in the studio, but mostly

on stage) simply to make the point that the keyboardist had just as big a contribution to make as anyone else.

Eventually, people listened to both musicians, and the world of modern music is a lot better off as a result. Yet although the state of play has changed to the extent that innovation (superficial or otherwise) is welcomed with open arms by most of the music industry, that doesn't mean to say perseverance is on the way out.

If anything, it's more necessary than ever, because there are now an awful lot of those young hopefuls vying for the industry's attention. If your bedroom wall is littered with rejection letters from record companies and you're lucky if you get more than 30 people to see you at each gig you play, it's the easiest thing in the world to give up on music and opt for merchant banking instead. But most of those that eventually achieve success do so because they have the courage to continue writing and playing their own music even when the odds are stacked against them.

So if there's a message to be gleaned from this month's crop of artist interviews it's this: Hang On In There. ∎

GiG SOUNDS

INTERFACE

Write to: Interface, E&MM, Alexander House, 1 Milton Road, Cambridge CB4 1UY

If you've a view, query or problem, write to E&MM at the above address. We will endeavour to answer every letter regardless of whether there is sufficient space for its inclusion in the magazine, so please include your full address and phone number.

South Bank Show

Dear E&MM,

I was surprised that the likes of Klaus Schulze and Tangerine Dream weren't even mentioned on the South Bank Show's electronic music special. Surprised because I'd believed such people were important to the development of electronic music. Are they?

Perhaps E&MM could come up with a more detailed study of the serious and influential musicians within the field.

And by the way, is the music shown on The South Bank Show available on record or cassette?

Paul Baxter
Coventry

May we just say that we were as surprised as anyone when the South Bank Show completed its programme about electronic music without mentioning either Schulze or Tangerine Dream, among many others. However, the show's musical consultant, Tim Souster, sheds some light on the reasoning behind this phenomenon in this month's Music section. The interview starts on page 54.

Some of the historical pieces featured in the programme are still available on disc – contact your local record shop for details. However, we aren't aware of any plans to release The Isle is Full of Noises, *the piece commissioned specifically for the programme, in any shape or form.*

On another tack, you may like to know that E&MM has already published a nine-part series entitled 'The History of Electronic Music' – see indices Feb 83 and 84 for details.

Max Settings

Dear E&MM,

I'm the owner of an SCI SixTrak who (like many, I would imagine) was not totally satisfied with the factory preset sounds. You mentioned in the SCI MAX review (Jan 85) that the new SixTrak models had the same voices as those supplied with the MAX.

Now, I've already altered some of the presets on my SixTrak, but would very much appreciate information as to how to set up the patches for any new voices – especially percussive sounds.

A Pearson
Derby

Sequential in Holland tell us that the new sounds from the MAX can be obtained simply by linking your SixTrak to a MAX and downloading the voices via MIDI; or you can load them in from disk using the appropriate Commodore hardware and software. Alternatively, Sequential will be pleased to let you have a photocopy of the actual patches if you drop them a line at PO Box 16, 3640 Mijdrecht, The Netherlands, though they'd appreciate an SAE (or International Reply Coupon) initially, as a small copying charge may have to be levied if they're inundated with requests!

Sample View

Dear E&MM,

I agree with Paul McPhee's comments last month that *Interface* is becoming dead boring. Unlike Mr McPhee, however, I have some suggestions on specific improvements.

A good topic of dicussion might be my belief that sound-sampling is the biggest setback ever to hit the synthesiser market. It's going to lead to a myriad of variations on existing themes, but it's unlikely to create any genuinely new sound – remember the wonderfully fresh synthesised sounds that excited us so much a few years ago? Where *is* the second generation of new tones?

Another drawback of modifying a sample is that many people will never understand the basic components from which sounds are constructed, leading to yet more stagnation in the originality department.

Then there's the topic of the good old LFO. How about discussing the suggestion that improved keyboards and lower prices are pointless unless you improve the degree of modulation available for the main sections *via* independent LFOs? A separate LFO for each PWM, filter and amplifier can alter the sound of a two-oscillator synth so dramatically that you can be fooled into thinking the sound comes from a mortgage-price system instead of an ancient modular set-up worth a few hundred quid.

Was that provocative enough? Come on, let's get some interesting dialogue going instead of a load of waffle about syncing this and that.

Martin Straw
Southampton

You said it...

Life, the Universe, and Everything

Dear E&MM,

Could you possibly let me know if you have circuit diagrams for the following: LinnDrum MkII, Emulator I and II, Greengate DS3 Polyphonic Digital Sound Sampler.

I will pay for photocopying costs if you could send me circuit diagrams.

Hoping to hear from you soon,

K Sing
Designer
Birmingham

The answer is 42 – now what's the question?

Retrofit Blues

Dear E&MM,

Being a regular reader, it seems to me that the only thing musical instrument manufacturers are concerned with is new instruments: in their eyes, our six-month-old keyboards are ready for the museum.

I recently traded in my Wurlitzer EP200 electric piano for the Yamaha PF15, with its 88-note, weighted action keyboard and excellent voices. And with the advent of MIDI, it seemed to me that the touch-sensitivity and piano-like feel of the PF's keyboard would be the ideal controller for the expanders that were starting to appear on the market. So I waited for a MIDI retrofit for my PF...and waited...and waited...

Why should I have to spend money on a synth with an inferior keyboard or, even worse, have to spend £1400 to £1600 for a 'Mother' keyboard, when I'm perfectly happy with the one I've got?

Please, manufacturers, give some thought to the instruments you've already produced, so that those of us who are not millionaires can make the most of what we already own.

Kevin R Miller
Newcastle Upon Tyne

Although most updates, MIDI or otherwise, come directly from the manufacturers, we're happy to report that an enterprising French company, Polytrak, have designed MIDI retrofits for the Yamaha PF10, PF15 and YP40 keyboards. Contact Polytrak at 29 Rue de Versoix, 01210 Ferney-Voltaire, France. ☎ France (50) 404816.

& EVENING NEWS,

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     Emery

MUSIC

Steve Joliffe and Ron Berry will be performing a concert of electronic music at the Carribean Association Hall, Woodbridge Road, Ipswich on Saturday May 4. Doors open at 8pm, and tickets are available at £3.00 from Tonal Productions, 226 Arkwright, Harlow, Essex (Students/UB40s £2.00 on the door). The Hall is five minutes' walk from the bus station and 30 minutes' walk from the rail station, but the concert will finish in time for members of the audience to catch the last London-bound train.

The 1985 Essex Rock and Pop Showcase is a non-profit making event organised by local musicians for the benefit of Essex-based amateur and semi-pro bands. All those eligible should contact Monkey Business at 6 Victoria Road, Romford, Essex RM1 2LA (☎ 0708 754548 or 25919) and make a date for the Cliffs Pavilion, Southend on Sea, on Sunday May 12.

A **Neuronium Fan Club** has been set up by E&MM reader Paul Walker. All enquiries should be addressed to Paul at 25 Ravenswood Avenue, Rock Ferry, Wirral, Merseyside L42 4NY.

'Man and Machine' is the title of a two-day festival of electronic music taking place at Stockton Parish Church, Stockton on Tees, on June 7-8. The festival will feature workshops, open rehearsals and trade stands demonstrating equipment. Ian Boddy will be presenting a concert of pieces from his new LP, *Spirits*, and talking about his music, while the final concert will include Tim Souster's new music theatre piece entitled *Work* for computer-operating pianist: this is probably the first piece of concert music written specifically for the Music 500 system. Ticket prices are very reasonable, and the organisers can book a local hotel room at £8.50 for the Friday night for those visiting the area. For further information contact: Jim Easton, Administrator, Man and Machine, Stockton Parish Church, High Street, Stockton on Tees, Cleveland (☎ 0642 611734 – office hours) or Douglas Docherty, Musical Director, Man and Machine, 14 Bewick Road, Gateshead, Tyne and Wear NE8 4DP (☎ 0632 784762 – evenings and weekends).

Also from the Stockton area comes news of a **Community Electronic Music Studio** which is currently being established. The studio will be offering rehearsal and recording facilities as well as practical educational courses dealing with specific synthesisers. Both MIDI and non-MIDI equipment will be available, together wih multitrack recording facilities which will be offered 'at sensible rates'. Anyone interested in using the studio should send an SAE to: Electro Arts, 4 Jacklin Walk, Eaglescliffe, Stockton, Cleveland TS16 9JH.

COMPUTER
MUSICIAN

MIDIC is the name of a new intelligent buffered interface that can be used between any MIDI device and any computer or terminal with an RS232 interface, working at most standard baud rates up to 38.4kBd. MIDIC adapts to the baud rate of its host computer, and allows MIDI application programs to be written in any language.

MIDIC 1.0 is the first released software product for MIDIC. This provides a variety of filters to screen unwanted information from the MIDI input. After-touch, Pitch-bend, Real Time and Active Sensing information may all be independently filtered, and Channel information may be limited to any selection of channels. Active Sensing clocks may also be generated automatically. An upgrade/replacement (entitled MIDIC 1.1) is due for release in April, and enhancements of the 1.0 system provisionally include a System Exclusive filter, a Tempo Generator and provision for 16 keyboard assignments.

Further details from : Hinton Instruments, 168 Abingdon Road, Oxford, OX1 4RA. ☎ 0865 72171.

STOP PRESS

The Programming People is the name of a brand new agency for synth, drum machine and sequencer programmers. At present they have 12 programmers on their books full-time and another 15-20 part-time, and the initial response has apparently been very encouraging. So if you want anything from a Synclavier to a DX7 programmed, or conversely, if you feel you have programming skills of your own to offer, then Debbie Kempson will be glad to hear from you on ☎ 01-229 0055. Overseas readers may be interested to know that Programming People are intending to become international in their coverage.

PPG UK have ceased trading as a result of reorganisation in the parent company. Information regarding a new UK distributor will be published as soon as it is known.

John Palmer informs us that he's received an excellent response to his letter (*Readers' Technical Directory*) published in E&MM December 84. He's now in the process of replying to everyone and organising a newsletter.

Brighton Home Recording Show – Norfolk Resort Hotel, Brighton, April 19-21. The Show will include lectures on Home Recording techniques and a free draw with a Tascam Porta One as Star Prize. Further details from: Dianne Lower, Tape Shop, 6 Preston Road, Brighton BN1 4OF. ☎ 0273 67281.

ROSE-MORRIS COMPETITION

Seen here is delighted competition winner Norman Fay (centre) being presented (somewhat belatedly, the competition was run in E&MM August and September 84) with his First Prize RK100+EX800 system by Korg UK's Rob Castle (right) and well-known session keyboardsman Brian Chatton (left).

Middlesex Polytechnic is offering an introductory course in electronic music-making at its third annual summer school, August 1 to August 28. To quote from their brochure, the course offers 'an opportunity to acquire skill in the use and application of sound recording techniques and the VCS3 synthesiser with keyboard attachment. The focus is particularly upon electronic sound and its manipulation, rather than the electronic processes which produce it.' If that sounds like your cup of tea, further details may be obtained from: Admissions Enquiries, Middlesex Polytechnic, 114 Chase Side, London N14 5PN. ☎ 01-886 6599.

Back Issues: Sadly, the following back issues of E&MM are now sold out: October 84, December 84, and January 85. Sorry, everybody.

THE ROGUES' GALLERY

Having fooled everybody into thinking we'd managed to cover the entire proceedings of February's Frankfurt Musik Messe in glorious colour in last month's E&MM, we were faced with something of a dilemma in the form of a whole host of photographs (most of them in good ol' monochrome) depicting new and previously unshown hi-tech music products.

What could we do with them? We couldn't write much about them because, in a lot of cases, the manufacturers' knowledge of the products in question was decidedly sketchy, so our own couldn't be expected to be 100% accurate. Yet on the other hand, to hold them back until their subject matter gradually became available in the shops seemed like a positively gloomy prospect – after

all, why not do a bit of readership-teasing when you've got all that scoop material at your disposal?

Then we had a brainwave. A brief Frankfurt Appendix of photographs and artist's impressions showing the gear which, for one reason or another, didn't get shown in E&MM March. So on your marks, get set, go...

An artist's impression of the PE101, said to be Roland's next introduction to their range of MIDI sound modules. This one's an ensemble machine majoring on strings and brass voices, but more than that, nobody knows...

Korg are thinking harder about their new gear than many. Their new MR16 MIDI Rhythm Sound Unit has PCM recordings of 19 different drum sounds built in, these being controllable from any MIDI sequencing source such as a computer software package or dedicated keyboard recorder. The idea is simple: why duplicate software when all the hardware can be controlled by the same stream of MIDI codes? Quite.

Klangwerk's Audio Operator computer-based musical instrument was certainly an unforgettable Frankfurt sight, – it should give the PPG system a run for its money when it gets into full production later in the year.
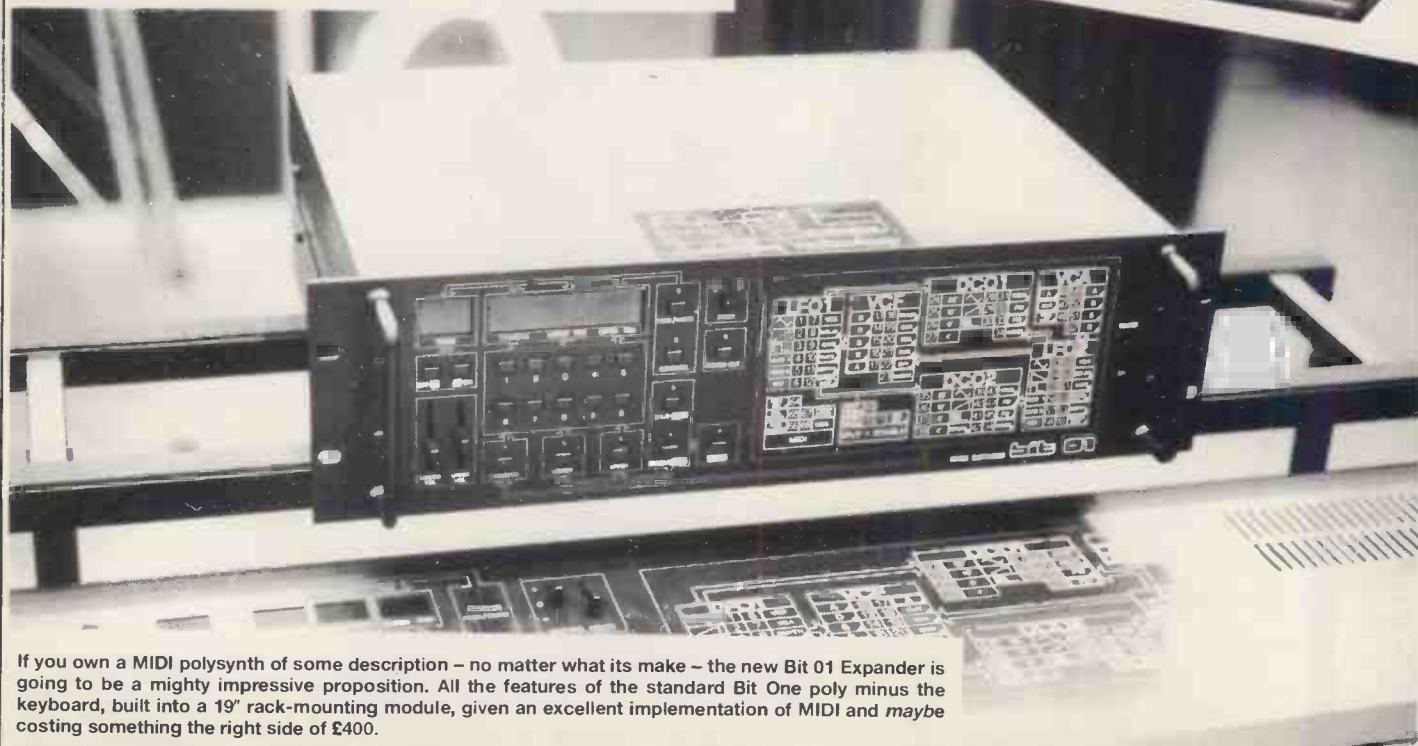
If you own a MIDI polysynth of some description – no matter what its make – the new Bit 01 Expander is going to be a mighty impressive proposition. All the features of the standard Bit One poly minus the keyboard, built into a 19" rack-mounting module, given an excellent implementation of MIDI and *maybe* costing something the right side of £400.

Quite apart from the S612 sampler we featured in colour next month, Akai now have an entire computer music system up and running in prototype form. Central to the new network is the rack-mounting CPZ1000 MIDI music computer, coupled with the RZ1000 Recorder Sync Operating Board and EZ1000 Edit Operating Board. The system isn't going to be easily affordable when it comes to this country later in '85, but it does show what the tape recorder company are capable of, if they put their minds to it. ▷

Another innocent-looking black box, the Boss MI10 is in fact a world first. Analogue-to-MIDI converters have been around a while now, but up until Frankfurt, no device had attempted to convert things the other way. The MI10 does just that. ◁

Roland's first electronic drum products won't be cheap – £2000 odd for a decent-size kit – but the sound quality of their digital samples is unimpeachable, and the pads themselves look superb.

11

Siel DK80
Dynamic Bitimbric Polysynth

Aimed fairly and squarely at the Poly 800s and Bit Ones of this world, the DK80 has a comprehensive paper specification that promises a great deal. Does its sound live up to that promise? *Trish McGrath*

Probably the biggest deliberation a synth manufacturer has to make over the character of a new product is exactly which features are financially feasible to incorporate, and which can safely be disregarded as unnecessary luxuries. Siel must have gone through much that sort of process when they drew up the spec for the new DK80, but at first glance, you'd be forgiven for thinking they'd decided to leave out nothing at all.

Let's face it, a bitimbric, 12-voice programmable synth with a dynamic keyboard, decent MIDI spec and onboard two-track sequencer sounds suspiciously like incredible value for money when it only costs £699. And with an Expander 80 due any day now at only £399, it really makes you wonder what the catch is. Where have Siel cut the corners? OK, you'll be happy to learn that the Italians haven't programmed the DK80 with a never-to-be-wiped-out sequenced rendition of 'O Sole Mio', thus saving all concerned a bit of money and a lot of aural torture. No problems there – anything more serious? Well, actually, there are a few compromises that start to spring to light as you spend more and more time with the new poly, but we'll come to them in good time.

Layout

One area Siel haven't compromised is the keyboard, which is of standard size (for people with normal-sized hands), and a full five octaves (C to C) of grey and white keys. (Yes, I know they're usually black and white, but when in doubt, ask your Art Editor: he said they were black, so we knew they *had* to be grey.) And although it betrays its plastic origins a little more than most in the under-a-grand price range, it's nonetheless quite pleasant to use.

Siel have also smartened their housestyle and livery from the Opera 6 and DK600 days, the DK80 coming encased in a modern dark grey plastic mould, with pale grey pushbuttons and parameter backgrounds lightly adorned with touches of yellow, red and green (*sounds lovely – Ed*). It doesn't weigh too much either, so those of you with back trouble should save on doctor's fees.

Sections, from left to right, are aptly named Modulations, Programming Unit, Edit, Sequencer, Masters and Cartridge, followed by the now familiar-looking Edit Map of parameter options attached to the resident DCOs, VCFs, and VCAs. An accompanying multi-function pedal unit (PD80) is an optional extra, while the external power supply unit is capable of supplying juice to two 80-series products, so you could also power an Expander 80 from the one grey box.

The back panel recess harbours the Power on/off switch, the (you'll never guess) MIDI In, Out and Thru connectors, and quarter-inch jack sockets for Sequencer Clock, Pedal, stereo headphones, and Mono/A and B audio outputs.

Siel have geared the 80 to accept both ROM and RAM packs *via* its cartridge slot, and the addition of these brings a total of up to 150 programs on line. The synth is equipped with 40 preset sounds (00 to 39) and a further 10 user patches may be stored on board. Frankly, it would have been infinitely preferable if Siel had given their new baby more in the way of internal RAM and allowed all 50 internal voices to be edited and over-written at will. As it is, you're stuck with 40 sounds which, one or two goodies apart, are really nothing special.

A optional ROM pack, meanwhile, brings another 100 factory preset sounds on-line in two banks, while a RAM pack gives you the chip power to save 50 of your own sounds: all cartridge sounds – whether from ROM or RAM – are accessed by program numbers 50-99 on the synth.

Programming

The Programming section on the DK's top panel provides the 'window' (actually, a two-digit LED) to the program, parameter or value being called up or edited, and the DK80 follows the Siel tradition of

the Enter button, whereby any program or parameter values have to be 'entered' before being processed.

It seems that, due largely to reasons of economy, it's no longer possible to edit synth sounds simply by selecting a patch number and twiddling a few knobs. As a thoroughly modern synth, the DK80 requires you to press Program, select two digits, press Enter, select Voice A or B, press Parameter, select two digits, press Enter, and then use the Up and Down cursor buttons to change the value. All a bit tedious, I reckon, especially when you consider that it takes all of 13 seconds to change the Cutoff frequency value from 00 to 99. This is Siel's first stab at giving a self-contained polysynth digital parameter access, and they've got nothing better out of the system than anyone else. Why doesn't everybody give it up as a bad job? Because it's cheap, I guess.

Features

Let's get down to brass tacks. The DK80 is a 'bitimbric' synth, which means you can layer two sounds simultaneously over the whole keyboard or, using the Split parameter, assign Voices A and B to opposite ends of the keyboard: they can even overlap in the centre. The 12 DCOs are split six apiece between Voices A and B, so with a split keyboard the synth is 12-note polyphonic. Siel have assigned each note of the keyboard, from left to right, a number from 00 to 61. Voice A can be introduced from the left side up (ie. 00 is off, 61 is full on), while Voice B creeps into the picture from the top of the keyboard down (ie. 61 is off, 00 is full on). So overlapping is easily accomplished, and you can set your split points anywhere you want: a good, flexible system.

Programming or editing sounds entails first of all selecting either Voice A or B to work with. Siel haven't provided an instant method of switching either voice out of harm's way while the other is being edited, and this is a shame, since it's often difficult to distinguish which Voice is making which racket – especially in Overlap mode. The quickest way to do things with the siuation as it stands is to select the Voice you *think* you don't want to hear, select the Volume parameter, and zero the value down. Shame.

The actual layout of parameters bears an uncanny resemblence to another well-known manufacturer's style, though Siel have managed to arrange the various sections in a different, and somewhat illogical, order. With a couple of exceptions, the parameter sections available to Voices A and B are identical, and comprise DCO, VCF, DEG VCA, DEG VCF, LFO1, LFO2, Chorus and Volume.

The DCO A-B section lets you set the Split and select the waveform (either Sawtooth, Square, or Off). If you go for sawtooth, the next parameter along lets you set it to either 4', 8' or 16' while square wave selectors have the next four parameters with which to set the volume of the square wave at each footage of 16', 8', 4' and 2' respectively. It seems pulse width modulation has gone out of fashion, more's the pity.

Voice B has all this plus a Detune facility for detuning the pitch from that of Voice A in semitone intervals (up to 11 semitones) or finely (up to a quarter-tone, for more subtle detuning effects). Noise of the pink variety can also be added to Voice B, and a touch of this can do wonders for creating 'breathy' acoustic-type sounds. With this species of sound, selecting Single Trigger (more on this later) and playing legato allows you to trigger the noise element only when it's needed – at the beginning of a phrase, say.

Both the VCA and VCF Digital Envelope Generator sections are, not surprisingly, similar in concept, with Attack, Decay, Break Point, Slope, Sustain, and
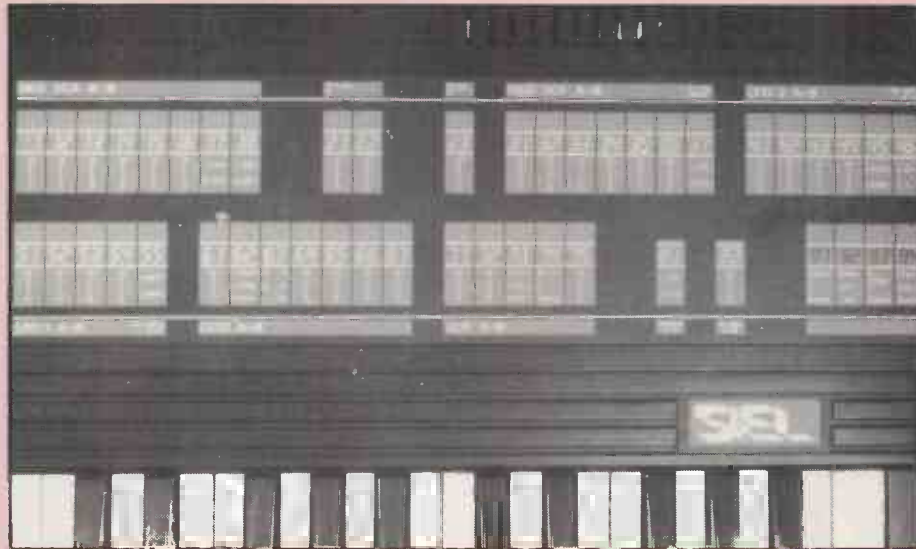
Release phases, all variable between values of 00 to 15. Nice to see Siel adopting Break Point (the level at which the Decay halts) and Slope (the rate at which the contour moves from Break Point to the Sustain level), because both make useful extensions to amount of control you have over the way the sound changes through time.

Dynamics can be switched on or off for both sets of DEGs, while the filter's dynamic content is dependent on the DEG Level parameter (ie. the effect the envelope has on the filter in the first place). So if you've set the DEG Level to zero, the envelope will have no effect on the filter, and dynamic control will be a somewhat redundant facility.

Siel are also marketing an optional double footpedal, one pedal of which acts as a damper (or sustain) device that allows the envelope cycle to continue to the Sustain phase even when the keys have been released. The second pedal is multi-functional, and can be set to step through programs, start and stop the sequencer (both in record and playback modes), or interrupt MIDI transmission (see later).

Filtering Noises

The VCF consists of Cutoff (for adjusting the cutoff frequency of the 24dB per octave low-pass filter), Resonance, Keyboard Tracking (either off, half, or full), Trigger and DEG Level.

Since the DK80 possesses only two

filters, one for each of Voices A and B, Trigger lets you choose between single and multiple triggering, a facility similar to Korg's way round the single filter on the Poly 800. Complete newcomers to these terms would do well to glance through this month's episode of *Back to Basics* for an easy-to-understand explanation. In the case of the DK80, Single mode means the filter's EG will trigger on the first note played, and all subsequent notes will follow the envelope of that first note. In other words, the second and subsequent notes played won't have the benefit of the DEG's full effect on the filter, and if the VCF's Sustain level is set low, they hardly even sound at all (this is very noticeable on presets 02 and 12, for instance).

One element the Poly 800's designers didn't try to contend with was dynamics, but Siel have (laudably) given the DK80 the power to vary the amplitude of a note depending on the velocity (or speed) at which the key is struck. That's all very well in theory, but in Single Triggering mode, the filter's dynamics feature can only affect the first note played – though since each note has its own VCA envelope, the VCA's dynamics operate unhindered. But you'd be amazed at how dull subsequent notes can sound once you've heard the first with dynamic timbre in all its glory...

Fortunately, you do have the option of selecting Multiple Triggering, in which case the VCF's DEG will fire every time a new note is struck.

The side-effects of both these triggering options vary from the negligible to the unbearable, depending on the construction of the sound you're using. So although it's nice to get the option of single triggering on a polysynth, the provision of only one filter per voice does reduce the feature's appeal.

Modulation

Well, the DK80 has no fewer than four LFOs, so no compromises here. The triangle LFO1, destined for the DCO section, consists of Frequency, Final Level, Initial Level, Delay Time, and Delay Mode (either Auto or Manual) parameters. For those used to simple speed

and depth variables, Initial Level is the depth of modulation obtained as soon as the key is struck, while Delay Time is the time it takes to proceed to the Final Level or depth. So, this system gives you the power to introduce vibrato and have it disappear after a prearranged delay time, or start off with none and introduce it after a delay, or even change its intensity between levels. That's certainly a pretty comprehensive range of modulation options for a synth in this price range and, indeed, LFO2 (modulating the VCF) goes one step further by offering a choice of either square or triangle waveforms.

And bear in mind that the LFOs belonging to Voices A and B are completely independent of each other, which means that, if you have the perseverance to undertake the required level of programming, you can have two layered sounds with different modulation effects on each of their oscillator and filter sections.

I must admit to being a bit dubious at first about the placement (at the top left-hand side of the front panel) of the Bend wheel and Depth button, but if you manoeuvre the wheel with your middle finger and press Depth with your index, the set-up works surprisingly well. The Bend wheel allows pitch-bend up or down of about a tone or so (not variable, unfortunately), while the Depth control introduces modulation when the LFO is in Auto mode. However, it's worth noting that Depth introduces only the LFO's Final Level – so if the Initial Level is set for vibrato effects and the Final Level's value is zero, pressing Depth actually *suppresses* the modulation effect. Just thought I'd mention it.

Masters & Mixing

Nobody will die of shock when I mention the fact that the DK80 has both Master Volume and Tune controls, though I found the 80 to be a 'quiet' keyboard, meaning the review amp needed a good crank up to give the synth any balls, even with the Siel's Master Volume at full.

One-finger-chords can be composed easily by selecting Hold, keying a few notes, and pressing the Chord button; and the data isn't wiped out when Chord mode is exited, either. If you select just one note, the keyboard becomes, in effect, monophonic.

The really odd thing about the Hold facility is that it doesn't allow you to select a chord that can't be executed by your own hands. This isn't the usual way of doing things: Hold mode on the Korg Polysix (for instance) lets you pick 'n' choose six notes over the whole keyboard range for its Chord Memory, so you can build up a chord over a wide span and take as long as you like over doing so. But the DK80, even in its so-called 'Hold' mode, starts from scratch again every time you key a note if no others are held down. So if you can't execute the chord with two hands, it can't be memorised as a chord. A bit daft, if you ask me.

On a more positive note, Chorus can be either on or off for each Voice (and is

very effective); Volume lets you mix the relative level of each voice; and Program Write can be enabled to allow edited or user programs to be stored in a suitable location.

As mentioned earlier, the DK80 comes complete with 40 resident sounds and 10 programmable patches. So what does it actually *sound* like? Well, I'd be surprised if many punters find all the presets instantly likeable: I know I didn't. Overall, the brass and string sounds come across best, with an assortment of fat analogue synth sounds and more delicate piano programs a close second.

Sequencer

And so to the DK80's built-in two-track real-time sequencer... A modest affair this, and very easy to get to grips with. Recording entails pressing Record 1 and 2 (to clear any memory, and you'll want to clear the demo piece, believe me), switching Metronome on or off, using the cursors to arrive at a suitable tempo, and pressing Start. Since recording commences from this point and loops continuously from whenever you press Stop, the footpedal comes in handy for precise control. The second sequence is recorded in a similar manner, and by selecting Play 1, you can use the first track as a guide.

The good news is that program changes can be recorded, and if you want a simple chordal backing track, you simply record a monophonic sequence and use the Chord facility on playback. You're also

> 'It's hard to argue with the DK80 when it's packed with so many worthwhile features and priced at such a wonderfully realistic level.'

free to play over the sequences if you haven't engaged all 12 DCOs in the process. And no indistinct metronome to play along to either – this one is *loud*.

Sequencer Clock offers three options for trigger playback: MIDI, External, and Internal. MIDI refers to the clock received from a MIDI drum machine or sequencer; External opens the gates to a gate pulse received *via* the Seq Clock input; while Internal is self-explanatory (and quantised to 24 clock beats per quarter note).

But have no illusions. This is no QX1. Drawbacks comprise continuous and compulsory looping on short sequences in which both tracks loop independently of each other, though if you run out of memory (about 300 notes), playback occurs without looping. There's also no capability for recording pitch-bend or depth info, though in fairness, you shouldn't really expect much more at this price level.

In simple Receive and Transmit operation, the DK80 can be set to Omni On (Poly) mode, or any specific MIDI channel from 01 to 15. Why only 15? Well, if you select a channel between 01 and 10 inclusive, Voices A and B communicate on separate channels. So for instance, selecting channel 01 sends A down channel 00 and B down channel 01 (or 01 and 02 respectively if you count from 01 to 16). Thus, with a couple of Expander modules set to receive different channels, you can obtain some really beefy sounds and use the DK's Keyboard Split mode to even greater effect. Meanwhile, if any of Channels 11 through to 15 are selected, the assigned channel is the same for both sections.

The sequencer adopts a similar principle of channel assignment, with Sequence 1 taking on the role of Voice A, and Sequence 2 that of Voice B. However, sequence data can only be sent to an external keyboard *or* be played internally – but not both at the same time. Mind you, any notes keyed on the Siel will still transmit over MIDI even in Seq External mode, assuming the Expander can take it. Best be careful when interrupting playback to another MIDI synth, though, as it's all too easy to chop the Note Off command and end up being lumbered with the notorious MIDI drone.

On the plus side, you can actually record the sequences using an external MIDI keyboard and play back using either the INT or EXT option. Other bonuses are that the Damper pedal used on the DK transmits *via* MIDI, though the Hold and Chord facility doesn't. Which is something even Siel's MK900 (primarily a 'domestic' instrument) *will* do, come to think of it...

Program Changes are well catered for too, and can be purely Internal, governed by the External keyboard, or be operated both ways.

Conclusions

Priced at such a wonderfully realistic level and packed with so many worthwhile features, it's hard to argue against the DK80. Why, it even looks stylish.

Headaches? Well, if there's a DK90 in Siel's R&D lab right now, I hope they add some way of merging voices from different patches. That way, a favourite bass sound, for example, could be quickly teamed up with a potentially complementary strings or brass sound from another patch. And how about a Unison feature for soloing?

But looking back, quibbles are few and unlikely to be of paramount importance. What *is* important is the quality of sounds produced by the DK80. It's in this ball-game that I used to feel Siel were in the Second Division. The DK80 could just be the synth that takes them one better...

RRPs are DK80, £699; ROM packs £36.50 each; RAM packs, £29.40 each; pedal unit, £36.50; power supply, £25.75. Further information from Siel (UK), Ahed Depot, Reigate Road, Hookwood, Horley, Surrey RH6 0AY. ☎ (02934) 76153/4

Pearl DRX1

Electronic Drum Kit

There's no sign of any decline in the market for acoustic drums, but some of the traditionalist manufacturers aren't taking any chances. Pearl are the latest such company to enter the electronic fray. *Paul White*

I f this year's Frankfurt Musik Messe was anything to go by, the music world has well and truly woken up to the fact that electronic drums are big business. Wandering aimlessly among the exhibits, it seemed to me as if every other company had some sort of electronic kit on display, and these ranged from the visually exotic to the sonically excruciating. Some were even both at once. Out of this veritable avalanche of new hardware came the Pearl DRX1, seemingly the sensible man's electronic drum system. It sounded good, it was neatly packaged, and it had obviously been designed with the particular needs of drummers in mind.

So, once our Teutonic sojourn was at an end, I lost no time in getting hold of a Pearl kit (the first in the UK, no less) for review. And I'm glad to say that, in the main, those favourable impressions have been reinforced by hands-on experience, and that it offers a combination of facilities you won't find on any other kit in its price range.

Features

The DRX1 is an entirely analogue kit, but it offers a choice of eight user-programmable kit sounds, which puts it

one up from other models that offer only factory presets plus one user sound per drum.

The design of the pads themselves owes a lot to the original Simmons concept (in looks, especially), but their construction is more of a novelty, as each drum is based round a rigid plastic moulding fitted with a circular, resilient rubber playing surface. Pearl T900W stands are supplied as standard with the DRX1, and these lock firmly into the pads to give the kit as a whole a very solid, secure feel.

The bass drum is free-standing and supported by a couple of pretty fearsome-looking spurs, and the pedal attaches to the unit by means of a substantial metal plate.

The heart of the Pearl system, as it is with any electronic drum set-up, is the sound-generating module. Pearl have put their sound circuitry inside a 2U-high, 19" rack-mountable case, which seems as good a housing as any I can think of. The unit's front panel incorporates controls for pad sensitivity and sound editing, while the rear panel contains separate outputs for each of the five drum channels, stereo and mono outputs, and a Memory Protect switch. This last-mentioned is usefully equipped with three

positions, one that makes the memory totally secure, a second that makes it totally available, and a third that makes only one location available. Additionally, the inclusion of two small six-pin connectors enables the system to be triggered externally, and provision is also made for the connection of a remote program change footswitch, though this isn't yet available and nobody at Pearl Music could shed any light on the subject of what form this will eventually take.

Controls

Considering the amount of control over sound-generating parameters the Pearl system offers, the module's front panel is remarkably uncluttered. The five sensitivity preset stubs are flanked on the right by the main display, which shows all eight variable parameters as well as the number of the program currently in use. An LED lights above the parameter being modified at any one time during editing, and a further indicator shows which of the drums is being affected by the current editing procedure.

The parameters that actually make up a drum sound are Pitch, Bend, Oscillator/Noise balance, Overtone, Attack, Filter Cutoff Frequency, Decay, and overall Level. Each of these is variable in 20 discreet steps, and I imagine most will be familiar to keen followers of all things electro-percussive. As far as I know, however, the Overtone section is unique to the DRX1: for the uninitiated, it's a form of modulation that affects the basic pitch and whose purpose in life is no more sinister than to help recreate a more natural drum sound.

If you want to set up your own sound, the procedure you have to follow is both simple and logical. First, you wish to edit using the number keys. Then you use the increment/decrement pads to select which pad you want to work on, and once you've arrived at this point, the parameter to be modified is located by a pair of similar pads, and the LED status display acts as a constant reminder of what you're doing and where, should you be of a forgetful nature. Parameter values themselves are set by (yes, you've guessed it) another pair of up/down buttons, and the current value is displayed by a two-digit LED readout.

In fact, the entire process is so simple

that it becomes second nature within a maximum of five minutes, which is probably just as well, as the DRX1 handbook was still in Japan when I reviewed the kit.

Oh, I nearly forgot. The front panel also incorporates a headphone outlet with its own level control for silent practice – though when was the last time you met a drummer that didn't want to be heard? – and, strangely, Line Input level control through which an auxiliary stereo signal can be mixed with the drum sounds.

Sound & Feel

Surprisingly – because they don't really look all that impressive – the resilient rubber heads felt more like real drum heads than those of any other electronic kit I can think of. The pads' velocity-sensitivity means they exhibit a wide dynamic range during performance, and like real drums, they proved more responsive near their centre. I was really impressed by the attention that had obviously been lavished on this area of design. It's a factor that's vital to the success of electronic kits in the eyes of traditional drummers, yet so many designers choose to give it a lot less thought than it deserves. So full marks to Pearl – who should know what they're talking about, after all – on this one.

Despite the fact that the factory-set sounds are among the most nausea-provoking sonic events I've yet had occasion to witness, it doesn't take much to coax some better-defined, 'classic' electronic drum sounds out of the DRX1. When you think about it, the ingredients which make up an acceptable electronic percussion sound have changed little

> 'Considering the amount of control over sound-generating parameters the Pearl system offers, the module's front panel is remarkably uncluttered.'

since Simmons pioneered the field, and what that means in musical terms is that rival companies have had plenty of time to perfect their imitations, sure in the knowledge that the sounds will still be fashionable when the design is in a marketable state.

To Pearl's credit, however, the addition of the Overtone parameter means you can create some pretty decent conga and bongo impersonations in addition to 'conventional' electronic sounds and outlandish (read 'unusable') synth drum effects.

Another point worth making is that stick noise remains at a low level with the Pearl design, so these drums could be used for quiet cabaret or ambient gigs – as well as silent practice, of course – without sounding like a Crompton steam-driven carpet loom.

Conclusions

Pearl are already established as a leading manufacturer of acoustic drum kits, and I see no reason why their first venture into the electronic arena shouldn't earn them a similar reputation. The DRX1 should be available in April, and at an RRP of £1020 inclusive of all HM Government's taxes, it'll be an attractive proposition for anybody seeking programmability in an electronic kit.

And in addition to that programmability, the Pearl kit has the plus points of excellent feel and dynamics, and nothing that I could honestly single out as being bad. There are a couple of omissions, and personally I'd have liked more memory locations for user patches and some sort of master volume control. But as we all know, it's a human trait to want more than you've got, isn't it?

∎

Further information on the DRX1 can be had from Pearl Music, 11 Garamonde Drive, Windbush, Milton Keynes, Bucks. ☎ *(0908) 564956.*

HI-TECH XPANSION

If you've achieved the implausible and exhausted the possibilities offered by the DX7 on its own, Yamaha can now supply you with the means both to make it sound better and to make it play your music automatically.
Dan Goldstein & Trish McGrath
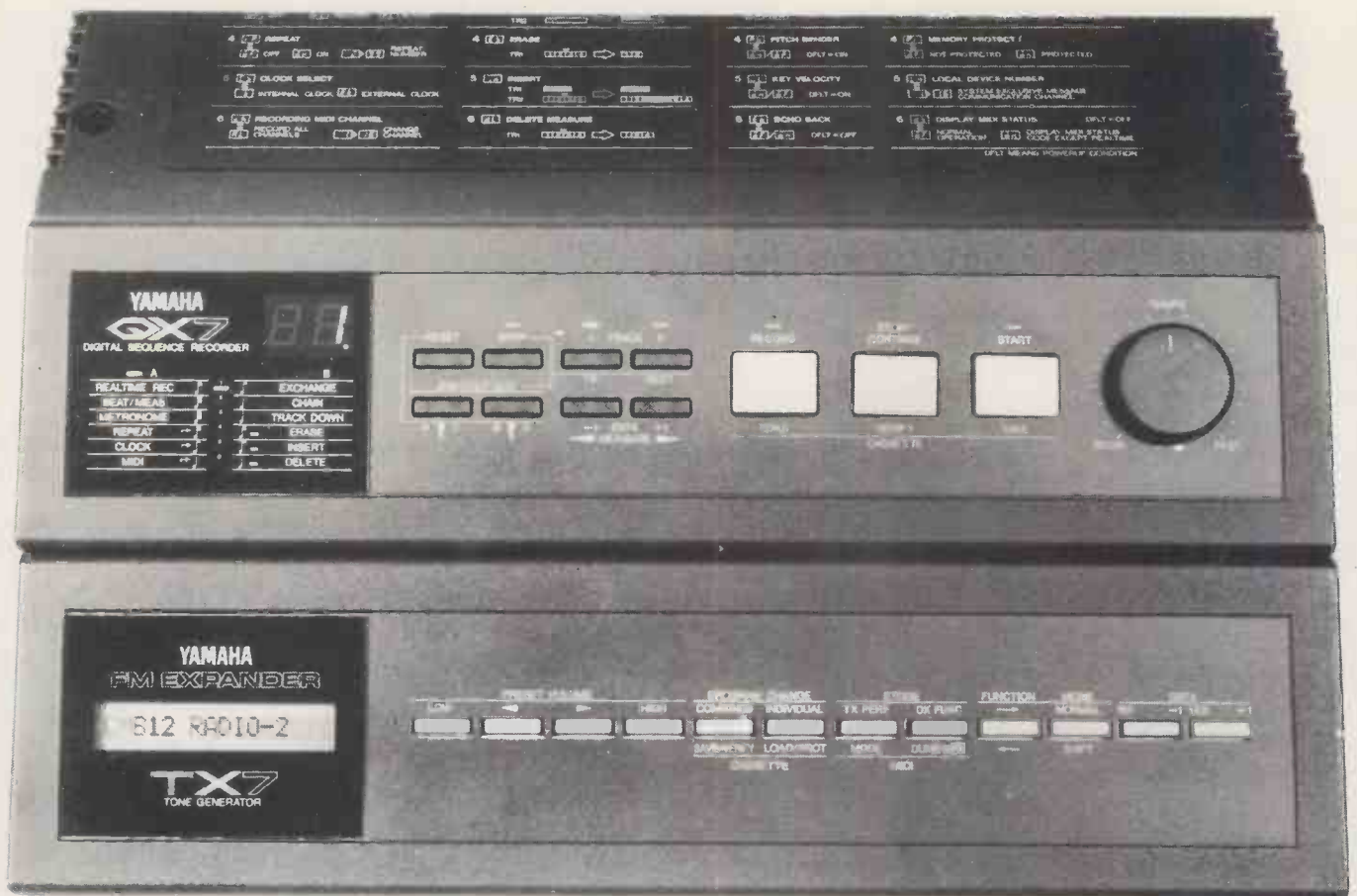
Yamaha TX7 FM MIDI Voice Expander
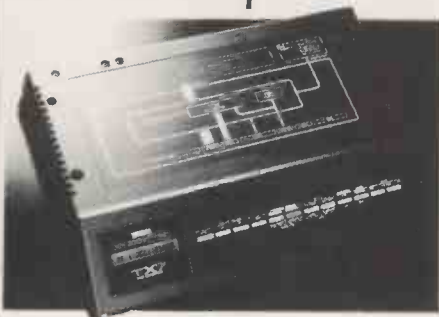
Now that world domination is the only outstanding item on the Yamaha DX7's List Of Things To Do Today, its creators are seeking some means of making sure those who've succumbed to the FM poly's undoubted charms don't get all over-confident and lose interest as a result. Hence a range of new goodies, unveiled by the company at

the beginning of the year, aimed fairly and squarely at the DX7 owner who thinks he knows it all.

Believe it or not, there are limits to what the DX7 can do. They're not rigidly definable or even easily visible, but they do exist, and once you've come up against them, it's all too easy to think that's all there is to FM Synthesis, Life, the Universe and Everything.

For a start, the 7 has no built-in sequencer, which is why Yamaha have introduced the QX7 digital keyboard recorder reviewed in the following pages. It also has a finite number of sound-generation possibilities. True, six Operators and 32 algorithms is a pretty flexible arrangement, but even that has its limitations, so Yamaha have come up with an elegant solution in the form of the TX7, a voice expander that duplicates the DX7's internal circuitry without burdening the end user with the cost of a second (unnecessary) controlling keyboard. Given that the new unit communicates with such keyboards *via* the ubiquitous MIDI, it should theoretically be possible for owners of compatible analogue syn-

ths to add it to their systems as a means of producing FM-type sounds, but the configuration of the TX7 is such that its most logical partner is a DX, as we'll see.

Specification

I could take the easy way out here and say no more than 'it's a DX7 in a box', but that would be unfair not only to Yamaha's designers but also to this magazine's devoted readership (thanks for everything, Jim), so I'll go into details.

The first point worth making is that the TX7 is remarkably small, light and neat, considering the amount of synthesis power it incorporates. Four circular indentations at the top of the box allow you to stack other TX7s (or a similarly-constructed QX7) on top without fear of the whole lot toppling over, but it's unfortunate that none of the DX synths have a suitably flat panel that would allow a selection of these add-on modules to be tidily and conveniently stored on stage, for instance. A rare production oversight on Yamaha's part, I fear.

And talking of oversights, DX9 owners are going to be left a little out in the cold

by the TX7, not because it's electronically incompatible (the two should link up fine), but because no printed diagrams of algorithm configurations (or indeed any other in-depth synthetic details) are in evidence on the TX. Which means you've got to be pretty familiar with the workings of a DX7 to get anything like the best out of its keyboardless brother: and I can't think there are too many DX9 owners who have that sort of working knowledge of a synth they don't possess.

What you get instead is a flow chart of how TX functions marry up with DX ones, but although this will undoubtedly be of value to first-time TX users, it is entirely obscured as soon as you stack up any additional modules in the above-mentioned fashion. Pity.

The TX7's front panel is laid out with even more economy than that of the DX7, which means fewer switches than ever before, and a corresponding plethora of functions for each switch. To their credit, Yamaha have made things as straight-forward as possible by colour-coding sets of functions in a familiar fashion, but my guess is that it'll take even DX devotees a while before the process of programming the TX becomes second nature.

Of the 12 identical green pushbuttons on offer, the two most crucial are pro-bably the Function and Mode switches. The former moves the editing process on by one parameter (the new parameter is then displayed, in somewhat abbreviated form, on the accompanying LCD), while the latter is used to instigate the switch array's second set of 'Shift' functions. And although the various modifiable parameters are accessed in a set sequential order, the Shift function does at least allow you to move through those parameters in either direction, using the Function key.

Parameter values are adjusted using Yamaha's now customary +1 and −1 data selectors (and as is usually the case, these double as Yes/No selectors), but in order to store any new values, you have to ensure the TX's Memory Protect circuit is off: better safe than sorry, I suppose.

Four switches to the left of the front panel allow you to program upper and lower preset levels for the TX's output as a whole – and there's also programmable attenuation for each of the voices you create with your Expander, which should save all that tedious mucking about adjusting levels between connected synth modules.

Connections

Rear panel features are confined to the Power switch, three MIDI sockets (In, Out and Thru, all on standard five-pin DIN connectors), an eight-pin socket for link-ing the TX7 with a mono cassette rec-order with a view to facilitating some means of data storage, and Line Out and Headphone jacks. The cassette interface is obviously slower and less convenient than competing disk-based storage sys-tems, but it has the clear advantage of

cost, especially so in the case of the TX7 because of its ability to dump 128 data files (each one comprising a complete set of 32 sound patches and a similar num-ber of function sets) on just one tape. So you shouldn't need too many C60s, all things considered.

The synergistic relationship between TX and DX is demonstrated by the extent to which the former uses MIDI System

> 'The TX7's front panel is laid out with even more economy than that of a DX, which means fewer switches than ever before.'

Exclusive commands to transfer data to and from a connected Yamaha poly. For example, Yamaha are making a big fuss over the fact that the TX7 stores a set of individual performance functions not only for each of its own sound patches, but also for those programmed into the connected DX. This usefully extends any DX's programming power (only the DX1 has this facility normally), because it means you don't have to adjust the list of parameters you've assigned to the synth's performance controllers (say) every time you change a voice.

The MIDI interconnection means you can also make program changes in one of two ways. Individual program change has effect only on the voices currently in residence inside the TX, while the Com-bined mode enables you to select a new

> 'Soundwise, the TX7 is a delight...but no machine that duplicates the DX7's sound-generating hardware has a right to be anything less.'

voice on both TX and DX simultaneously. So if you've got a glockenspiel sound assigned to Patch 15 on the TX and a vibraphone in the corresponding memory location on the DX, selecting that number

on either unit will call up both voices at once. This Combined mode will also come in useful for users of MIDI remote keyboards that have no sound-genera-ting circuitry of their own, as it will let you change TX voices remotely from the keyboard, leaving the module itself tucked safely out of the way.

Clever use of MIDI codes has resulted in a number of other refinements in the dialogue between TX7 and DX synth. For instance, an Edit Voice Out facility is built into the TX, and this transfers all the Expander's parameter-modifying codes to the controlling synthesiser so that said parameters can be varied remotely from the synth, assuming it's capable of de-coding the System Exclusive information.

There's even a facility that allows you to program the area of the keyboard over which the TX is operating. So if, for example, you've created a patch in which you want to hear the Expander's output over the bottom two octaves but the DX's output over the whole keyboard, you can program a split point at the appropriate place just for that patch.

Conclusions

It's neat, it's useful, it's well-designed and it's what the people want. All those things apply to the TX7, but the full story isn't quite so clear-cut.

For starters, its RRP of £699 doesn't exactly make it cheap. Attractive, maybe, but by no means a bargain: the removal of a five-octave, touch-sensitive keyboard, its associated controlling circuitry, and the case to go round it should have cut a bigger hole into the selling price of a DX7, make no mistake.

Soundwise, the TX is a delight. Then again, no machine that duplicates the DX7's sound-generating hardware has a right to be anything less. Fans of FM programming will revel in the extra pos-sibilities a TX opens up, but as I've already intimated, 16 characters of LCD don't provide the world's most compre-hensive information service, and working your way thoroughly through the TX maze is going to take a lot of perseverance. And if you've got an analogue poly from A N Other Manufacturer that you want to add FM sounds to, the TX7's internal con-figuration won't make your life any easier.

Its design lacks the spark of imagi-nation that would have taken it from the status of nicely inconspicuous add-on to that of possible Expander of the Year. But it'll still make a lot of people happy.
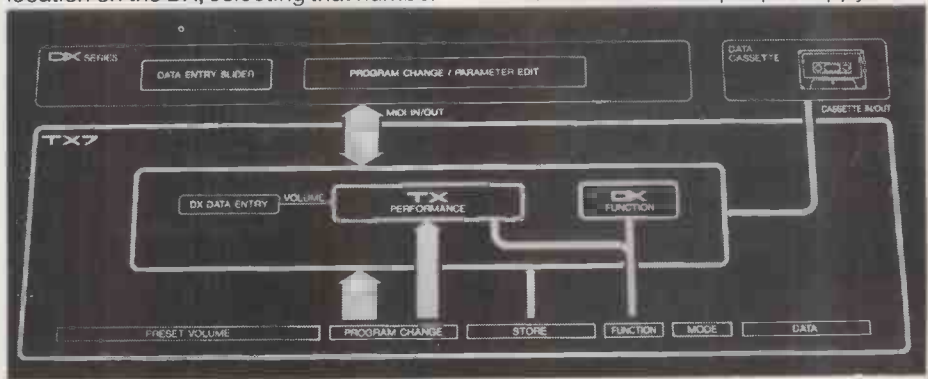
TX7's operational flowchart, neatly – if short-sightedly – printed on the machine's top panel.

Yamaha QX7 Digital Sequence Recorder

It was only a matter of time before Yamaha applied the proven data recording technology of their elaborate QX1 MIDI sequencer (still to be reviewed by E&MM pending Yamaha-Kemble having a spare sample to send to us) to a simpler and more readily affordable unit of more modest, but still extensive, capabilities. Such a device is the QX7, a dedicated sequencer that'll be of interest to most MIDI synth and drum machine owners, save those already bitten by the computer software bug. Yamaha CX5M users, for instance, will no doubt opt to save their pennies in anticipation of the company's forthcoming icon-driven four-track sequencer package.

Quite simply, the QX7 is a two-track digital sequencer capable of recording MIDI data polyphonically on all 16 MIDI channels and in either step or real time, with velocity, aftertouch, pitch-bend, modulation, foot control and even breath control parameter information intact. Overdubbing can be carried out by merging the machine's two audio tracks as many times as necessary up to a total of 16 simultaneous notes, using a wide range of editing features.

Designed to fit snugly on top of the TX7 FM Expander unit, the QX7's sleek black metal and plastic casing is decorated in traditional Yamaha hi-tech colours. The various editing functions are called 'Jobs' (as they are on the QX1), and are arranged in four banks labelled A to D. Yamaha have thoughtfully provided a comprehensive Job Guide on the top panel for reference purposes, while the front panel has a rundown of Jobs A and B, complete with accompanying LED indicators.

The remainder of the front panel controls comprise a two-digit LED display and a range of buttons with titles like Reset, Stop, Job/Step Size, Track 1, Track 2, Data Measure (-1 and +1), Record, Stop/Continue, and Start: there's also a large rotary Tempo control.

The sparsely-populated back panel offers the usual set of MIDI sockets (In, Out and Thru) as well as an eight-pin DIN socket for saving and loading sequences to and from cassette. And that's about it.

Real-time Recording

If you've got faith in your own ability to play music in something resembling the intended order and tempo, you go into real-time record mode, in which the QX7 will faithfully record exactly what you play to the remarkably fine resolution of 1/384th of a measure. Recording is always carried out on Track 1 (since only playback is possible on Track 2), and some Job conditions are worth setting before recording actually commences.

The Metronome function offers you a choice of hearing the metronome during recording and overdubbing or recording and playing: alternatively, you can have it permanently on or off. A flashing dot at the lower right-hand corner of the display indicates the chosen tempo, but to be honest, the metronome is a bit on the quiet side for comfort, so a MIDI drum machine will certainly prove handy in the keeping time department.

The Beat/Measure option lets you set a time signature from 1/4 to 16/4 and 1/8 to 16/8 inclusive (that should satisfy just about everybody), while accessing Clock Select gives you the option to choose

> 'The QX7's metronome is a bit on the quiet side for comfort, so a MIDI drum machine would certainly come in handy in the keeping time department.'

between internal or external MIDI clock. Moving on, Recording MIDI Channel lets you receive *and* record on any channel between 1 and 16, while Change Channel gives the QX7 the go-ahead to ignore the reception channel and assign incoming data to the channel number of your choice: handy if you've also got a DX7 and TX7, as the DX transmits on Channel 1 only. If you want the latter function to take place in real time, you simply select Echo Back – see later.

To economise on memory space (quoted as an approximate 8100 notes

without key velocity, 6000 notes with), Job C allows you to switch out any unwanted (and memory-intensive) functions such as polyphonic aftertouch, aftertouch, pitchbending, and key velocity. Incidentally, the first-mentioned applies to owners of instruments such as the Yamaha DX1 and Prophet T8, as it enables the QX to recognise aftertouch pertaining to each *individual* note played when the synth is in Mono mode (the facility needs a separate MIDI Channel for each voice in order to work properly).

Then there's the promised Echo Back feature which, when activated, transforms the QX's MIDI Out socket into a second MIDI Thru. But the difference between this MIDI Thru and the one next-door is that the data passing through it can be MIDI channel-shifted by the QX7's internal software. Mind you, it would have been nice if the QX7's designers had also given the existing MIDI Thru the ability to act as another MIDI Out. That way, you could have sequenced two MIDI synths (or synth and drum machine) together directly from the QX7. A small point, perhaps, but there'll be somebody out there cursing that omission, particularly as there's no logistical or financial excuse for it.

Once these recording functions have been fiddled with, selecting Track 1, Record and Start sets off the two-bar lead-in, after which recording can begin in earnest. No matter what you play, this process is always carried out in full bars, so if you stop recording in the middle of a bar, the QX7 will insert spaces until Measure End.

Step-time Recording

The new Yamaha will also happily record in step time, and available step sizes (ie. values of entered notes) range from halfnotes to hemi-demi-semi-quavers, and triplets of either crotchets, quavers or semi-quavers. And whereas the standard gate time is about 80% of the length of these notes, the QX7 also offers the means of entering crotchets, quavers and semiquavers with 100% gate times. In steptime mode, the Job lists on the front panel double up to indicate your chosen step size, and can be cycled through using the Step Size buttons.

You probably won't be flabbergasted to discover that pressing the Rest button inserts a rest equivalent in length to that of

QX7's Job Guide is an essential reference point for musicians experiencing difficulty getting the sequencer to do as it's told.

the step size, and only marginally more inspiring is the fact that notes can be prolonged by pressing the Tie button.

Although program change and key velocity information may be entered during step time recording, you can't – for obvious reasons – execute pitch-bend, aftertouch and control changes in this mode. But the really good news is that step- and real-time recording can be mixed at will, and the editing functions (see below) can be used to exploit this further.

Logically enough, pressing Start replays your recording from the beginning (or you can set playback to begin from a specific measure), while the Stop/Continue switch can be used to halt the playback operation temporarily and then restart it.

Editing Facilities

It's in the editing arena that the QX7 really scores. Because far from clinging to the belief that so long as you provide musicians with the means to record information and make mild alterations to it once it's been stored, they'll be happy, the QX7's designers have taken notice of the fact that today's keyboard player demands rather more in the way of compositional versatility, and given the sequencer a whole host of helpful and easy-to-use editing facilities. For instance, Quantisation (the means of correcting out-of-time notes) is offered to the nearest crotchet, quaver, or semiquaver and their respective triplet values. A factor of 1/32 would also have been a useful quantisation option, but apparently, the Japanese don't agree. If you're anxious to avoid committing any musical *faux pas*, you're best off setting the quantisation level to the lowest note value recorded, and once you've done that, pressing Start lets quantisation commence.

There's another nice feature incorporated within the QX7 that goes by the name of the Temporary Buffer. It's here that your original recorded data is automatically transferred as part of the quantisation process. Thus, if the results of the quantisation prove aesthetically unacceptable, the original data can be swopped intact back to Track 1.

In fact, the Save Temporary Buffer function can be selected at any point during recording or editing, and allows data on Track 1 to be saved within the buffer, while any data previously in the buffer is transferred simultaneously to Track 1. Which means a pattern that repeats itself many times during a song can be stored in the Temporary Buffer, retrieved whenever it's needed and chained to Track 2.

Overdubbing

Generally speaking, you do your first overdub by swopping the contents of Tracks 1 and 2 (using the Exchange function), and recording another sequence on Track 1. And once you've recorded this second part to your satisfaction, the QX7 offers a number of scintillating options. If you're happy with things the

way they are, you can commit your two tracks to posterity by activating the Memory Protect function. If you aren't, you can gamble what you've won and try for this month's Star Prize of multitrack musical satisfaction (no, not a ten-year subscription to *Electronic Soundmaker*).

If you've decided to go on, the Chain function joins Track 1 onto the end of Track 2, the result being transferred to Track 2 automatically. If you desire the reverse running order (ie. Track 1 before

> 'Once you've overdubbed the second part to your satisfaction, the QX7 offers a number of scintillating options.'

Track 2), you simply press Exhange before chaining. The QX7's most critical function revels in the name Trackdown, and merges the data on Track 1 with that on Track 2, transferring the result to Track 2.

If, by some mysterious quirk of fate, you make a mistake at some crucial stage of the recording process, you can clear the contents of Track 1 either in their entirety or from a specific measure using the Erase and Delete Measure functions. Conversely, the Insert option allows data from Track 1 to be inserted at a specified measure position on Track 2.

Among the QX7's less commonly used modes are Cassette mode (in which sequence data can be saved to and loaded from cassette tape, and the former procedure verified), and Local Device Number, which allows you to specify the MIDI channel used for System Exclusive data. Pressing Start in the latter mode sends the sequence data currently within a specified track to an external device along MIDI Out, assuming that said device is capable of decoding the System Exclusive information: another QX7 will do just fine...

Conclusions

There's no doubt that Yamaha have scored again with the QX7. It looks good, it's (relatively) easy to use, and it offers a lot more in the facilities-against-price battle than its nearest competitor in the dedicated sequencer market.

Its two-track format means you can create an overdub on a separate track, taking as much time and as many attempts as you like, and merging tracks together only when the new track is to your liking. In fact, an awful lot of the way the QX7 does its job adds weight to the impression that it's been designed to serve the musician, rather than acting as an unwanted source of operational distraction.

My only major criticism is that the QX7's designers haven't given it any more than one MIDI Out socket. Why not two, or even three? Is it because Yamaha are just about to introduce a new MIDI Thru Box? Your guess is as good as mine.

Speaking of new devices, Yamaha's YMC10 MIDI Converter should be just the job for anyone wishing to sync the QX7 to tape, while owners of non-MIDI drum machines can avail themselves of the Korg (shock, horror) KMS30, which'll convert the QX7's MIDI clock to the standard sync format as well as syncing everything to tape. Take your pick.

With so much sequencing software for both conventional home micros and Yamaha's own CX5M becoming available, it's nice to know that musicians unwilling to join the computer revolution have a feasible and cost-effective alternative. Because the QX7 is just that: there's nothing else to touch it. ∎

RRP of the TX7 is £699, while the QX7 retails at £499, both prices inclusive of VAT. Further details from Yamaha-Kemble, Mount Avenue, Bletchley, Milton Keynes, Bucks MK1 1JE. ☎ *(0908) 71771.*
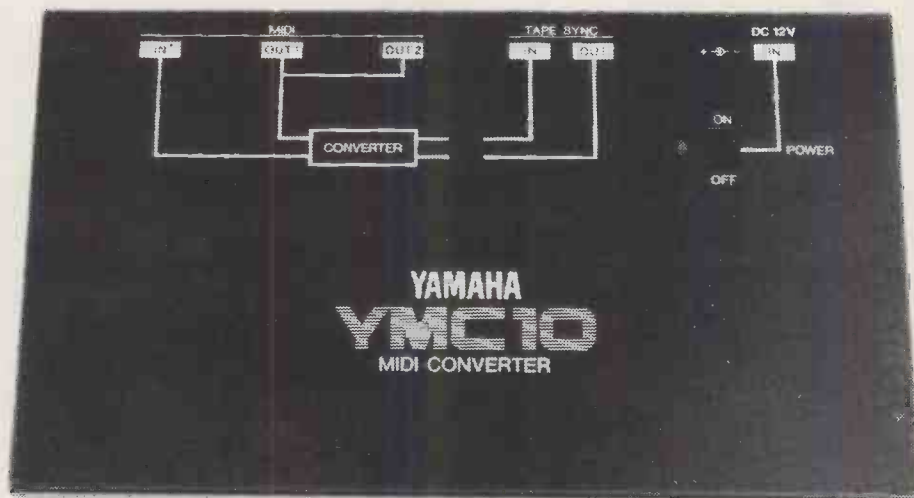
Yamaha's new YMC10 MIDI tape sync converter.

Linn 9000
Digital Drum Machine and Keyboard Recorder

When the LM1 made its appearance at the beginning of the decade, it heralded a revolution in the recording and performance of rock percussion. Now Linn have expanded the digital drum concept to control MIDI keyboards as well. *Paul Wiffen*

The brochure for the Linn 9000 has the following quote from Roger Linn himself splashed across its front page: 'The Linn 9000 is a dream I've had for a long time. It answers nearly all of the requests I've received from top recording artists over the years. To a far greater degree than has existed before, it allows the non-technical artist to quickly and accurately realize the music which exists in his mind'. Disregarding the Californian quasi-philosophy (the product of too much sunshine, surfing and mind-expanding medication), that's still a pretty far-reaching claim. Another piece of information we should probably impart at this point is that the UK price of the Linn 9000 is £4500 plus VAT. So Roger's

dream is not cheap. Does it represent value for money?

The first thing that strikes you is that the 9000 is a sizeable bit of kit, measuring 2' wide by 1' deep and over six inches high. It's also incredibly heavy at nearly 30lbs, due largely to its diecast metal case. This should make the machine extremely roadworthy, but it also means you can't pop it under your arm (*or in your handbag – Production Ed*) as you would an RX11 or a Drumtraks.

Pads

Quite apart from the fact that the new Linn's projected hardware updates (see later) are going to have to go somewhere,

another reason for its size is that it offers something larger than the standard 1" square drum voice selectors featured on most machines. Here we have significantly larger pads that should give you a considerably more than evens chance of hitting the right sound at the right point in your pattern. One unexpected feature is that the pads are contoured, but whilst this does throw up some interesting possibilities (like arranging groups of drum voices slightly angled towards each other for ease of playing), the actual implementation doesn't seem to be terribly logical. For instance, the machine's four Toms are positioned in a sensible circular pattern, but the pads themselves are angled outwards rather than towards each other. Daft, I'd have said.

However, the best thing about the pads is not their size or their inclination but the fact that they're both velocity and pressure sensitive. Now, there are probably more than a few drummers for whom these terms will be entirely new, but they're positively old hat to the keyboard player, so I won't embark on a 500-word lecture on what they mean. What's interesting in this case is the way they're applied to drum programming, as we'll see.

Because the pads are big enough for the user to have a fair go at, you can use the 9000's velocity-sensitivity to capture the instantaneous dynamics of live programming. So if your table-tapping is fairly accurate, then the programming style offered by the Linn is ideal. Autocorrection is available to eighth, 16th and 32nd note values and their triplet equivalents. Personally, I often tend to need quarter-note resolution when I've had a late night – it comes in useful for easy programming of ride cymbals and metronomes.

But there's a much more serious omission than that. Linn have followed the practice of almost every American manufacturer in offering no way to program in step time and no way to visualise the way programmed patterns appear musically: even the Roland TR707 offers both these facilities, and at an RRP of just £499 inclusive of VAT. It's all very well if you are a star American jazz-rock player with an impeccable sense of rhythm, but what about the poor guy who can't actually *play* the music which exists in his mind'?

Well, to be fair, there is talk of a software update that'll have provision for step-time programming, but I can't really

see how the 16 x 2 character LCD is going to give a decent graphic representation of the music, no matter how many updates there are.

Repeats

Mind you, there is one feature of the 9000 that I believe is totally unique: the way in which it allows you to apply pressure to the pads in conjunction with a Repeat button. This is how it works. You set the auto-correct to the time value you want, the repeats to play, for example, eight triplets, and then whenever you hold down the pad that relates to the sound you want (it works best with hi-hat) the machine automatically puts that sound on every eighth triplet. But the really great thing is that the level at which the voice is recorded varies depending on how hard you push the pad. The extra realism of hi-hat and ride cymbal parts this method imparts really has to be heard to be believed, and that's not all: holding down Repeat and moving quickly across the Toms results in great blistering tom fills *a là* Simon Phillips. Yes folks, you too can be a world-acclaimed session drummer at the touch of a button (or two). More than any other feature, this is the one that substantiates Roger Linn's

> 'A useful crossover from drum programming is the Work Loop, a bar or set of bars that goes round and round until you're happy with what you've recorded.'

claims. It's quick, accurate and very impressive.

But this is not the end of the authentic hi-hat story. In a second pass (that is, once the actual strikes have been recorded) you can add a separate decay either from a slider on the left-hand end of the panel or, for people who like to do things properly, from a footpedal plugged into one of the external footswitch sockets. The machine is intelligent enough to record a separate decay amount for each strike of the hi-hat, though this is in direct contrast to its lack of an up-to-date approach to tuning.

Wot, No Tuning?

Not strictly true, that. There is a static tuning facility on the new Linn, but it's accessed by the volume sliders when the Tune Drums button is pressed, which means that when you go back to live volume mixing, the knobs are in the wrong position. So, although you can set up one tuning level for each drum in each pattern, when you merge two patterns together (to make a song), the second one inevitably defaults to the tuning setting of the first pattern. This is pretty appalling for a machine in this category,

especially at a time when an SCI Drumtraks (RRP £999) allows you to set a different tuning level for each drum strike, and many others allow different tunings within each pattern.

If something like decay amount can be programmed for each note, then so can tuning: after all, it's merely a case of the unit memorising a change in the readout speed of the sample. But as things stand, the Linn 9000 can't do it, so let's hope the situation is speedily rectified by the first of the promised software updates.

Let's get back to good points. The Shuffle feature (we tend to call it Swing on this side of the Atlantic) is included in the 9000's Auto-Correct section, and if you've selected either eighth-note or sixteenth-note correction, you can choose to swing either the quavers or the semi-quavers. Which brings me to the point that Swing (or Shuffle) is probably one of the most misunderstood musical terms, probably because it comes from an area which is stronger on 'feel' than theory. But that doesn't prevent the 9000 from having the ability to swing at either the quaver or the semi-quaver level, which I think I'm right in saying is a world first. The amount is variable between 0 and 5 (which represents a range of between 50% and 70%), or a variation between two equal halves and a situation where the first half of the beat lasts twice as long as the second. This dual level of swing will be a great boon to jazz musicians, but most rock players won't even understand it, let alone use it.

Drum Sounds

If it's taken me longer than it should have to get onto the subject of what the Linn 9000 actually sounds like, the reason is that it's the first drum machine designed to make the changing of sounds a safer bet than pulling chips out and replacing them with new ones. On

the Linn, this process will be accomplished by loading sounds from cassette or floppy disk, or by sampling sounds directly (when the update becomes available).

In the meantime, those that are shelling out their five grand's worth will have to live with the 13 sounds provided. The promotional literature would have you believe this figure was 18, but the fact of the matter is that there's only one Tom sample, only one Ride sample and only one Crash sample: the extra pads for these sounds simply let you set up different tunings. Anyway, in the words of Ian Carmichael, the basic sounds are really jolly good. They've been recorded fairly straight, which is probably a good move as most studio engineers and producers prefer drum samples to be as dry as possible so that they can then be treated at will according to the musical context.

Specifically, the Bass Drum is good and meaty (though personally I'd have liked a little more click), the Snare is good and crisp (if a little anonymous), the Tom sample is simply excellent, and so are the Crash and Ride Cymbals. The same is true of the less commonly-used sounds – Congas, Claps, Cabasa and Tambourine.

There's a separate output for each panel-mounted drum pad in addition to a main stereo output, each drum has volume and pan position sliders, and there are even a couple of auxiliary inputs. This arrangement means the 9000 can be used equally well in both studio (where each drum is treated separately) and home (where all that's needed is a general impression of dynamics and stereo positioning) situations, though it must be said that the features themselves are scarcely revolutionary.

There are also two programmable trigger outs (to drive old-style sequencers or a sampling unit, say) and the built-in metronome click can also be taken out of ▷

a separate output socket. The cassette interface gives you a choice of line or mic level for the output, since signal level compatibility is one of the major headaches of dumping data to cassette. A welcome feature, that.

The remaining rear panel connections are three five-pin DIN sockets for MIDI In, Out and Thru. But they're not there to allow MIDI control of the Linn's drum sounds, at least not for the time being. Instead, they act as the means by which the 9000's built-in digital sequencer communicates with the outside world.

Keyboard Recorder

It's the advent of MIDI that's allowed Linn's engineers to expand their conception of a drum machine to encompass a digital sequencer, though until the step-time programming becomes available, 'keyboard recorder' is probably the more accurate description. For the time being, this side of the machine works in a fairly simplistic fashion not all that far removed from that of a multitrack (the Linn has no fewer than 32 of them) tape recorder.

You connect a suitably-equipped synth to MIDI In, set the Linn to Record, and play a series of notes to the drum part or the metronome click, which can be programmed to give anything from quarter-notes to 32nd triplets. These notes are then recorded with or without auto-correction, depending on your wishes.

What's really interesting is the way in which the 9000's percussion-oriented technology has given its keyboard recorder a couple of unusual – though highly useful – musical functions. For the first time, a Swing/Shuffle feature is available on a sequencer (long overdue, this), and another useful crossover from the drum programming ethos is the Work Loop (as Linn call it), a bar or set of bars that goes round and round until you're happy with what you've recorded. This gives you the chance to add in a few notes on each pass, something not previously available to keyboard players.

MIDI Control

MIDI signals can also be transposed and given program changes by the 9000, but most significantly, the new Linn is intelligent enough to change the MIDI channels of incoming data not just for replay, but for instant retransmission from the MIDI Out socket. This feature is known as MIDI Echo (the Yamaha QX7 calls it Echo Back), and makes overdubbing a lot of synth and expander tracks a very swift and trouble-free exercise – assuming you've got the playing ability. Once you've set each synth to separate channels (given that the manufacturers have seen fit to implement Poly mode and Channel Select), you can designate a master keyboard from which to play, yet still hear the part played (in both Record and Playback) on the synth and sound you want to use. The only possible hitch is that if there's no local Control Off on your master, you may have to resort to turning the volume down whilst recording.

Another welcome consequence of all this is that you can hear your parts autocorrected as you play them. This is a little weird at first, and can end up bringing on sloppiness as all your recordings come out so well (back to practising on the dummy keyboard, I'm afraid). However, you'll still need a certain amount of technical skill to get demi-semi-quaver triplets right (even with the auto-correct on), so non-technical artists will have to wait for the step-time software before their Linn 9000s will give them the sort of instant response to creativity the company are boasting of.

> 'When the SMPTE is ready, it'll be possible to match speed to Frames per Beat, and both European and American frame conventions will be catered for.'

Tempo and Sync

Tempo can be input as a number in Beats per Minute (with up to 0.1 resolution) or tapped in as crotchets on a button, to obtain either Record or Playback speeds. It's also possible to link sequences of different speeds together within songs, and future software will also allow speeding up and slowing down (accellerando and decellerando for the classical buffs out there) less instantaneously. When the SMPTE is ready, it will also be possible to match speed to Frames per Beat (up to ⅛ frame resolution), and we're told that both European (24/25-frame) and American (30-frame) conventions will be catered for. The standard sync code used by the 9000 is preset to 48 pulses per quarter note, and
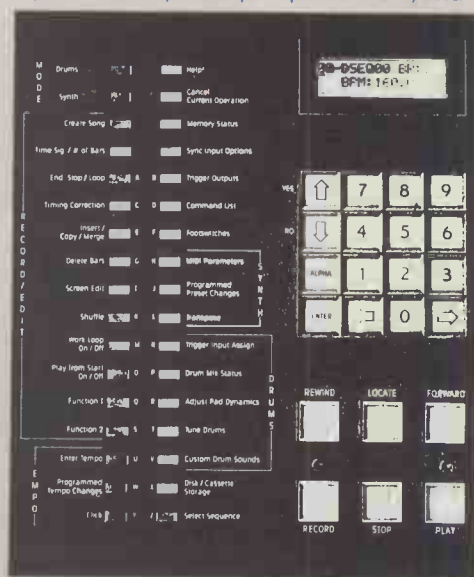
this can be output to and input from tape directly. Presumably, the standard sync sockets will also send and receive the SMPTE code when this update is available, to allow sequences to be dropped in halfway through without running sync

from the beginning of the track in question.

Future Updates

Apart from the various software updates I've already touched upon, there are several hardware updates said to be on the way. First, it'll eventually be possible to expand the existing 64K memory – which currently allows over 7000 MIDI notes and 24,000 drum notes to be stored – with 64K or 128K static RAM cards up to a total of 256K. Arithmetic was never a Wiffen strong point, but I reckon that should at least quadruple the amount of music the 9000 will hold before dumps become necessary.

Second, a 3.5" disk drive (they were all over February's Frankfurt show) will be available to ease the storage of both sequences and sounds, and this option will be a highly desirable addition to the third update, an audio input card that'll facilitate user-sampling and customising of sounds. This is the big one, chaps, and it'll be interesting to see who gets it on the market first, because E-mu's Drumulator II will have it fitted as standard when it becomes available in June.

A SMPTE reader/generator card will be available for studios equipped with the latest recording hardware, while those who want to play Linn's drum sounds from external triggers (eg. pads or sequencers) will soon be able to add one or two trigger cards, each with six assignable triggers.

Conclusions

While the 9000 in its present form introduces a number of welcome programming facilities to contemporary music's vocabulary, they don't offer enough in themselves to justify its dollar-inflated price tag.

Its basic concept is elegant enough: what could be better than a machine that alleviates all that tedious syncing between drum machines and sequencers by putting the two together in the one box? Only problem is, the 9000 has too many operational omissions to make it everyman's answer to electronic music recording. The lack of a step-time programming facility has to be the biggest failing, but as I've already discussed, there are a whole load of smaller shortfalls that really shouldn't exist on a machine that costs so much and comes from such a prestigious design and manufacturing stable.

I'm willing to concede that most of the problems I've mentioned will be cured as and when the promised updates become available, but bear in mind that even if the software additions are free to existing owners, the hardware ain't gonna be.

As it stands at the moment, the Linn 9000 is a dinosaur that'll need every update it can gets its hands on if it's not to become extinct.   ■

Further information from Linn's exclusive UK distributors, Syco Systems, 20 Conduit Place, London W2. ☎ 01-724 2451.

Zlatna Panega ACS100 & TCS100

Anticipation Sampler & Thought Control Add-on

As from this coming Spring, the acknowledged centres of the hi-tech music world are going to be under threat from a new Eastern European hardware co-operative. The company's first product is a revolutionary sound-sampler from Bulgaria, and E&MM has managed to get hold of the first production model for an exclusive review.

Dan Goldstein & Paul White

While there's no doubt that the established musical instrument manufacturers are doing a sterling job in maintaining the levels of innovation on the hi-tech side of the industry, it could also be argued that it's been some while since a totally new approach was adopted by anybody.

True, plenty of people are forever seeking new ways of achieving the usual musical objectives, but too often, those objectives have remained unquestioned, and few designers or engineers have sought to produce a music system with new goals as well as new methods of approaching them. Because no matter how revolutionary their operating principles, the DX7 is still just another polysynth, the Music 500 a computer add-on no different in conception from a hundred others, and the LinnDrum simply an attempt at confining the role of drummers to playing *Space Invaders* in the games room.

Thankfully, this situation looks like being changed, thanks to some timely technological intervention from an unexpected quarter. It's been known for some while that electronics engineers in Eastern European countries have been working on a variety of musical projects, but until the beginning of 1985, it seemed unlikely that their work would ever escape the confines of the laboratories. Then, tucked away in a corner of one of the Frankfurt Music Fair's gargantuan halls, E&MM's illustrious Publisher came

across a stand taken by Germany's importer of Balkan musical instruments. Now, most of the gear on the stand was of the traditional balalika-and-panflute variety, but the Germans had booked a soundproof booth at the rear of the stand in which they were demonstrating the abilities of a revolutionary Bulgarian sound-sampling device and showing a number of interesting-looking peripherals.

'Have you seen the Transbalkan stand?' quizzed said Publisher when Music Maker staffers next congregated at the Press Bar. 'They've got a f***ing amazing sampler in there – looks great, and it's gonna be under two hundred quid!'

So off we went, armed only with a hurriedly-prepared A4 factsheet and five words of Serbo-Croat between us. And for the first time in many years, it seemed our Publisher was actually telling the truth. For there, lurking in the Musik Messe's uncharted backwaters and housed in an unpromising gunmetal grey 19″ rack-mounting case, lay the Zlatna Panega ACS100, besieged by a number of worried-looking Oriental gentlemen in navy blue and orange sweatshirts.

The pitch of the ACS was being controlled *via* MIDI by a Yamaha DX1, but the Bulgarian demonstrator, Louda Yana, was using none of that keyboard's sound-generating circuitry in his performance. Every sound we heard was, in fact, generated from within this appar-

ently innocuous little grey box, and we were intrigued as to what might be going on inside the unit.

Background

The only spokesman of a technical bent on the stand turned out to be Professor Gerganin Izvor, a systems analyst who for the last three years has devoted his attention to the development of this revolutionary electronic sound system. In what can only be described as the Bulgarian equivalent of a Geordie accent, he explained that the ACS only came about because somebody at the Central Office of Technology in Sofia thought that lessons learned by the Bulgarian State Railways in the course of updating signalling equipment could equally be applied to the hardware used by local folk musicians.

But Izvor wasn't satisfied. He realised that the potential of such technology would never be fully realised while its range of applications was limited to the music-playing population of the Balkans, and deliberately set about designing a product that was just too complicated for most of the locals to appreciate.

And luckily for him, The Bulgarian People's Export Committee approved of his aims and gave him a grant to enable him to continue his work. Thus, two years after the project had commenced, Professor Izvor unveiled his finished ACS100 towards the end of 1984. As things turned out, it proved to be a technological revolution – and not just for the Eastern Bloc.

Technology

Using negative-ion implant IC technology, Izvor had developed an entirely new concept in sound synthesis. He called it Phase Reversal Autocorrelation, or PRA for short. For the technically-minded, this principle works by paying particular attention to the fact that nothing ever occurs without something causing it to occur. That may sound obvious,

but Izvor reasoned that by using ultra-high speed logic circuitry to examine spatial and temporal minutiae, it should theoretically be possible to predict the outcome of the interaction of a finite number of parameters before that outcome actually takes place in real time.

Izvor and his growing team of collaborators encountered no small amount of difficulty in putting this principle into solid state practice, but eventually discovered that if the interacting parameters exist *in software*, the extrapolation becomes both more viable and more accurate.

The musical outcome of all this is that pre-echoes can actually be generated in real time, and by selecting the pre-delay time in accordance with an algorithm embodying the concepts of advanced Fourier synthesis and Bessel transforms (and developed by Professor Izvor himself), it's possible to create multi-dimensional comb filtering effects, which in turn enables a homogenous ambisonic sound field to be created from a single-point source. And because this circuit works on the principle of Phase Reversal Autocorrelation, that source need not actually exist at the time of musical performance, although in order to get the system to function correctly, the user must sincerely undertake in writing to acquire one at the Bulgarian Institute of Creative Art in Plovdiv.

Incidentally, and frankly this was where our credence became somewhat over-stretched, it appears that Izvor has since had to return to Bulgaria to face charges of Pre-Emptive Embezzlement after attempting to predict future football results using several ACS100s connected together in series.

Further Development

So, there we were at Frankfurt, utterly agog at what Professor Izvor appeared to have succeeded in creating. For a few uncertain and terrifying moments, all our technical and musical knowledge seemed to pale into insignificance by comparison with the genius that was this tiny, undernourished Bulgarian scientist. Fortunately, Izvor was a man of humility, and, realising that the concepts he was discussing were way above our editorial heads, he went on to discuss a new subject about which we thought we knew everything – MIDI.

It seems that the redoubtable Professor considered MIDI technology to be a spent force not worthy of inclusion in his magnificent musical creation, and that as a result, the first production ACS100s could be pitch-controlled only from the keyboard of an obscure (and somewhat outmoded) Russian mainframe-based electronic harpsichord.

However, pressure from the Romanian Central Bureau of Export and Commerce forced him to reconsider. And although Izvor himself remained unconvinced as to the merits of the hybrid interfacing system, he did at least deign to commission Mikhail Beecherescu, the Romanian importer of Kawai products, to design a suitable MIDI interface for the ACS.

Beecherescu's unparalleled working knowledge of MIDI (acquired largely through retrofitting the interface to obsolete Kawai organs and Teisco mono-synths) enabled him to complete the necessary work in time for the first public demonstration of the ACS100 in Frankfurt.

But there were problems. Izvor's reservations concerning MIDI's viability in such a technologically-elevated arena were to some extent justified by the fact that, when several MIDI-compatible ACS100s were connected together, the last machine in the chain invariably anticipated what form of data was going to be sent to it and started playing a fraction of a second before all the others. And the more machines the Balkans connected together, the more acute this potentially rather embarrassing phenomenon became. Which is why only one MIDI interconnection was attempted at Frankfurt, in spite of protests from the Beecherescu contingent.

Testing

Upon returning to Cambridge after our Teutonic sojourn, we were a little doubtful as to whether or not an ACS100 would actually turn up for review purposes. An intensely busy Izvor had been eager to promise us an exclusive review sample,
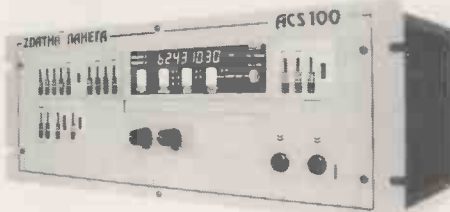
especially as none of the other UK magazines had even bothered to visit the Transbalkan stand, let alone express any interest in the ACS. At first, it seemed our worst fears had been confirmed. No new packages arrived from Bulgaria, Romania or any other nation situated behind the ferrous window-covering material, and we were forced to come to the conclusion that the whole thing had been an elaborate hoax.

However, it was during an Indian Restaurant conversation with the Editor of *Organist Today*, Malcolm Harrison, that we realised the machine had anticipated our desire to review it and arrived at the Music Maker offices shortly before we all left for Frankfurt. Between mouthfuls of succulent chicken pieces marinated in piquant nitromors sauce, the bearded home keyboardist told us of a strange rack-mounting box he had taken home to Reading one day, mistaking it for a Wersi organ kit. He was at a loss to explain the machine's uncanny ability to play Bulgarian folk favourites whenever he touched any of its front panel controls, but we were already familiar with the technology that was at work.

It was another fortnight before E&MM's Production Editor found a small black box on the banks of the River Cam while feeding the ducks one lunchtime, and this transpired to be the production model of one of the prototype peripherals displayed – but not demonstrated – at Frankfurt. Preliminary investigation revealed that said box was the eagerly-awaited Zlatna Panega TCS100 thought-control biofeedback add-on, but it was some while before we realised the full potential of this even more innocuous-looking unit.

Thought Control

It has long been the dream of the musician to produce music by thought alone without the physical restraints imposed by a keyboard or some other mechanical controller, and although the TCS100 add-on is far from perfect in this respect, it is capable of following fairly complex mentally-generated melodies with uncanny accuracy, and with any voicing you have the imagination to dream up. However, the device's built-in quantisation mode should be employed if you don't have perfect pitch.

Of course, the machine can't pick up human thoughts directly, but by using a pair of hand-grip electrodes and a light-weight headband, minute changes in skin resistance and alpha wave patterns are converted into a 16-bit binary code for subsequent digital filtering and processing within the TCS100 itself.

We found that pitch-tracking could be improved by sitting in a bath of tepid saline solution during performance, but this is not recommended for reasons of both electrical safety and artistic credibility.

In practice, almost everybody at the Music Maker offices succeeded in getting a reasonable musical output from the thought-control device, with the exception of the Publisher, who managed only a disjointed gurgle interspersed with impromptu arrangements of Glenn Miller classics.

Conclusions

The ACS100 is a remarkable machine at any price, but the fact that it retails for less than the VAT on the latest Fairlight software update makes it a thoroughly irresistible investment. The only possible problem we can foresee is that of spare parts, but as the machine can foresee this also, it is capable of ordering replacements by modem before the failure actually occurs. Which means the system can be up and running again before you even realise it's gone wrong.

On its own, the ACS100 is a revelation, and should prove of immense musical value to anyone currently forced into working with altogether inferior MIDI-based hardware. Yet it's in conjunction with the thought-control interface that things become really exciting: just think, now even the family pet can compose music. The only problem could be getting the headset to stay on the tortoise... ■

The Zlatna Panega ACS100 retails at £199.99 including VAT, while the TCS100 is a further £99.99. More information from The Body Shop at Carcass Music, above the Abattoir, Brick Lane, London E1. Regular deliveries are expected to begin around April 1.

Delta SX301

Add-on for Digital Delay Lines

As the world and his wife run round the latest generation of digital delays equipped with sampling facilities, a tiny British company comes up with a device that gives older machines the same facility. *Paul White*

The ability to trigger sampled sounds stored in a digital delay unit is a useful asset to just about anyone involved in the production and recording of modern music, and both Korg and Roland are now producing affordable products to fulfill this requirement (other companies will be following suit in the course of 1985). However, there are a great many digital delays already in circulation that don't sport this desirable feature, and this is where the Delta SX301 add-on unit comes into the picture.

The package includes a remote control box that plugs into the back of your DDL *via* a nine-way delta connector but, as internal modification to the delay line circuitry is involved, you'll have to live without your unit for some 10 to 14 days while the job is done.

As the photo of a modified Powertran delay line shows, a small PCB is mounted inside the unit, and this is in turn connected to the rest of the circuitry by way of another multipin connector. Apart from the Powertran, the system works with the Roland SDE series of products, the Boss DE200, the Yamaha D1500 and the Korg SDD3000. Models from Ibanez and Cutec have also been successfully modified, and if your model isn't listed here, just give the manufacturers a call: chances are they'll find some way of modifying it.

The SX301 contains a sound-operated trigger circuit that enables, say, a percussive sound to be loaded accurately into the delay line without your having to worry about pushing buttons in sync with the sound. Once safely stored, the sound may be coaxed into life simply by pressing the Play pushbutton on the remote control unit, or by feeding a positive-going trigger pulse into the 2.5mm jack socket on the controller.

Operation

Using the Delta unit is quite straightforward. Once you've connected everything up, select the longest delay setting on your machine and set the Feedback to zero. If the sound you want to sample is fairly short, it's advisable to set the Fine Delay control to give the shortest sample length, as this will give improved sound quality and reduce the amount of (noisy) silence that follows the sound.

The next step is to switch to Sample and press the Record button. The circuitry is now in a state of attentive waiting and will trigger as soon as the input to the DDL exceeds the preset triggering threshold. The sound is then stored ready for use, and if your machine has a Fine Delay control, you can use it to tune the pitch of the sample. At this point, you can trigger the stored sound manually using the Play button or (and I imagine this'll be the more popular course of action) use a drum machine trigger output or similar to run the system.

And that's about it, really. The sound quality of the sample is as good as that of your delay line, and the system posed no operational problems when it was in my hand.

But we haven't reached the end of the story. Realising the huge demand that currently exists among musicians for pitched sampling, Delta are about to release a new unit by the name of SX303. This is very similar to the SX301, but has the addition of a CV control input that allows the pitch of the sample to be ▷
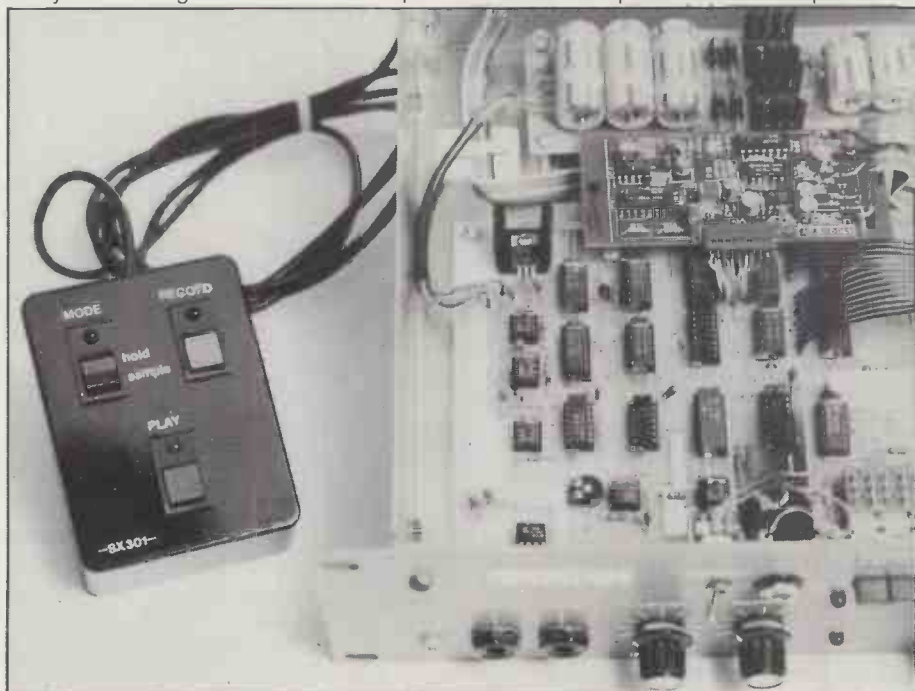
SX301 remote control unit alongside modified Powertran DDL.

controlled (over a two-octave range) from the keyboard of a synth with CV and gate outputs.

In the Play mode, you can set the 303 to play through to the end of the sample or to cut off when the key is released, which makes for a surprising (at this price level) degree of flexibility. And once your sample has been stored, it may be tuned over a one-octave range, so just about any sound can be converted into music of some description.

Conclusions

It's clear to me that these add-on units provide not only the means of ensuring your DDL doesn't become obsolete, but also just about the most cost-effective way of getting into high-quality sampling, albeit at a fairly basic manipulative level. The ability to trigger off-the-wall and 'found' percussive sounds is something an awful lot of drum machine owners would give an awful lot for, and the SX301 would seem to be an excellent way of extending the range of sounds at your disposal. In fact, it seems many pro recording studios have already had their machines updated for this very purpose.

For my money though, the SX303 is the better bet, as the possibility of playing samples chromatically from a keyboard is something well worth having. The only doubt you could be justified in harbouring would be the two-octave pitch range, but in practice this is more than adequate in most musical applications, and opting for a more expensive machine that offers a wider range won't guarantee you acceptable sound quality once you deviate too far from the original sample pitch. ∎

RRPs are £89 for the SX301 and £129 for the SX303, inclusive of VAT and fitting: a negotiable extra call-out charge is made for on-the-spot modifications. Further information from Future Music, 10 Baddow Road, Chelmsford, Essex, ☎ (0245) 352490, or directly from the manufacturer, Audio Engineering Services, 'Chelmer', Woodham Walter, Maldon, Essex CM9 6RZ. ☎ (0245) 412641.

Prototype SX303 being used in conjunction with Roland SDE3000 delay and System 100M controlling keyboard.

MUSIC

ON CASSETTE

In which another set of E&MM readers' demo tapes is examined, criticised and generally ripped to shreds... *Neville Unwin*

Teknik
No address supplied

It's not often a Fairlight finds its way onto an E&MM demo cassette, but the first track on Teknik's tape features one, and to good effect. 'Shhh...' is an instrumental piece that explores a wide variety of sound textures, merged skilfully together in a clean arrangement and recorded on an unspecified electronic music course (hence the availability of the Fairlight, presumably). Meanwhile, 'Redemption City' makes use of a vocoder amongst other instruments, and here too a wide range of equipment is used to produce a host of different sounds, without the piece itself being pretentiously written. The same can be said of the last track, 'Fantasy', which is effective in spite of some fairly simple harmonies.

Teknik's Rosina Cousins.

Tom McEwan
Gateshead

Recorded live in Paris, all McEwan's songs are guitar-based and have a heavy folk bias, and even if the style differs somewhat from that of the majority of readers' tapes, there's no denying this demo is one of the most entertaining in the April batch. The· songs themselves range in mood from the melancholy ('Breaking Hearts') to the raucous ('Booze Booze Booze'), all are delivered with humour, wit and style, and performed using nothing more sophisticated than a guitar and a drum machine.

The Jazz Masters
Merseyside

If standards of playing, singing and recording technique were the only criteria we applied when judging the worth of demo tapes, this one would be fortunate to get a cursory mention, but the originality of the band's music itself shows through in spite of failings in those areas. The vocals possess a hard edge that cuts through the murk of repetitive riffs and sparse though continually changing accompaniment. The longer tracks such as 'Has:Is:Will' are written in what I can only describe as an avant garde classical style, while 'Just a Doggone Minute' is nearly a blues, with vocals that owe something to Lou Reed and lyrics whose intelligibility might be aided by the consumption of certain illegal substances. Definitely a strange one.

Busy Busy World
London

Though hardly revolutionary in their musical outlook, BBW have a kind of easy-going appeal and songs with a melodic structure that's undeniably attractive. They're·basically a duo comprising Russell Taylor and Steve Cooke, but that line-up is augmented on this demo by a large number of friends playing a variety of different instruments. Cooke's vocals have a refreshing versatility and blend well with the instrumental backing, and both these are complemented by a high-quality recording. Mind you, I can't help feeling there's an urgent need for more acoustic drumming to liven up rather unimaginatively-used drum machine, the sound and metronomic beat of which lend the tape an air of resigned predictability.

Stratis *West Germany*

Each of the tracks on this tape has something slightly different to offer: some are highly accessible dance tracks while others are angry, though still stridently rhythmic, outbursts. Changes in instrumentation and texture abound, and you'll have trouble finding two verses that are musically identical. Repetition doesn't seem to be in Stratis' vocabulary. Synths are very much the dominant instruments here, and they're played well, with plenty of regard to the way they influence the sound of the group as a whole.

The Best of the Rest

From Australia comes a tape from **Russian Caravan, I Can Jump Gravestones** and **Helen Back,** who describe themselves as 'a three-band collective from Brisbane'. Once you've listened to the tape though, you could be forgiven for thinking that all the songs were by the same band, as they all incorporate a sparse arrangement that puts female lead vocals very much in the spotlight. The music follows similar patterns throughout, and although there's an occasional verbal sparkle, most of the lyrics fall happily into the cliché category. Pity. More light-hearted – and much the better for it – are **Quosh E,** a group of friends who, on the evidence of this tape, should seriously consider making their musical association a more permanent one. Quality of the vocals is inconsistent, and the band also have a tendancy to rely too heavily on the more technical aspects of arrangement·at the expense of the music itself, but in spite of all that, the songs really work. Oddest cassette of the month must be the **JA to GB Project.** This represents a search for the original style of Jamaican reggae (as opposed to the commercialised variety) and as such is largely concerned with an 'expression of violence in the West and East suburbs of Kingston, Jamaica'. The beat is simpler than today's reggae rhythms, but continues unabated through the social turmoil of the lyrics, producing a decidedly unsettling effect that's heightened by an almost Indian feel to some of the melody. Underneath an almost impenetrable layer of tape hiss lies an inventive if straightforward set of songs from one **Chris Gee.** Although he denies the fact, there's no doubt the songs are blues-influenced in style if not in harmony. Battling against a chronic shortage of equipment, Gee succeeds in giving each song a distinctive character of its own, thanks in part to some resourceful drum machine programming. Finally, **Syndrome** are another act whose demo's chances of success are severely impaired by duff sound quality. The music just about makes it through: gentle, ethereal synths forming a novel background to wailing vocals, with a refreshing lack of drum machine addiction. In fact, Syndrome's only musical problem is an all-too-obvious desire to be constantly inventive, though I guess that's better than being tied to convention. ∎

If you've made a demo tape you'd like us to hear, send it securely wrapped (accidents do happen) to On Cassette, E&MM, Alexander House, 1 Milton Road, Cambridge CB4 1UY. Try to ensure the tape itself is as reliable a brand as you can manage, include plenty of equipment details and a recent photo if possible.

Another Chapter in the Story…
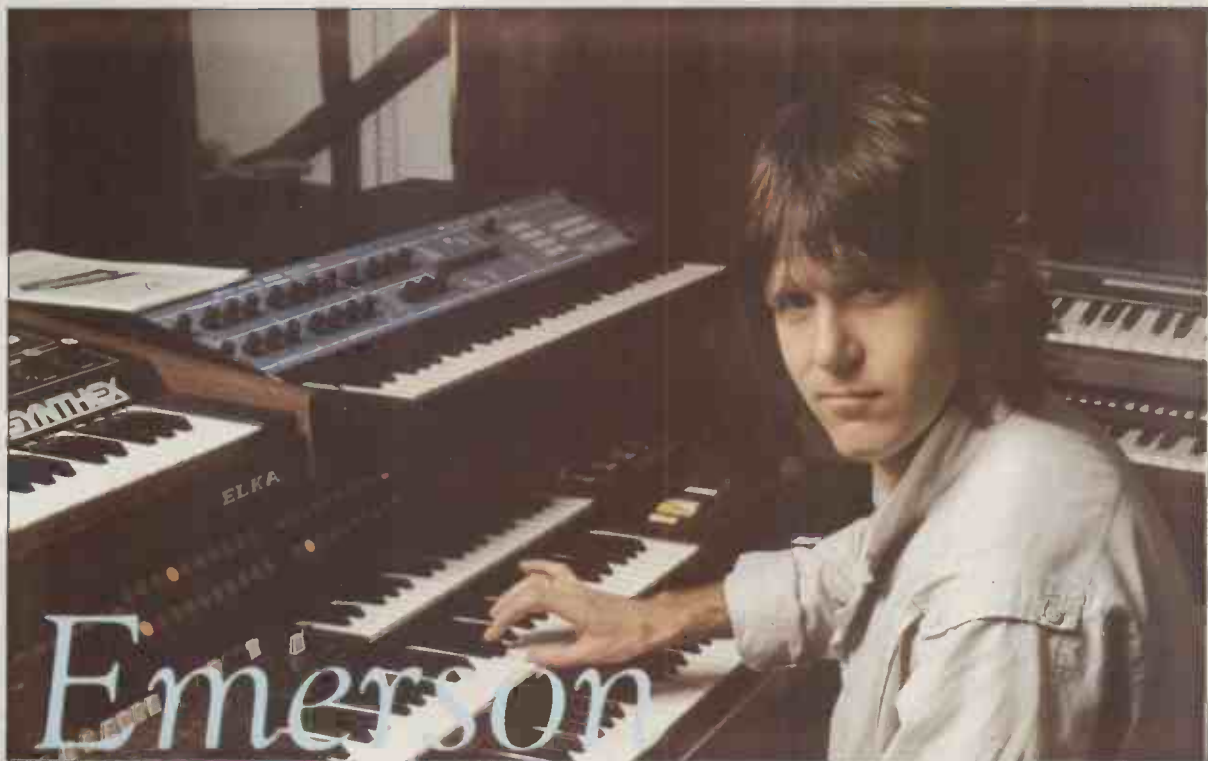
Emerson

Words by Dan Goldstein & Paul Wiffen; photographs by Matthew Vosburgh

Keith Emerson ended his last conversation with this magazine (E&MM May 83) with the following sentiment:

'Once I find the right combination, musicians I enjoy playing with, and we've put a repertoire together, we'll go out and play a little club somewhere – whatever the record companies think and, for that matter, anybody else.'

Two years on, that still hasn't happened. In that time, the world has heard little from the man who once led the field of rock keyboard players by a fairly considerable margin. There's been no record deal, no band and no gigs. Aside from the odd film and television soundtrack, Emerson has kept himself to himself, playing, writing and recording at his home studio in Sussex – letting the contemporary music world get by without him.

And got by it has. Synthesisers and computers have become integral parts of the commercial pop field, the keyboard player has begun to enjoy a status at least equal to that of other musicians, and better synth sounds have been made more easily accessible to a greater number of players.

Yet Keith Emerson has been oblivious to almost all of it. His distaste for much of modern music – and the way it's written – is already well-documented, and it's probably a major reason for his recent lack of creative output. Since the demise of Emerson, Lake and Palmer, he's showed no great interest in the prospect of compromising his musical ideals just to get back into the swing of things. So while he's stuck to his artistic guns, his

admirers have had precious little to get their teeth into.

Until now, that is. Months of trying to find 'the right combination' have proved fruitless, so Emerson has teamed up with Greg Lake in an attempt to rekindle the progressive spirit of old.

'I think it's no secret now that I've been working with Greg Lake and a whole load

> *'If anyone out there knows someone who might be able to design a MIDI retrofit for a Yamaha GX1, I'd love to hear from them.'*

of drummers. We needed a 'P' so we asked Buddy Prich to join, and we've been talking to Phil Pollins, Ringo Parr, and a few other names like that. I don't want to speculate too much, except to say that we are viewing things very seriously at the moment: I just don't want to name any definite names until we've got something solid in the can.'

Problems

Unfortunately, getting something in the can might be the first of Emerson's problems. He's got plenty of material, make no mistake, but an awful lot of it is written with a particular synth in mind.

The name of that synth? Yamaha GX1. Thirty thousand pounds' worth of handbuilt polyphonic prototype, the ultimate analogue performance instrument, and for all its idiosyncracies, Emerson's pride and joy.

'I bought this great beast back in 1976, and used it for the first time on ELP's *Works* album (from which the chart hit 'Fanfare for the Common Man' was taken). It was a lot of money then – those were the days when the pound was actually worth something.

'I remember these little fellows coming over from Japan with it and setting it up at our recording studio, which was in Fulham at the time. We wandered back down there one day after a very heavy lunch, and found all these Japanese guys bowing courteously at us. Anyway, I sat down and tried it, and that was that: I was at home with it from day one, and I had to have one. It was just so complete, even with its silly little drum machine, which seems very home organ-ish now but was quite revolutionary at the time. In fact, I often use it still for writing.

'The main thing about the GX1 was that it had such a good, fat sound that I've really been able to make my own, probably because very few other people were ever able to afford one, I suppose. The only thing it can't do is produce really strong bass sounds, but I got round that by getting my engineer, Nick Rose, to put CV and Gate connections on it so that I could control a Minimoog from the lower keyboard (there are three manuals on the GX). That was long before the days of MIDI, of course, though I'd really like to get it brought up to date in that respect. If

anyone out there knows someone who can design a MIDI retrofit for a GX1, I'd love to hear from them. That way I'll be able to use it in conjunction with the latest MIDI synths, which would really be something.

'But the real problem is that I'm so scared of moving the thing, not because it takes eight grown men to lift it, but because it's also extremely delicate, and servicing would be a real problem: I don't think there's anyone outside Japan with any circuit diagrams. In fact, we just moved it from one side of the studio to the other, and even then, a few things on it stopped working. So I think when it comes to recording it we'll have to bring a multitrack machine down here and do it, or alternatively, sample it into something. I'd hate to have to abandon it, because like the Hammond and the Moog system, it's got a sound people associate with me.'

Background

Yes, those were the other two great Emerson instruments. The Hammond L100 organ ('the first instrument I actually owned') on which he made his name as keyboard player with The Nice in the late sixties, and the modular Moog system (along with its more manoeuvrable successor, the Minimoog) which earned him the reputation as being one of the most refreshing and dexterous keyboard players in the rock arena.

Looking back, Emerson finds it difficult to pick a favourite from those key items of hardware, though his own particular Moog story is a fascinating one...

'The first time I actually used one on stage was with The Nice at the Royal Festival Hall in 1969. It belonged to Mike Vickers, and he came along and programmed it for me. He hid behind it for most of the duration, then occasionally he'd leap up, pull a few jacks out, swap them around, and change the sound. That was how we proved you could use the Moog modular system live – just.

'Just after ELP started up, I got in touch with Bob Moog and he sent me an instrument which he said was probably the nearest he could get to something that could really be used on stage on a regular basis. And it did allow me to make a few changes. You could make alterations to things like filtering and pitch, and at that time, strange noises were still very much in vogue: people had that underground mentality of thinking that wild noises equalled freaky music!

'You know, I was in Japan last year, and it was really weird to see all these ELP clones: they treat those odd noises as gospel. They feel they've got to copy every whoop and glitch I happened to produce on the particular evening we recorded on the *Pictures at an Exhibition* tour. They don't have ribbon controllers (as I did) to make these sounds, but they waggle away on Poly 800 joysticks, making noises they obviously regard as music. I certainly never thought of them like that – to me they were just part of the ▷ live show.

'The Minimoog was the other great godsend of that middle ELP period. It's interesting that I never really got into that bend-wheel thing which guys like Chick Corea have really made their own, but the reason for that was that I never had the spare hand. At that time, Greg was starting to play more guitar on stage, so I used the Minimoog to do the basslines while he was otherwise occupied.

'But it really was a classic instrument – once I'd got the switch to defeat the output while I was using the visual tuner.'

Current Instruments

Sadly, Emerson no longer has that Minimoog, and both his L100 and its successor, an extensively-modified Hammond C3, are lying dormant awaiting extensive technological surgery that may never come their way. What with the GX1's inherently delicate disposition, what newer machines has the player seen fit to invest in?

'Well, the first point to make is that I still do almost all my writing on piano, a Steinway. And most of the material I'm working on currently is still GX1-based, because when it's working as it should do, it's still unbeatable. Nick Rose built me a sequencer and several control

levers for my knees and feet, so it's a highly personalised instrument.

The one thing I really love about it – and which is going to be very difficult to replace – is the way I can use the upper solo keyboard in conjunction with the ribbon controller. You see, not only is the keyboard sensitive to both pressure and

> *'I never really got into using the bend-wheel on the Minimoog - but the reason for that was that I never had the spare hand'*

side-to-side movement (which is great for personalised vibrato – much more expressive than introducing an LFO), but you can transfer at any time to the ribbon controller. It runs above the keyboard along its entire length, and you can take complete pitch control from it while the gate is automatically held open. It's a unique feature and much nicer to use than a wheel system, because when it comes to transferring back to semitone intervals on the keyboard, you don't end up with a transposed keyboard.

'Still, I suppose I've got to be realistic. I've started to use a few more modern keyboards, like these little Poly 800s Korg keep sending me. But you can't deny things are difficult these days. There's so much good stuff coming out, but a lot of it is also quite expensive. I desperately want the next big instrument I buy to be the *right* instrument.

'I tried the Kurzweil, but before I make a decision on that, I need to hear their sampling update. The one I tried just had the standard sounds in it, including a nice Bosendorfer piano sound which could come in useful if you were in a studio that didn't have a piano. But then again, I've got my Steinway, and I'm perfectly happy with the sounds I get from that. The Kurzweil's string sounds were a little bit scratchy, but at least you can really *play* them, which is more than most synths allow you to do: usually it's more a case of just holding the notes down. I'm still toying with the idea of the Kurzweil, but it's a lot of money just for the presets: I'll have to see the sampling first.'

A sampling capability is important, then?

'Yes. Ideally I'd like to have a really good sampling system now, but there are so many due to come out over the next few months that I'm just going to have to wait and see. I've yet to try the Emulator II, but I've heard some very good things about it.

'On the synth front, I've just got myself an OSCar, which I really like. I suppose it's a sort of modern version of the Minimoog, if you like, but in some ways it's better because of its ability to create digital waveforms and because it doesn't need continual tuning. Then there's the Elka Synthex, which is good for string and brass sounds, and I've just got a PPG Wave that also sounds pretty good, though in a different way, obviously.'

If you're getting the impression that Keith Emerson is only just beginning to realise the sort of leaps and bounds musical technology has taken over the last three or four years, you'd be right. Because up until now, he's been content to remain aloof from the march of progress, and in fairness, anyone with a Yamaha GX1 has a right to pause for reflection while the rest of the keyboard-playing world catches up.

So it's only now that Emerson has decided he's waited long enough, that there's little point trying to gain a reputation as 'Emerson the Composer', that he can't go on writing music for low-budget films and television programmes, and that it's time to get well and truly back into the limelight.

And in the long run, that can only be a good thing. Because while contemporary popular music has seen a more democratic distribution of musical power, it lacks the presence of musicians who really care about the way instruments are played, not just programmed.

Work on the new ELP album is expected to begin in earnest within the next month or two, so watch out, world. And welcome back, Keith Emerson. ∎

ON STAGE

The man seems reluctant to play any gigs in the UK, but he's just undertaken a major tour of the rest of Europe, and E&MM sent spies to the Flemish University of Brussels to see Klaus Schulze perform some of his most recent material. *Jeanette Emsley*

We arrived at Brussels Nord station 12 hours after leaving our warm and cosy beds in deepest Essex. The temperature in Belgium was a good 10 degrees below, the snow thick and slippery underfoot.

No chance of seeing the soundcheck now – if only the Jetfoil had been operating! But the most difficult part of the journey – actually finding the venue – was still to come. When the guy at the information office of the main Brussels railway station doesn't know where the University is, you know you've got problems. Needless to say, we eventually arrived at the well-hidden (most University halls seem to be) location only a matter of minutes before the doors opened. It had been a close-run thing...

The Hall itself was an impressive one, with steeply-raked seating for about 600

and an extremely low (about 18″) stage that ensured everybody had a good view. The empty spaces at the edges of the auditorium were merely a result of people preferring to sit in the aisles in order to hear the music to better advantage – a good turnout.

As the concert began, plush velvet curtains opened to reveal a fine white net stretched across the front of the stage, through which filtered images of Crumar GDS and Fairlight monitor displays could just be seen. As the lighting behind the net increased in intensity, the ghost-like figures of Klaus Schulze and his long-time collaborator, Rainer Bloss, became apparent. And the music? 'From Long, Long Ago', taken from the Bloss album *Ampsy*.

The end of this piece (about three and a half minutes' worth) is cued by what

seems to be Schulze's favourite Fairlight orchestra preset: and you don't need to be a Schulze fan to know the one I mean. This in turn acted as the cue for stage manager and chief technician Claus Cordes (backstage) to flick the switch that would release the net. This he did, to the apparent astonishment of the newly-revealed Schulze and Bloss.

Next piece on the concert agenda was the title track from recent co-operative long-playing venture *Drive Inn*, performed with unexpected verve and given extra sonic colour courtesy of Bloss' electronically-treated vocals.

But out with the old, in with the new: the major part of the set was dominated by rhythm, generated by unmistakably Schulzian rhythm patterns and sequencer lines. The pieces themselves had a semi-improvised feel. The odd familiar

refrain would make itself evident now and then, but disappear again before it had time to become recognisable. Towards the end of the set we were treated to what *might* have been a rendition of 'Audentity' – the sounds were all there, but they were in the wrong order!

Schulze opened the gig's second half with a few words in English and French to what was a predominantly Flemish-speaking audience, but the crowd loved it, and gave the Germans a rousing reception as they launched into 'Drive Out', also from the *Drive Inn* album. A gentle piano refrain was gradually assaulted by synths and drum machines to produce a powerful if predictable climax, after which some steady rhythms reminiscent of the first half took over. Again, the music sounded new, with just the odd hint of the familiar.

An impressive electronic percussion solo (seemingly entirely improvised) gave way to one final sustained note that faded with the lighting – Schulze and Bloss had already left the stage.

Loyal Schulze fans will tell you that persistence pays when it comes to getting him to play encores, and so it proved in Brussels. But full marks nonetheless to the Belgians for their perseverance – I can't see a British audience ever waiting so long. Maybe they'd been tipped off in advance that the encore would comprise a startling rendition of Schulze's new single, 'Macksy'. The disc should be released by the time you read this, and if it's anything like the Brussels version, it'll be a catchy, beautifully-arranged instrumental with as much fire and energy as we've seen from the man for a while.

The encore's other elements were instantly recognisable – 'The Oracle' from *Ampsy*, with Bloss' eerie treated vocals, and the unforgettable 'Cellistica' from *Audentity*.

Like most of the audience, we reckoned that would be our lot. Photographer Alf Annison had already slipped backstage in readiness for a posed photo session, and some members of the audience had begun to make their way out of the Hall...when Schulze surprised everybody by returning to the stage for an impromptu solo performance. The composer tried to calm the audience by asking for patience as he set up a Yamaha CS80, and moments later, he launched into a rendition of some completely new solo synth music, played with verve and vigour in his inimitable style.

An enjoyable evening, then, if not a classic. Maybe it was us – had the long journey dulled our capacity to be entertained? Quite possibly. I just wish Schulze would acknowledge the fact that his British fans aren't going to get to see him in the present climate unless they're both devoted enough and rich enough to make the trip. I may be devoted, but like an awful lot of electronic music fans, I'm not even the tiniest bit rich. ∎

THE ISLE IS FULL OF NOISES

Tim Souster, avant garde composer, soundtrack writer and new-found
TV star, hits back at his critics. *Dan Goldstein*

Before the start of the decade, few people outside the academic and avant garde music field had even heard of Tim Souster. His name became half-famous after he helped write the music for Douglas Adams' *Hitch-Hiker's Guide to the Galaxy* on Radio 4, but his face didn't become widely known until a month or two ago, when he appeared on the South Bank Show's electronic music extravaganza, *The Isle is Full of Noises*.

History

Like many avant garde composers, Souster found himself immersed in the conventions of classical training at an early age. The only chance he got to expose himself to other forms of music was when he could sneak into the family bathroom and tune into Radio Luxembourg, though that only lasted a short while before his mother discovered what he was doing and promptly confiscated his radio.

'I suppose that was what attracted me to rock and jazz initially – the fact that it was forbidden. But I did the decent thing and studied Music at Oxford, and subsequently did a postgraduate degree in Composition.

'I was a contemporary of Hugh Davies, and it was he who built me my first piece of electronic equipment – a ring modulator with two inputs and one output. It must have been about 1968, and I started off by feeding records of ethnic and folk music into the circuit, with somewhat unpredictable results. Shortly afterwards I bought a sinewave oscillator from an Army Surplus place in Lyle Street – the thing was indestructible, built like a tank – and a grotty old Selmer 50 watt amp that was so loud it could kill a man at 100 paces...

'The initial motivation for writing electronic music was hearing Stockhausen, and also experimental rock bands like the Soft Machine.'

But rather than incarcerate himself within the confines of a recording studio, Souster opted to take his new-found enthusiasm for electronics to the people. During the time he was Composer in Residence at King's College, Cambridge, he formed a revolutionary avant garde performance group by the name of Intermodulation: the other founder members were composer Roger Smalley, who'd occupied the King's post before Souster, and Andrew Powell, who subsequently found a little fame and not inconsiderable fortune as a composer/arranger in the progressive rock arena.

Intermodulation were ground-break-ers. They wrote music in a huge variety of styles and for just about any combination of instruments they could think of: boundaries were few and far between. One of their pieces, a monstrous 73-minuter by the name of 'World Music', involved quadrophonic sound rotations to simulate a satellite orbiting the Earth, and required one of the group's then members, Robin Thompson, to play soprano sax, bassoon, electric guitar and electric piano at intervals during its length.

But such technical and musical complexity has its drawbacks, as Souster himself is only too aware.

'We had a remarkably good track record with electronic reliability in both Intermodulation and 0dB, the second group I was involved with. There was one occasion during a performance of 'World Music' when one of the speakers around the room went down. It was a bit embarrassing because you got a sudden attack of drastic minimalism by cross-fading into complete silence, so we had to stop. We traced the fault afterwards to a bad join in that particular speaker cable, but there was no way we could have known that at the time. Luckily, that sort of thing happens less and less these days with the increased quality of the electronic gear that's available. Mind you, there are

still people snatching defeat out of the jaws of victory and producing really duff concerts, even now.

'The really negative aspect is that if you write music specifically for a certain combination of musicians, as I have, it restricts the future performability of the piece. If you write things for strange combinations or certain groups of individuals you're always going to be restricted, and it's only now that we're getting away from that with groups like Electric Phoenix, for example. They're the first proper electronic ensemble that actually commissions people to write music for them and will then take it to different countries, giving maybe 30 performances of it instead of the usual one or two. I wrote a piece for them that they've played in Britain, the US, Canada, Scandinavia – all over the place.

'One of the problems with my own stuff is that it's difficult to get Publishers interested in putting money behind and promoting music that's written for odd combinations of instruments. But again, that's all beginning to change now.

Turning Pro

Yet in spite of his having received a good deal of critical and public acclaim during his years with Intermodulation and 0dB, it wasn't until the collapse of the latter assemblage that Souster decided he'd free himself from the bonds of academic convention and go it alone. Up until that time, he'd succeeded in taking up research posts at Schools and Colleges in places as far afield as Berlin and California, but there came a time when academic achievement simply wasn't enough.

'It was when I came back from America that I decided to take the plunge and start off my own studio in earnest, and try to actually *live* from my writing alone. I knew I'd like to carry on doing serious concert music, but although there are a very few people who can live entirely off writing serious music, I'm not one of them. My kind of concert music simply isn't mainstream enough for me to be able to do that, so right from the start, I've had to do whatever commercial work has been necessary to allow me to continue working on serious projects.

'My situation still isn't altogether satisfactory, because if you turn your hand to different sorts of musical writing, you get known as being a jack of all trades and master of none. I don't think I come into that category, but it's all too easy for people to form that impression. If you write library and TV music in addition to string quartets, people throw their arms up in disgust: they just don't understand the situation.

'Personally, I rather like turning my hand to different sorts of music because you can get interesting cross-fertilisations between them. One can learn an awful lot of technical things from doing music for commercials, for example. Working with such discipline and such intensity, making something last exactly 28 seconds, is certainly a valuable ex-

perience. I've had a couple of utterly mind-crunching sessions, one in which I worked through the night until 4pm to get a score fully mapped out to the tenth of a second and recorded so that the Director could hear it – and when I played it to him, he loathed it. So I just had to start all over again.'

In fact, that episode is one of many memorable adventures Souster has experienced at the hands of an advertising film director revelling in the name of Lester Bookbinder. It was for him that the composer wrote the matchless collection of sound effects that decorated the Alberto Balsam Jojoba commercials of a year or two back, though Souster himself

> 'A lot of people still think electronic music started when Rick Wakeman bought his first gold lamé cape: but it goes back a lot further.'

reckons his greatest work with Bookbinder will never be witnessed by the public...

'Lester was mad-keen on my *Spectral* piece, which imitates the song of the hump-back whale. He made a commercial for Hitachi electric razors using that piece, and he'd listened to it so intensely,

he'd managed to tie the image of a winking eye reflected in a razor blade exactly with a modulated pizzicato in *Spectral*. The finished film was really quite extraordinary, but when it went back to the Japanese, they hated it. In the end I think they put Holst's *Planets* on it instead, though I really couldn't think of anything less appropriate. It had been a marvellous piece of irony on Lester's part, because the Japanese are one of the few nations still hunting whales, but I don't imagine it'll ever be shown now.

'Lester's always wanted music that's as unconventional as possible – some of

the things I've written for him haven't been nearly weird enough – and now he's experimenting with sound himself. He's got an Emulator, and I don't think he's going to be the only Director to start doing things that way. After all, as the equipment gets better and gives you more and more direct control over sampling sound effects and so on, the more the people involved with visuals are going to want to get closer to the sound elements. And who can blame them? You can't say it's unjust or anything, because in the case of certain sorts of soundtrack, it's irrelevant whether the guy manipulating the sounds is a musician or a Director.'

At this point, both interviewer and interviewee start entertaining visions of the world's soundtrack composers joining dole queues *en masse*. We dismiss the possibility simultaneously, but it leads us on to the thorny topic of just how most film and television soundtracks are commissioned, and the headaches composers have to face as a result.

'Film music is still done as an afterthought in the majority of cases, at the last minute and when almost all the budget has been used up. Then there's a sudden panic and it all has to be finished by Tuesday, and the film companies wonder why the results aren't ideal half the time. It's such a short-sighted attitude, and it's really up to us as composers to get our message across.

'There have been a few exceptions

where the Director has been particularly sensitive to music and let the soundtrack be developed side by side with the rest of the film. Apparently, the music for the end of the first version of *Close Encounters* was written first, and Spielberg actually edited the film to fit this great chunk of symphonic music. That's something that ought to be done more often, I think, to eliminate the phone-call-two-days-before-screening syndrome.'

Studio

When E&MM last featured Tim Souster back in May 1981, he'd moved his centre

of operations from Cambridge to London's Camden Town. Now he's back in Cambridge, with a pleasant semi in a quiet part of town and a recording studio in his garden shed. That studio's equipment list has grown considerably over the past couple of years, due partly to *Hitch-Hiker's Guide* royalties and partly to some lucrative audio-visual work, which might be unremittingly mundane (the worst example featured the upper echelons of Metal Box Company management discussing export prospects to the Solomon Islands) but does at least pay well.

So whereas the studio of '81 was a simple eight-track affair with a Serge modular synthesiser as its centrepiece, Souster's current set-up is a Tascam-based 16-track with a full complement of contemporary synthesiser technology. The JP4 of four years ago is now a JP8, the Minimoog has been replaced by a DX7, and a BBC Micro has added music to its range of activities thanks to Souster's recently-acquired Music 500 add-on.

As it happens, Souster has already put the 500 to good use by writing a new piece, 'Work', for piano and appropriately-equipped Beeb. The piece received its premiere performance last month at the Cambridge College of Art and Technology, and will be doing a round-trip of the avant garde music circuit later in the year. The composer is pleased with it, though he's at pains to stress his experimentation with computers is at nothing more than an embryonic stage.

'I'm very interested in the concept of the synthetic performer. I wanted to set up an electronic system that was responsive to and interactive with the human performer, but there simply wasn't the time to do that with 'Work'. But the structure of AMPLE should make it possible to make the computer respond to certain strategies and take its own decisions accordingly.

'What the piece turned out to be was a sort of duel between man and machine. There was an interesting contrast between the musician getting a piece of music and phrasing it the way any trained pianist would (but which a computer finds great difficulty in doing), and churning out random chords very rapidly *ad infinitum*, which a computer excels at but which a pianist has extreme difficulty keeping up with.

'The other exciting thing is building up networks of dissemination. If you create a complete artistic package like *Ghost-busters* or *King Lear* or a symphony, you're laughing because you've got a fixed article that people can only respond to in a passive way. You refine it as far as possible, of course, which is where hi-fi and video come into it. And you can have a copy of *King Lear* with no typesetting errors in it, which means you've got a perfect object. But it's still just that – an object.

'Douglas Adams (the *Hitch-Hiker's Guide* author) is often going on about the idea of interactive literature: his computer game tries to go some way towards

achieving that because it involves the user so much. I'm sure that's not a new idea, but the computer is an ideal medium for the dissemination of art through a sort of modular audience network, because it can store all the various permutations. It's a very exciting development, and I'd love to do more with it in musical terms.'

Technology

So, with a studio whose high sound quality continues to surprise its user and whose range of hardware has proved itself capable of coping with the most demanding performance situations,

Souster only has one or two things left on his list of technological avenues worth following up.

'I need a sampling machine, though at the moment I can't afford one. I'm impressed by both the Emulator II and the Powertran MCS1. Having experimented with synthesised sound for so long, I'd like to get more into the sampling of concrete acoustic sounds, a proper miking technique, which is a fantastic art. Sampling enables you to inject that richness of acoustic sounds into electronic music: things sound a lot more interesting if you can add the enormous complexity of acoustic sounds to the simplicity of electronic ones.

'It's unfortunate, but you can already hear lots of sampling clichés in pop music, though I suppose that's a result of the highest technology being in the hands of the entertainment industry, where there's so much pressure to produce something people are already familiar with.

'But what is exciting is the prospect of generating a composite sound that isn't a banal, obvious concrete sound and isn't a plastic synth sound, either. Psycho-acoustically, that's fascinating because the brain is constantly teased as to what area it's actually in.'

And as well as his potential addiction to the sampling drug, Souster is also anxious to link as much of his current gear as possible to his BBC B, for reasons of synchronisation, automation and centralisation, if you get my drift.

But it's all got to be financed from somewhere, and Souster is under no

illusions that life as a freelance avant garde composer is going to get any easier. He's in Australia for almost the whole of April to tour a piece for flugelhorn, synthesiser and tapes, and expresses some pleasure at the fact that the piece will also be played solo at IRCAM in Paris while he's thousands of miles away. Once that academic sojourn has been completed, it'll be back to whatever commercial, audio-visual and film work the composer can find.

And while his recent TV appearance on Melvyn Bragg's South Bank Show will have given Souster's publicity rating a welcome shot in the arm, not all that publicity will necessarily have been good. For *The Isle is Full of Noises* has had more detractors than admirers, the principal criticisms being that it lingered for too long on historical and academic developments, touched all too briefly on technological advances in the popular field, and failed to illustrate technology's downward cost spiral and subsequent increase in accessibility. I put those points to Souster and found, to my eternal surprise, that he agreed with all of them. What's more, he succeeded remarkably well in convincing a complete sceptic like me that the motives behind the programme were right, even if the results weren't quite what was originally intended.

'I set out to show the historical roots of electronic music, which a lot of people still think started when Rick Wakeman bought his first gold lamé cape: it goes back a bit further than that.

'I also wanted to reflect as much as possible of the good music that's being made in this country at the moment. Obviously the pop thing is important, but we didn't want to make another programme dominated by pop musicians, because you can hardly turn the TV on these days without seeing a pop player of some kind.

But the criticisms are valid, because the programme's coverage of the pop world was a little perverse. That came about partly because the Director, Bryan Izzard, was very anxious to feature Andy Mackay, and because although I'd have preferred to have a chat with one of the contemporary computer music producers like Trevor Horn or Steve Levine, those plans eventually fell through. So the ramifications of all that were that we had The Explorers, whose music didn't really incorporate the technological innovation so many others have got involved with.

'You can't please everybody. I didn't want to spend a lot of the programme explaining the workings of various different synths, because that's done already in other places – E&MM does it every month. So one isn't exactly short of ways of finding out these things. I was more interested in relating those technical points to the music itself – what's actually going on in the forefront of music today, and the reasons *why* people ever choose to use electronics in the first place.'

A pretty fundamental point, you've got to admit.

WHITE CHINA

Merseyside's synth wonders continue their trek
across pop music's outer perimeter with the release
of a third album this month. Surprisingly, it's been
produced by Steely Dan's Walter Becker.
Dan Goldstein

iron, it doesn't hurt at all because all your nerve ends are burned away.

'Well, tell Paul White I think he's a liar', moans Eddie, as he skulks away in search of some more cannons.

The Connection

Clearly, I'm not likely to get the truth behind The Becker Connection from Eddie, but I might get it from bass player Gary, who's since walked over and become engrossed in the author's Frankfurt Report of E&MM March. Tell us how it came about, Gary.

'There are two sides to it, really. One is that when we were in Los Angeles, a couple of people at Warner Brothers (which is our label in the States) told us Walter had heard our last album and really liked it, and since all of us had always liked Steely Dan, we thought it might be a nice idea to do something with him if we ever got the chance.

'Then quite a while later, Virgin in this country told us they'd had an enquiry, and would we like to work with him? Apparently what had happened was that

WHITE CHINA

Steely Dan had signed to Warners but split up almost immediately afterwards, with the result that he'd never actually recorded anything for them. They'd more or less given up on him ever doing some more recording under his own steam, so they were trying to get him work as a producer instead. They showed him their roster of artists and tried to get him to work with the B-52s and people like that, but when he saw our name on the list he said he'd rather work with us.

'And that was just about it, really. Warners flew him over here, we met him and got on well with him, and we decided we'd work with him here in Britain.'

It all sounds blissfully smooth. Were there any problems? Clashes of personality or differences of opinion, that sort of thing?

'Nothing major, no. Obviously our musical backgrounds were different. Walter was very much reared on jazz, and that influenced the way he worked with us – you could tell from the sort of chords and pieces he worked out with us. But for the ideas that we had before we started recording, he was just the right person. The right person at the right time...'

So China Crisis were changing direction anyway, even before the transatlantic intervention?

'Oh yeah. Definitely. You see, we approached this album in a completely different way to the last one. For *Working with Fire and Steel* we did most of the

writing in the studio, putting things on tape more or less as they came into our heads. But this time we wrote everything first as a band, and went into a rehearsal room before the recording to make sure we had everything worked out the way we wanted it to be.

'It was at that stage that Walter came in. We spent two weeks at Rock City in Shepperton with him first of all: they've got what amounts to a disused 24-track down there away from the main studio, and that was where we set up with Walter, playing everything live and putting it down on tape. We've listened back to those tapes since we finished the album, and the material is incredibly close to what we eventually ended up with.

'From Shepperton we moved on to Parkgate Studios near Battle in Sussex. The reason we recorded there was that although we were keen to work at a residential studio, we didn't really have the time to book it all months in advance, and Parkgate was a well-equipped studio that was available straight away for however long we wanted it.'

Studio Work

'Well-equipped' is right. Parkgate is a fully-fledged, SSL-fitted studio that also happens to include a Fairlight in its list of hardware assets. Yet curiously, China Crisis and their producer had no need for the CMI's capabilities when it came to recording *Flaunt the Imperfection*. Instead, they got by with a remarkably down-to-earth set of gear. Surely there must have been times when the temptation to start working with the Fairlight was great?

'No, not really', says Gary. 'We used our own keyboards, a Jupiter 8, Yamaha DX7 and a Juno 60, linked up to an MSQ700 using Roland's MD8 converter, which lets you use MIDI keyboards with DCB ones. Apart from that, there was nothing really special at all.'

And yet in spite of (because of?) this lack of technological complication, the band's working relationship with Becker was going more smoothly than either side

> 'I think what both us
> as a band and Walter
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> was someone to
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> an objective attitude
> to things.'

had even dreamed of anticipating...

'It was incredible, really. I think what both us as a band and Walter as a producer needed was someone to bounce ideas off, some way of keeping an objective attitude to things. He'd always had Donald Fagen in Steely Dan, so it was a new experience for him, and it was the first time we'd ever written ▷

U nder the arches of a main line railway bridge doesn't sound like the ideal place to build a soundproof rehearsal studio complex, but there's one in just such a location on North London's Holloway Road that seems to be doing very nicely, thank you. At the time of writing, one of the bands taking advantage of the studio's facilities is China Crisis, the Liverpool synth combo who've proved time and again during their brief career that in today's pop atmosphere, you don't need to conform to succeed.

Now they look as though they're about to prove it all again, with the release of a new album (their third) entitled *Flaunt the Imperfection*. The disc was recorded towards the end of last year under the guiding hand of former Steely Dan mainman Walter Becker, and I decided from the outset that I'd make it my business to find out what had been behind this seemingly unlikely coupling. Was it a fruitful partnership born purely from reasons of artistic dedication? Or was it an unhappy arranged marriage, the result of heartless boardroom decision-making?

As I entered the Crisis rehearsal room, guitarist Eddie was hunched over a hot soldering iron, risking life and limb in the quest for that elusive object, the jack-to-cannon lead.

'All the synths have got to go direct into the mixer now that we've started rehearsing, and all the connectors on the back of the desk are cannons', he explains, in a scouse accent so strong it makes Frankie's Mark O'Toole sound like a latter-day Alastair Sim. 'So I've got to take six jack plugs off these leads and fit them with cannons instead – it's a messy business.'

It's certainly that. Eddie yells in mortal pain as he tries to pick up his iron from the wrong end, and I tell him of a theory, long held to be incontrovertible truth by E&MM Technical Ed Paul White, that if you burn yourself badly enough with a soldering

▷ everything so far in advance of the recording, so I suppose we just needed each other.

'Whenever Walter came up with a certain idea, it seemed as if it was what we were trying to do but hadn't quite managed: maybe we would have managed it in the end, but he gave us the idea sooner, which was great.'

Without wishing to turn out the 'wimp-rock band come up with heavy funk album' cliché, it's got to be said that *Flaunt the Imperfection* is a revelation. Or rather, a confirmation of the belief that all those clever musical ideas the band had already shown themselves to be capable of harbouring would one day be put forward in a more aggressive and confident manner. But more of that later.

As it turned out, Becker wasn't the only outsider China Crisis worked with to help achieve their newly-shaped aims. The band also availed themselves of the services offered by session keyboardsman Nick Magnus, who some of you might recall as being a key figure in Steve Hackett's band not so very long ago.

Equipment

'Nick had his own set-up', continues Gary, 'which consisted of a DX7 MIDI'd up to two Roland synth modules, a Planet S and a Super Jupiter. He was a great help, because he's one of those guys you can just ask to do something, and seconds later, he's done it — with no fuss. If there was an idea for a certain sound that we had in our heads or that Walter was thinking of, we just had to describe it in words and Nick would come up with it straight away.

'He was useful from a playing point of view as well, because there were quite a few things we knew from the rehearsals we would have trouble playing ourselves, but Nick just played everything straight off.'

But he wasn't using a mother keyboard to control the modules?

'Oh no.' Rejuvenated after his eventual success with the connecting cables, Eddie rejoins the conversation. 'I can't really see the point of those, to be honest. Well, I suppose if you're a really top-class player and you can appreciate a key-

board that's really well put together, they're worth it, but for the average muso, I just can't see them. That's why Yamaha cornered the market with the DX7 — they brought out a keyboard that was responsive to musicians but sounded really good as well. Give something like that a low price and it's obvious you're gonna be onto a winner.

'We didn't really get into the Fairlight when we were doing this album, and in some ways I'm glad we didn't, because there's no way we can afford one! One thing that does look good though is the new AMS keyboard interface. We've used quite a lot of AMS delay and reverb in the past, and although their sampling thing is only monophonic, it looks like it should do the job.'

Elsewhere on the equipment front, Eddie has fallen in love with Roland's GR700 MIDI guitar synth, though he's still playing it from an ageing G303 controller — 'I'll only get a 707 when I can afford the boots and helmet to go with it.'

His playing is only one of many positive points on *Flaunt the Imperfection*. Other

qualities that shine through after only a casual listening are an improvement in the clarity of singer Garry's vocal delivery, a tighter 'band' feel to the playing as a whole and, conversely, a looser, more relaxed atmosphere that must go down to the influence of Becker.

As for the immediate future, China

> '*I love the GR700 guitar synth, but I'll only get the 707 controller when I can afford the boots and helmet that go with it.*'

Crisis are now past the interviewing stage and are busy tidying up their live act in preparation for an extensive UK and European tour due to take place within the next couple of months. They hope to be venturing farther afield, too, with the States as the final goal, as Gary explains.

'It's such a massive market, but we'd love to go and play there. We did a lot of gigs in Europe last year as support to Simple Minds, but this year I think we've built up enough of a following to go out there on our own. America's different: if we have to play support to get there, then we'll do it.

'But I don't see any reason why we shouldn't make a success of it. We're all playing a lot better than we ever have, we've got a lot more experience of playing in front of an audience, and in general we're a lot more confident than we used to be.'

And they're justified in having that confidence. China Crisis have always written music that was worth going out of your way to pay attention to. They've always had something new to say, and they've always said it with invention and courage. But they're getting better. ■

The Time Machine

This project won't give you next week's racing results, but it might be the answer to your syncing problems if you're plagued by drum machines and sequencers of varying timebases. *Paul White*
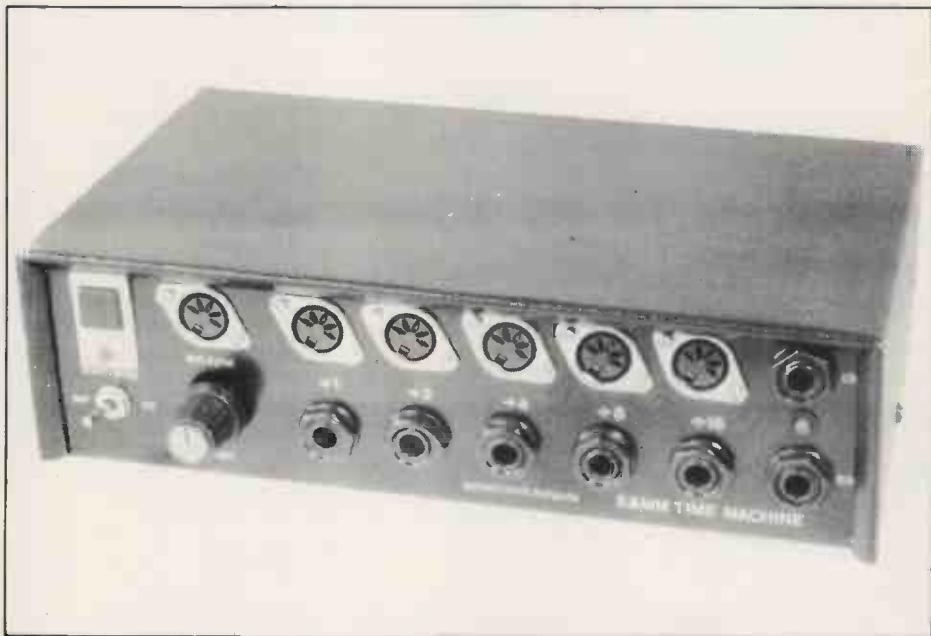
D esigned especially for those who still think that mercury comes from Hg Wells, the E&MM Time Machine overcomes two of the most common incompatibility problems musicians encounter when trying to persuade drum machines and sequencers of varying origins to run together in sync.

Problem number one is that not all machines use the same timebase (ie. the number of clock pulses per bar), which means you've got every chance of linking up two drum machines or sequencers only to find that one runs twice as quickly as the other. That's precisely what will happen if you try connecting a Roland Drumatix or Bassline with a Korg DDM series drum machine, for instance.

Problem number two is the way in which various clocking devices are started and stopped. If your sequencer or drum machine is fitted with a Roland-style DIN sync facility, its pulse will be stopped and started by means of a discreet control signal which is set to logical '1' for Go and logical '0' for Stop:

the clock itself runs continuously. On the other hand, something like a Yamaha RX15 requires only a clock to operate correctly. When the clock is present, it runs; when the clock stops, it stops.

What this means is that if the clock output from, say, a Drumatix is plugged into a Yamaha RX15, the latter will start to play immediately, even though the Roland may not be running yet. This is obviously bad news as far as synchronised starts are concerned: one machine will be halfway through the first verse before the second machine has even got its playing underway, and in most modern musical applications, this is clearly not on.

Design Aims

I'd like to be able to tell you that the Time Machine will synchronise absolutely everything to everything else, print money, and stop the Editor stashing his old Kentucky Fried Chicken wrappers in my in-tray. Sadly, not all of these prob-

lems are easily solved, and in the event, I've been forced to confine my attentions to the incompatibilities mentioned above.

The first of these can be overcome in one of two ways, *viz* using the fastest (in terms of clock pulses per bar) machine as the master and then dividing down the clock frequency until it's correct for the slower machine(s), or alternatively, using a master tempo clock to drive a divider chain, plugging your machines into the appropriate parts of the chain so that they run at the same speed.

E&MM's Time Machine offers both options, with four stages of division giving ÷1, ÷2, ÷4, ÷8 and ÷16 which should be adequate for most applications. If you need more than this, adding further stages shouldn't be too complicated, though if you're rich enough to buy the kind of specialist gear that runs at these higher speeds, you can probably afford something more sophisticated than this project to drive it.

In addition to Roland-style DIN sync connections, the Time Machine has a corresponding number of standard jack sockets to supply a gated clock output, ie. a clock output that's only present when the unit (or the master rhythm machine) is in Start mode.

The Circuit

IC1 forms the built-in tempo clock, and has a very wide range to accommodate machines being connected at different points in the divider chain (see Figure 1). It's still fine enough to allow you to set the tempo fairly accurately, but you could add a lower-value pot in series with VR1 to act as a Fine control, or a multiturn pot if your requirements are more stringent still.

ICs 5 and 7 are simply 'D-type' flip-flops configured as divide-by-two stages, while the latch formed by part of IC2 eliminates false triggering due to the switch bounce that will inevitably be

Figure 1. Time Machine circuit diagram.

GC = Gated clock output jack
S/S = Start/stop to output DIN
Ck = Clock   "   "   "
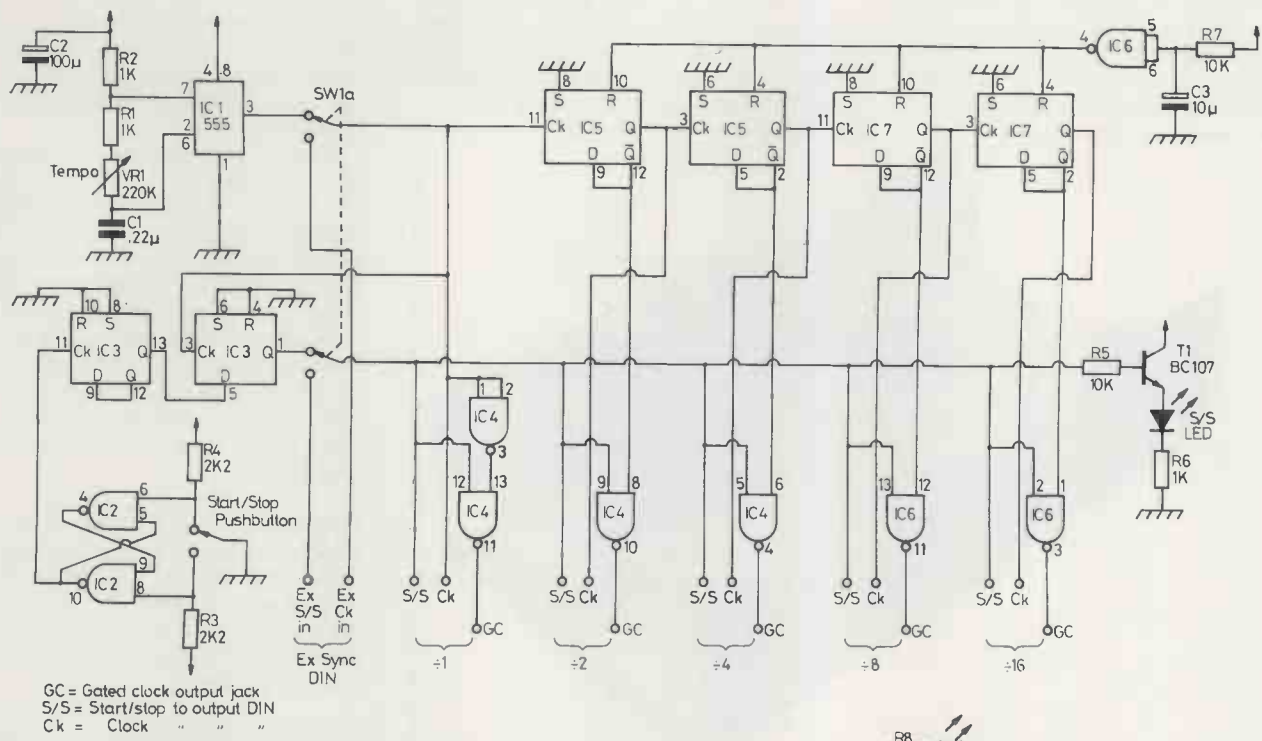
IC1	555
IC2	4011
IC3	4013
IC4	4011
IC5	4013
IC6	4011
IC7	4013
IC8	7812
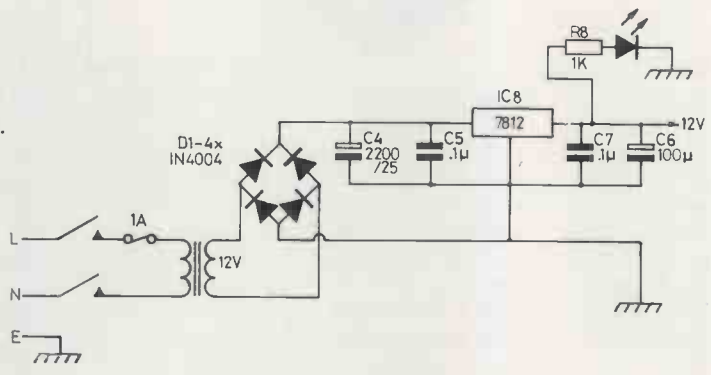
Figure 2. Time Machine PCB component overlay.

present in the mechanics of the start-stop pushbutton.

IC3a divides the switch output so that pressing the switch will alternately start and stop the system, while IC3b synchronises the rising edge of the start signal with either the system clock or external
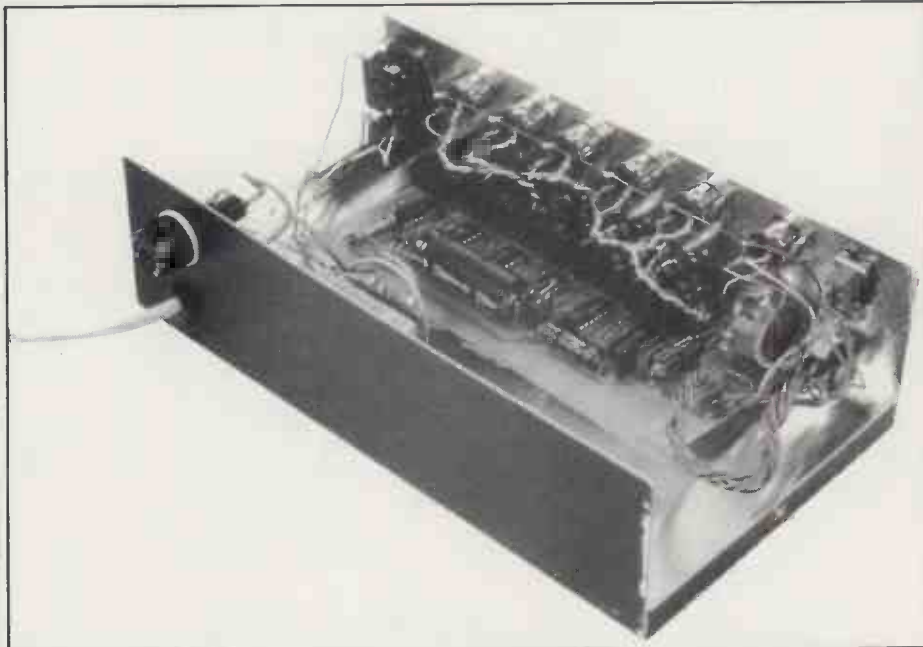
rhythm machine clock depending on what's being used.

IC6 generates a power-on reset which initialises all the counters when the unit is switched on, and the remaining gates simply 'AND' together the divider outputs and the start/stop signal to produce the gated clock outputs. T1 drives the start LED, and the entire circuit is powered from a regulated 12V supply which is built on the PCB to minimise wiring.

Construction

Referring to Figure 2, fit the components and wire links into the PCB, taking care to get the ICs, capacitors and transistor the right way round – if you're particularly paranoid, put the ICs in sockets. Use resistor leg offcuts to wire the jack sockets directly to the PCB – *don't* use special PCB-mounting sockets! Take particular care when soldering as some of the PCB tracks run very close together. Work in good light and check the track side of the PCB carefully before switching on – solder shorts are easily missed.

As you should with all mains powered projects, treat the mains voltage with respect and sleeve the transformer and fuseholder tags with rubber sleeving.

Whatever type of case you choose to house your Time Machine, you'll have to drill a few holes to accommodate the sockets and switches. This should present no problem to the intrepid E&MM reader providing he or she has access to a Black and Decker drill and a centre punch. If you possess a craving for a mains-indicating LED, wire it to the +12V supply *via* a 1K resistor, though note that there is no specific provision for this on the PCB.

Testing

Put at least two coats of whitewash on the windows, build a shelter against an inside wall using furniture and bags of earth, and make sure there is at least one sofa between you and the Time Machine before you switch on. And don't forget to check the date...

Seriously, the first thing to check is that the 12V power supply is running OK: this *must* be verified before you proceed any further. With the int/ext switch set to internal, check that the start/stop LED goes on and off when the start/stop switch is pressed. The gated clock outputs are best checked with a scope, but as it's perfectly conceivable that some of you don't actually have a scope, one alternative is to try plugging them into an amp with the volume turned well down and listen for a rasping tone. The next step is to connect a DIN-sync compatible unit such as a Drumatix (set to receive external sync) into the DIN outputs. It should run from all sockets but the speed will obviously differ by a factor of two for each sync socket with the ÷16 output giving the slowest speed. If you have two such machines, switch to external and use one of these as the master to check out this facility.

In Use

Just about any DIN-sync (no, not MIDI) machine may be plugged into the 'ext' socket to act as the master clock, but 'gated clock' machines cannot be master, I'm afraid. However, the Time Machine's internal tempo clock can be used to drive all the sync outputs so that 'DIN sync' and 'gate clock' equipment may run simultaneously.

One word of caution: check that the slave machine isn't midway through a pattern before you sync everything up, as devices such as the Yamaha RX15 will happily continue from where they left off

when triggered externally. Just thought I'd mention it.

There's also no reason why the Time Machine shouldn't be used in conjunction with an MPC Sync Track so that rhythm machines and sequencers can be synced to tape, so the system offers a lot of flexibility for very little outlay.

However, one species of machine that won't accept a sync code from the Time Machine is that which needs a Start tone in addition to the clock code: the Roland MC202 falls into precisely this category. You can still use it as the master machine in the system, though, so all is not lost.

All components are readily available from Maplin or other component suppliers and the total cost should not be much over £25 including a suitable case so you'll still be able to afford next month's E&MM. ∎

Time Machine Parts List
Resistors

R1,R2,R6,R8	1K ¼W
R3,R4	2K2
R5,R7	10K
VR1	220K lin

Capacitors

C1	0.22µF
C2,C6	110µF/25V axial electrolytic
C3	10µF/25V axial electrolytic
C4	2200µF/25V axial electrolytic
C5,C7	0.1µF
LEDs	Red (2 off)
Diodes	D1-D4 1N4004

Semiconductors

IC1	555
IC2,4,6	4011
IC3,5,7	4013
IC8	7812
T1	BC107 or similar

Miscellaneous

SW2 SPDT pushbutton or biased toggle switch (min)
SW1 DPDT toggle (min)
3- or 5-pin DIN sockets (6 off)
¼" jack sockets (7 off)
control knob
instrument case
12V/200ma transformer
Fuse holder
Mains switch
6ba nuts, bolts & washers
PCB

BACK TO BASICS

We've covered how synths generate sound and how that sound can be filtered – now our synth beginner's series looks at how things are shaped using amplifiers and envelope generators. *Steve Howell*

The third stage in the creation of a synthesiser's output comes under the loose title of 'shaping'. In order that this process can take place, most analogue synths come equipped with two distinct bits of circuitry, a Voltage Controlled Amplifier (VCA) and an Envelope Generator (EG).

The first of these is extraordinarily simple, and the best way to think of it is as a voltage-related volume control. All VCAs work on the principle that the higher the voltage you feed into them, the louder their audio output will be.

Figure 1 shows the block diagram of a typical VCA, and the first thing to notice is the similarity between it and the diagram of a VCF we published last month. The major difference is that, in the normal run of things, the VCA has only one CV input, normally connected to the voltage output of the Envelope Generator.

There are a few exceptions to this rule. Some synths such as the Roland Jupiter 6 have a second input that enables you to modulate the VCA with the output of another controller such as a Low Frequency Oscillator; old ARP synths have a DC voltage available on a slider which allows you to keep the VCA permanently 'open' for indefinite drones; and the new breed of touch-sensitive models such as the Prophet T8, Akai AX80 and Siel DK80 have their keyboards' dynamic voltage routed to another CV input on the VCA. But hang the exceptions – let's concentrate on the average design that gives the VCA one CV input from the EG.

Amplifiers

Because of its inherent simplicity and lack of user-variable parameters, the VCA doesn't actually appear on many synth control panels, unless it's at the machine's overall volume control. This stage might consist of just that – a simple potentiometer between the VCA and the synth's output jack – or it might be designed in such a way as to let you regulate the output level by the amount of voltage you send to the VCA's EG control input, in which case you'll find a level control between the two modules. It's unlikely your synth will be fitted with both design options, unless it's a programmable model on which the EG modulation amount can be stored as part of a program, so that you can match the levels of all your own sounds and still have the overall level control thrown in for convenience.

I hope all that has made sense, because I'm going to go straight on to the Envelope Generator without any further ado.

Korg's DW6000 adds two parameters – Break Point and Slope – to the traditional ADSR envelope control format.

Figure 2 shows the block diagram of a typical one.

Any sound, whether it's the result of a load of synth modules doing their stuff, the gentle breath of a flute, or an embarrassing burst of unexpected flatulence, has an envelope shape of some kind. This shape is best described in layman's terms as the way the sound's amplitude varies during the course of its existence.

Some sounds reach their full volume very rapidly but decay in level with similar speed; others take rather longer to get to their loudest point but will sustain at that level for as long as a note is played. The possibilities are endless, and on a synthesiser, there are (at least) four controls that allow you to regulate the comings and goings of a sound's amplitude changes.

ADSR

The controls in question are represented by that mysterious set of initials ADSR, and I'll explain each of these in turn.

The time it takes for a sound to reach its full volume is known as the Attack (hence 'A') time, and the appropriate control allows you to vary this time over a range of between, say, a few milliseconds and 10-15 seconds. The Decay ('D') portion enables you to set the time it takes for a sound to fade away to the Sustain ('S') part. Finally, the Release ('R') segment of the envelope lets you predetermine the length of time it takes for a sound to fade away to nothing after your finger has left the key.

If all this has succeeded only in leaving you all the more confused, have a look at Figure 3. This graph actually shows the voltage the EG generates whenever a note on the keyboard is pressed. Note how the voltage rises during the attack phase, falls during the decay portion, remains constant for the sustain time and falls to zero during the release phase. And bearing in mind that the VCA is sensitive to incoming voltages, you should be able to work out that the audio input will get louder as that voltage increases. It follows, therefore, that as the voltage output from the EG rises and falls, so the sound will get louder and softer in accordance with those voltage changes.

Figure 1. Block diagram of a typical Voltage Controlled Amplifier.
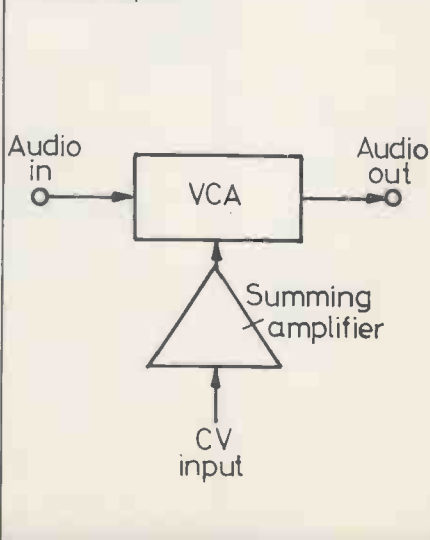
Figure 2. Block diagram of a typical Envelope Generator.
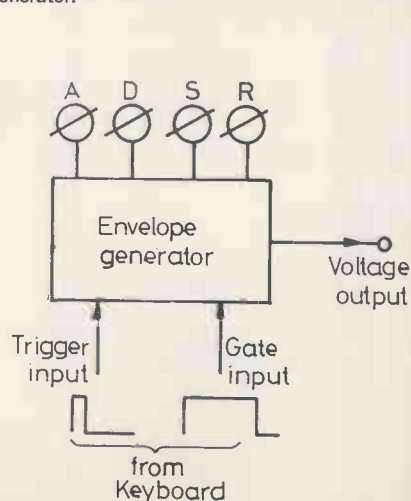
And if *that's* confusing, a quick glance at Figure 4 should rectify matters. You'll see that the sound at the VCA's audio input is at a constant level, but that it becomes the shaped signal appearing at the output once it's been tampered with by the EG's voltage changes. It's that same shaped signal that appears at the output jack of your synth, and Figure 5 shows a few examples of the sorts of shapes that are made available by an ADSR-type Envelope Generator.

Many of you will no doubt be in possession of synths with more elaborate envelope-shaping systems, as the ADSR set-up is by no means the world's most comprehensive. However, all the more advanced systems do is split the envelope into smaller sections so that you can draw a more detailed picture of your sound's amplitude-thru-time pattern.

Triggering

Only snag is, the EG can't actually generate this variable voltage until it gets the necessary inspiration, and more often than not, this takes the form of a trigger or gate pulse derived from the keyboard. Because at the same time as it delivers a control voltage to the VCOs and VCF, the keyboard also sends a pulse to the EGs.

In fact, a synth's VCOs produce sound continuously, so it's left up to the VCA to shut that sound off completely until it receives the EG's voltage shape, which it won't unless the EG has been triggered by the depression of a note on the keyboard.

But why do you need a trigger *and* a gate? Well, some sounds require a style of playing in which the whole envelope shape is heard for each new note played, whilst others do not. A pianist effectively re-triggers the entire sound envelope every time he or she plays a new note, but a flautist only has to blow once to play a whole series of notes. To imitate the sound characteristics of the former, you need to create a patch that lets you hear the attack and decay phases for each new note played, but a realistic recreation of a flute sound needs only the provision for a change of pitch during the sustain portion: the attack and decay phases don't have to be repeated. And this is where gate and trigger pulses come in, because whereas the former is reponsible for the entire envelope, the latter concerns itself only with making sure that the attack and decay portions are heard each time a new note is played on the keyboard.

Versatility

The true value of having both these means of triggering simultaneously available probably won't become apparent until you start programming on a regular basis, because it's only then that you'll realise the number of different playing styles it allows you to use. ▷

The Moog Prodigy – just one example of a monosynth that uses single triggering to initiate envelope cycles.

Figure 3. An Envelope Generator's voltage output shape.
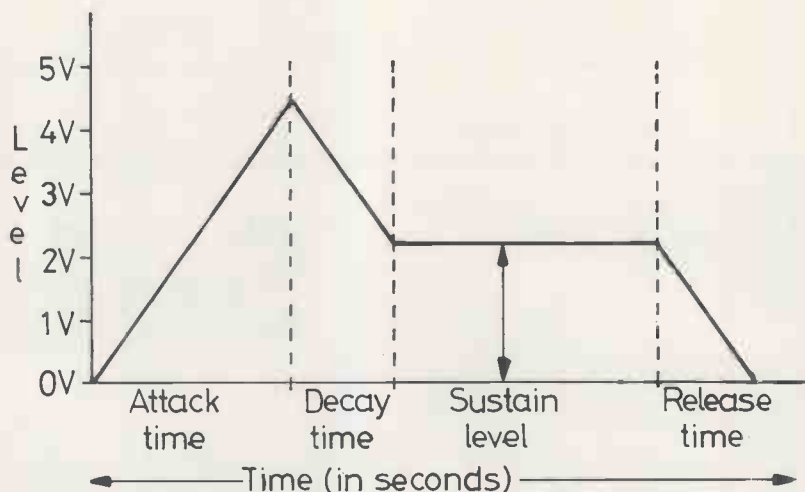
Figure 4. Envelope shaping in action.

Piano   Strings   Organ   Flute   Percussive
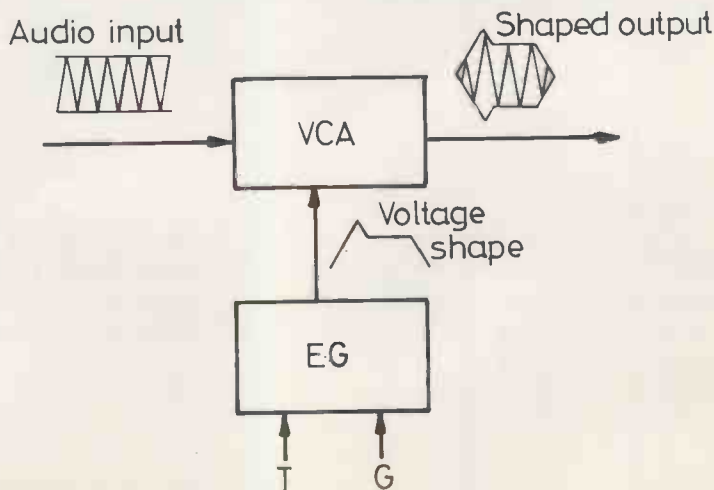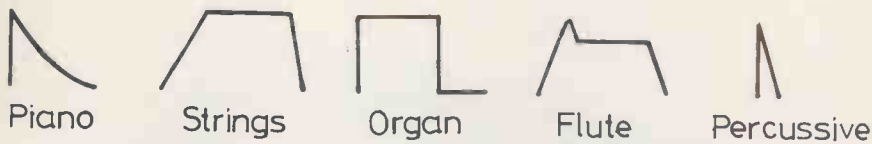
Figure 5. The Steve Howell Guide to Instruments and their Envelope Shapes.

Various 'Electronic' envelope shapes

Figure 6(a). Single triggering.

Notes played

Gate pulse produced

Resulting envelope shape

— time —

Notes played

Gate pulse produced

Trigger pulse produced

Resulting envelope shape
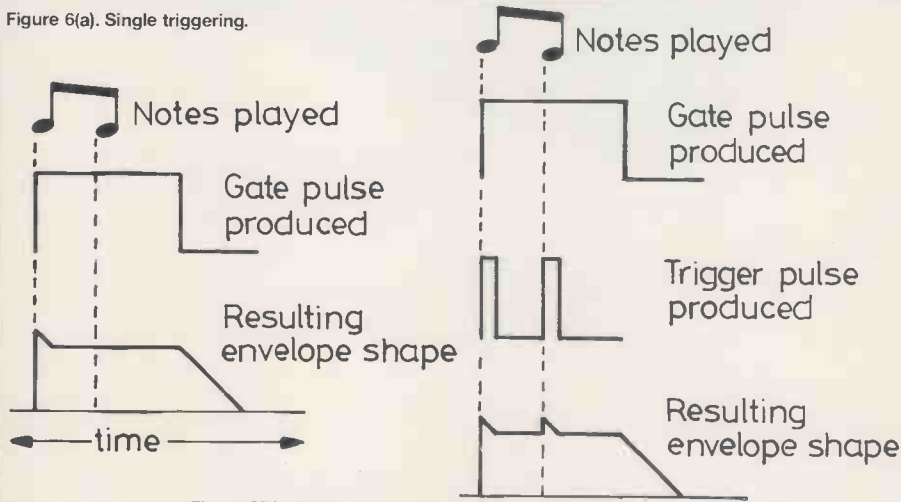
Figure 6(b). Multiple triggering.

▷ For instance, legato styles can be easily accommodated by removing the trigger pulse and relying only on the gate for firing the EGs, while more percussive playing styles can be used if both gate and trigger systems are used to initiate the envelope cycle.

True, older synths usually had either one or the other. The Minimoog and Moog Prodigy, for instance, incorporated only a gate system, which meant that percussive phrases were difficult to play unless you employed impeccable fingering: if you didn't, you got some notes coming out without any attack or decay portions. That 'gate only' system was (still is) referred to as Single Triggering, while any synth that uses both gates and triggers to initiate an envelope cycle is said to use Multiple Triggering. Figure 6 shows the difference between the two systems in visually-digestible form.

Most recent monophonic synths incorporate a switch that allows you to select between the two methods, while polysynths employ Multiple Triggering in all modes except mono ones.

Well, if all that hasn't left you utterly dazed and perplexed, collect your souvenir badge on the way out. And if it has, don't despair: it's probably the single most confusing element in the theory of sound synthesis, save the decoding of Japanese instruction manuals. At least we've got it out of the way.

PATCHWORK

Readers send in details of their own synth patches and how they can be played...

Here's another chance to show off your latest sonic creations and share some favourite patches with fellow readers... If you feel like blowing your own trumpet, send your offering on a copy of an owner's manual patch chart (including a blank one for artwork purposes) to *Patchwork*, E&MM, Alexander House, 1 Milton Road, Cambridge, CB4 1UY.

ROLAND SH101   'Expressive Flute'   K B Thomas Mid-Glamorgan

We were mightily impressed by this flute patch, and especially by the manner in which an increase in amplitude can be introduced with the 101's Bender control. To obtain the characteristic 'breath' element of an acoustic flute sound, simply add a small amount of Noise.

KORG POLY 61   'Steel Drums'   John Martin Lancs

DCO I			DCO II				VCF				EG				VCA	MG			
11	12	13	21	22	23	24	31	32	33	34	41	42	43	44	51	61	62	63	64
8	1	0	4	1	1	5	33	4	1	1	8	13	0	12	1	9	0	0	0

John must have taken a break from his guitar for this one. What's that? That one spells his surname with a 'y'? Dammit. Thought we had someone famous writing in for a moment there. Not finding the Steel Drums patch supplied with the Poly 61 to his personal liking, John has created an alternative version which can also be adjusted to produce more off-the-wall effects when Parameter 64 is set to about 4.

ROLAND JUNO 106 — 'DX7 Brass 1' — Steve Clark, London

Steve's getting to be something of a *Patchwork* regular, and seems to have devoted most of his recent programming towards trying to make a Juno 106 sound like a DX7 — more specifically, the Yamaha's warm 'Brass 1' preset. He mentions that the VCF and ENV settings are fairly critical, but if you're anxious to make up for the fact that unlike that of the DX7 the Juno's timbre does not vary depending on what key velocity you apply, you can adjust the VCF ENV parameter to give a variety of different timbres. The setting shown results in an approximate imitation of a harder key strike on the DX7.
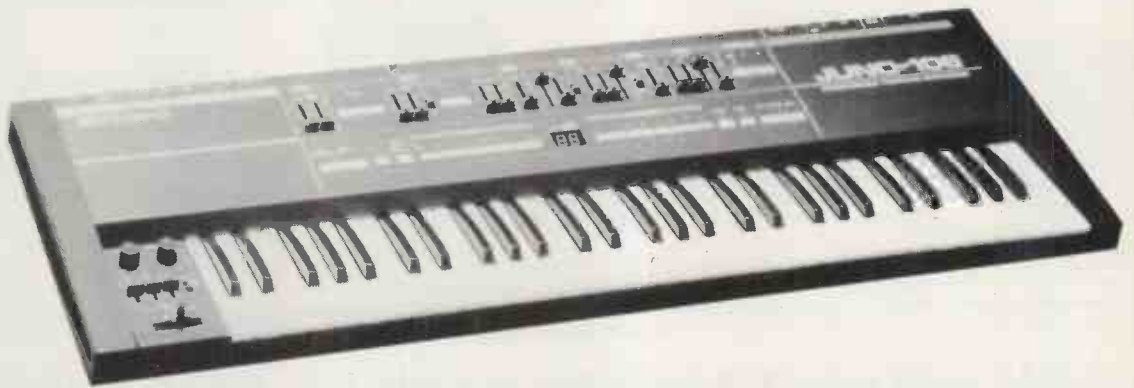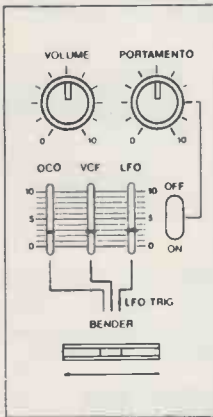
YAMAHA DX7 — 'HonkiGrand' — Robin North, Swansea

So, while people who haven't got a DX7 try to make their synth sound like one, people who have will do anything to avoid making the fact obvious. Why anyone would want to synthesise a 'clapped-out Bechstein' grand piano sound is beyond the comprehension of us mortals at E&MM, but that's exactly what Robin has done. He swears no disrespect to the great piano maker, but we have our doubts...

YAMAHA DX7 VOICE DATA LIST

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
18	0	TRI	30	0	0	0	OFF	4							
ALGO RITHM	FEED BACK	WAVE	SPEED	DELAY	PMD	AMD	SYNC	PITCH	AMPLI TUDE						
				LFO				MOD. SENSITIVITY							

DP: 6=0, 5=0, 4=0, 3=0, 2=0, 1=0

	POLY/MONO	PITCH BEND RANGE	STEP	PORTAMENTO MODE	GLISS ANDO	TIME
	POLY					

OP			FREQ COARSE	FREQ FINE	DETUNE	RATE 1 2 3 4	LEVEL 1 2 3 4	BREAK POINT	CURVE L R	DEPTH L R	K.BOARD RATE SCALING	OUTPUT LEVEL	VELOCITY SENS.	RATE 1 2 3 4	LEVEL 1 2 3 4	KEY TRANS POSE	VOICE NAME
6	R		5	00	−3	92 71 58 36	99 0 0 0	A2	−L −L	0 98	3	78	1				
5	R		0	50	+5	90 71 33 31	99 0 0 0	C2	+E −L	0 26	3	94	7				
4	R	OFF	0	50	+3	97 27 10 25	99 86 48 0	C8	+L +L	0 0	3	60	1				
3	R		7	00	+4	90 32 20 90	99 65 0 46	C8	−L −L	4 4	5	80	0	94 67 95 60	50 50 50		C2
2	R		2	00	+1	95 16 28 75	99 99 0 0	G#2	−L −L	0 0	2	86	0				
1	R		1	00	0	86 93 23 50	99 99 0 0	A-1	−L −L	0 0	3	99	0				
	MODE/SYNC		OSCILLATOR		DETUNE	EG RATE	EG LEVEL	KEYBOARD LEVEL SCALING			SCALING	OPERATOR		PITCH EG			

17	18	19	20	21	22	23	24	25	26	27	28	29	30	31	32
RANGE	PITCH	AMPLI TUDE	EG BIAS	RANGE	PITCH	AMPLI TUDE	EG BIAS	RANGE	PITCH	AMPLI TUDE	EG BIAS	RANGE	PITCH	AMPLI TUDE	EG BIAS
MODULATION WHEEL				FOOT CONTROL				BREATH CONTROL				AFTER TOUCH			

/COMPETITION

THE DIRTY DOZEN

Your Chance to Win Akai's AX80 Polysynth in Another E&MM Exclusive Competition

A lready a force in the hi-tech music field after only a few months in the business, Akai will be unleashing a whole range of new computer-based hardware over the next few months. The first model in that line-up, the AX80 polysynth is available now, and looks like being a strong contender with its dynamic keyboard, two oscillators (plus a sub-oscillator) per voice, MIDI compatibility and headache-easing bar graph parameter displays: now there's a modified selection of factory presets to show the 80's Curtis sound chips off to better advantage, too.

But before you take off for your nearest music retailer in search of one of the new Akais, cast your eye over the next couple of pages, because if you can answer the questions relating to the 12 photographs reproduced below, an AX80 could be yours for no more than the price of a postage stamp.

The questions aren't easy, we admit. But if you've a good memory for historic photographs or a penchant for modern musical trivia, they shouldn't pose too many problems. The quiz is divided into two sections — the first is concerned with aged musical instruments that met with varying degrees of commercial success during their careers, while the second deals with the musicians that may or may not have used them.

Once you've got a dozen answers you feel stand at least a half-decent chance of being right, fill in the relevant spaces on the entry form and post it to us, though make sure you've left plenty of time before the closing date of May 7: the Cambridge post is getting worse.

Best of luck!

THE FIRST HALF

1 It looks like something out of the Dalek City in an old Doctor Who *adventure*, but this see-through synthesiser was actually shown at exhibitions a couple of years back, and rejoiced in the name of *Gleeman Pentaphonic*. Which country spawned it?

2 This complex-looking creation was hailed as Britain's answer to the LinnDrum, and Kaja-googoo used an orange one with some success. What was the name of the company that produced it?

3 The performance wheels of a synth whose designer is an enthusiastic advocate of giving musicians new ways of controlling electronic sound as alternatives to keyboards. Name the synth.

4 Above is a hard copy screen dump from a music software package for the BBC Micro. The package is being sold now through a range of retail outlets, but which well-known record company is doing the marketing?

5 This Canadian computer music system was unveiled at last year's Frankfurt Music Fair and shown again in 1985, but its final specification, availability and price have still to be confirmed. What's the name it carries?

6 The gentleman in the photograph on the right is doing his best to demonstrate a breath-controlled synthesiser of German origin. The idea was to recreate brass and woodwind sounds with greater tonal accuracy, but it never really caught on. What was its name?

THE SECOND HALF

7 The man on the extreme left was on the cover of E&MM August 84, while the one on the far right was instrumental in the creation of the Band Aid campaign. Name the late seventies group that featured both of them.

8 This man can number amongst his collaborators such notable figures as The Edge, Karlheinz Stockhausen and Jah Wobble. Of which early improvisational group was he an important member?

9 Following several years' involvement with a famous German electronic music trio, this man recorded a new version of a song made famous by Frank Sinatra. Name that song.

10 The accompanying photo shows a band whose main claim to fame is their extensive use of 'scrapyard percussion', and whose first album was entitled Machine Age Voodoo. Name the duo's female half.

11 Kraftwerk used this elaborate stage set to house their performing and recording hardware during a world tour a few years ago. Which of their albums was released to coincide with that tour?

12 You're probably more used to seeing this musician dressed in elaborate stage costumes or sitting next to mixing desks on E&MM front covers. His stage name is well known, but what does his Birth Certificate list him as being called?

OFFICIAL ENTRY FORM

My answers to The Dirty Dozen are as follows:

1 ....
2 ....
3 ....
4 ....
5 ....
6 ....
7 ....
8 ....
9 ....
10 ....
11 ....
12 ....

Send this form to The Dirty Dozen, E&MM, Alexander House, 1 Milton Road, Cambridge CB4 1UY, to arrive no later than second post, Tuesday May 7. We regret that no photocopies can be accepted.

Name..
Address ....
....
....
Post Code ....Daytime ☎ ....

RULES

All entries must be on the official entry form published in E&MM April 85: no photocopies can be accepted. The winner will be the competitor whose entry is the first all-correct to be picked out of the hat during the week following the closing date. The winner will be notified by post or telephone shortly after that date, which is second post, Tuesday May 7 for all entries. The judges' decision is final, and no correspondence regarding the choice of entries will be entered into. Employees of Music Maker Publications, Akai UK and Alexander Public Relations are ineligible for entry.

SHORT CIRCUIT
The MIDI Index

Got a MIDI synth that won't play ball on anything other than Channel 1? Here's a quick project that may be of use.

To be honest, we weren't entirely sure whether we should publish this circuit at all. And the reason for this uncertainty? Simply that, as far as we're aware, nobody has actually built the thing to find out whether or not it stands an even chance of doing its job.

The designer, one M J Dodd from Worcester, certainly hasn't put it together, and nobody here at E&MM has had the time to gather up all the components and put soldering iron to veroboard, so if you do decide to give it a go, you're on your own.

Well, not entirely. Technical Ed Paul White sees no reason why the circuit *shouldn't* work, so long as all you want to transmit and receive is note off/note on information. Some slightly more off-the-wall MIDI commands don't incorporate the channel number as part of their control word's end, and seeing that the circuit shown relies on that being the case to operate successfully, more complex MIDI activities such as patch-dumping from one device to another will have to be undertaken with care.

The MIDI Index diagram illustrated is accompanied by a 5V power supply circuit knocked up by said Mr White in about 30 seconds, so all you need to add are the MIDI In and Out connections, and the design will be complete. We think. ∎
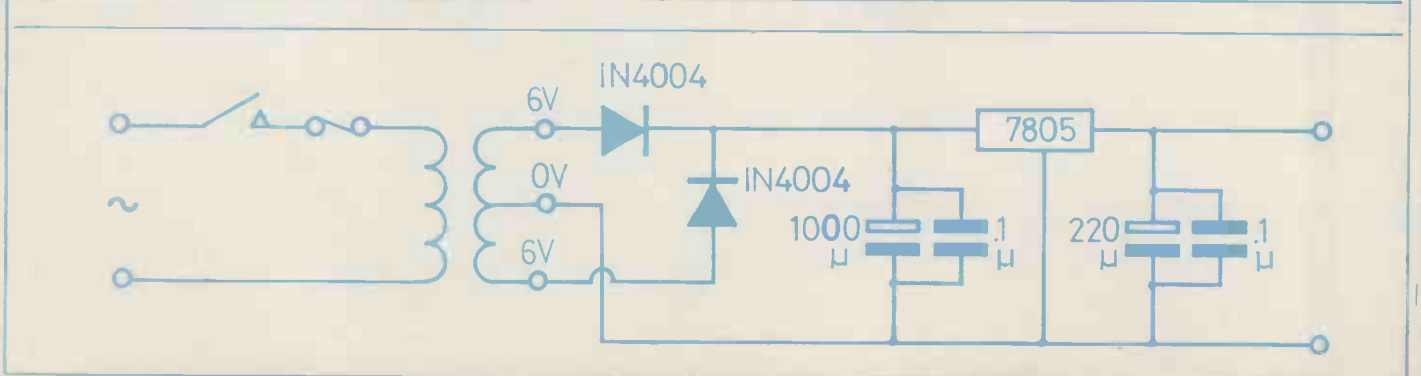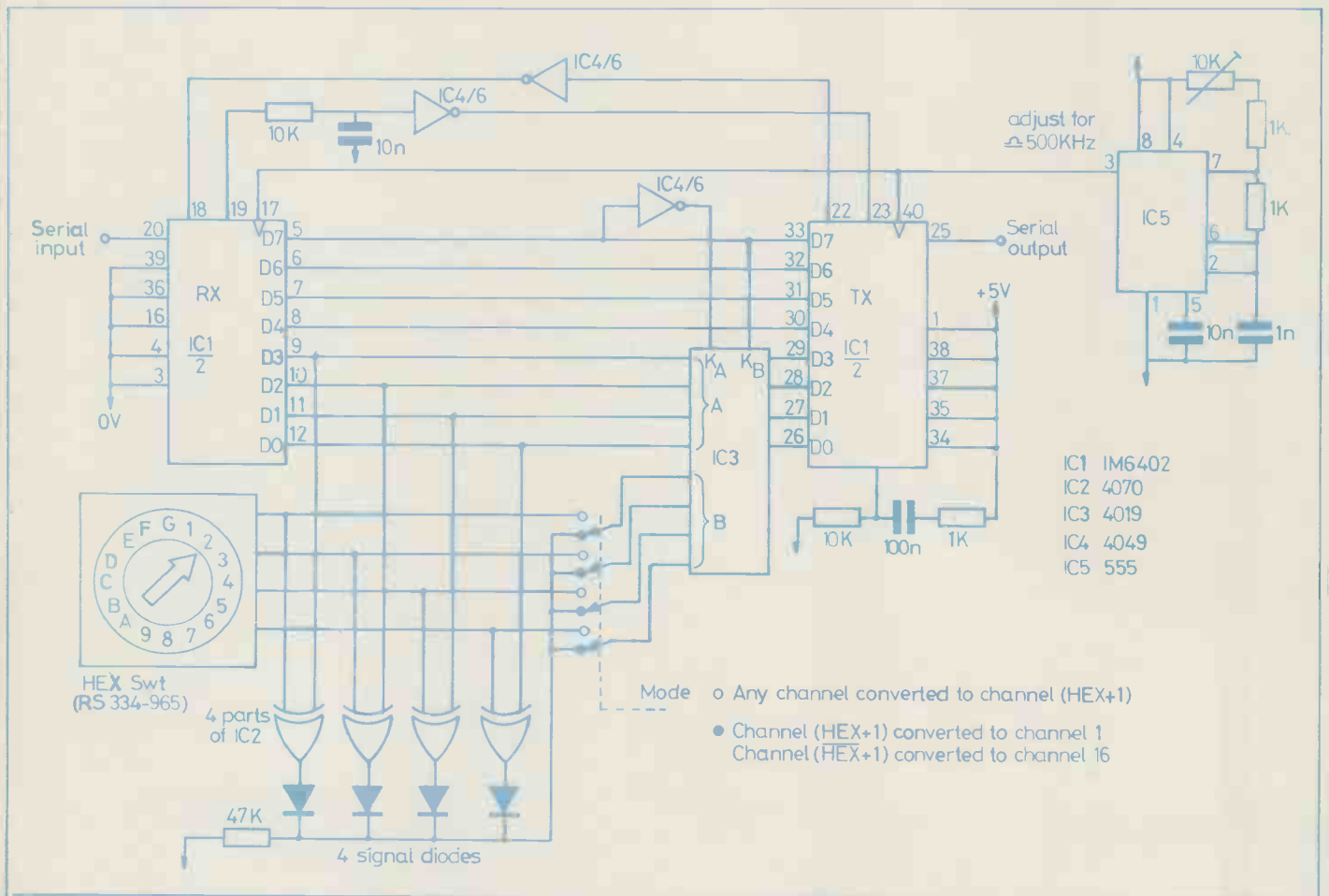
IC1 IM6402
IC2 4070
IC3 4019
IC4 4049
IC5 555

HEX Swt
(RS 334-965)

4 parts of IC2

47K

4 signal diodes

Mode o Any channel converted to channel (HEX+1)

● Channel (HEX+1) converted to channel 1
Channel ($\overline{\text{HEX}}$+1) converted to channel 16

SIGHT READING

A book that aims to clarify the workings of Yamaha's DX7 comes under the scrutiny of E&MM's resident FM expert. *Jay Chapman*

We all know the feeling. You've got a specific, long-standing need that has to be satisfied, and as soon as something that might do the trick becomes available, you make a bee-line for it. It isn't exactly what you want, of course, but it'll do something approaching a good job. Such an object is *Yamaha DX7 Digital Synthesizer* by Yasuhiko Fukuda. It isn't really what you want because it's not the answer to all – or even most – of your pleas for help on programming the DX7. But you'll probably buy it anyway because the market isn't exactly overflowing with books on the subject of the DX7, and every little bit helps.

Contents

It's a large-format book of 139 pages. It's imported from Japan and costs £7.95 in the UK. The cover is strengthened and protected by a plastic coating and should stand a fair amount of wear and tear. And the outside third of most of the pages is given over to diagrams illustrating points made in the text: this is generally a good idea, since if you're trying to follow a sequence of parameter changes described in the book, there's a visual confirmation of the DX7 display at regular intervals along the way.

Now for some details. The book's contents are arranged into three chapters and a five-part Appendix followed by a 'Special Introduction to DX7 owners in the USA' (*just what you've always wanted, eh? – Ed*). Chapters 1 and 2 deal in some detail with the material covered by the DX7 Owners' Manual, ie. the concept of FM synthesis, the use of Operators, and a parameter-by-parameter exposition of what all the front panel keypads do.

Chapter 3, meanwhile, is about 'Making Original Sounds' and *should* be the book's crucial element. It consists in the main of examples of how to create voices, with comments on each voice's most critical programming details.

Part 1 of the Appendix deals with the mechanism the DX7 uses to generate sinewaves, and discusses the mathematics of how Operators modulate each other to produce complex output waveforms. Part 2, which is all of one page long, suggests possible solutions to mis-setting controls, or what to do if playing your DX7 results in no sound output whatsoever. Part 3, also one page in length, deals with

the use of a Yamaha KX1 Remote Keyboard to control a DX7 (or two, or three), and this is followed by Part 4, which takes two pages (gosh!) to look at the DX7's implementation of MIDI. Part 5 (pages 99 to 110) lists the parameter data for the 32 internal voices supplied with the international (read 'non-USA') version: the parameter data for the US model's voices forms half of the 'Special Intro' mentioned earlier.

Problems

The main problem with *Yamaha DX7 Digital Synthesizer* lies somewhere between Mr Fukuda's ears and the translator's inability to convert a sow's ear into a silk purse. It's difficult to say precisely who's been responsible for what, but whatever the causes, the text is often misleading for reasons of inaccuracy, over-simplification or generalisation. Fukuda has a tendency to assume everybody's grasped a programming point after he's presented a shallow and ineffectual explanation decorated with a couple of examples. That's why Chapter 3 fails to live up to its promise.

Perhaps I'm being over-critical, but a technical reference book of this kind simply shouldn't contain the sort of linguistic ambiguities that abound in Fuk-

uda's epic. The author – or alternatively, his translator – insists on using 'tickles' instead of 'modulates', and talks of 'a radical algorithm': does this mean some Operator configurations are more politically active than others? I hope not. And as for 'a confused voice'... well, words fail me. Confused readers are an altogether more likely prospect.

I thought I'd give what for me is the highlight (lowlight?) of the book a paragraph of its own. See what you make of this little gem:

'The DX synthesizer operates tone by increasing harmonics, and if the output level of Modulator which seems suitable is kept unchanged, even in the range of higher tones, and no change is made in the degree of tone change, then the tones in the higher range become unpleasant.'

A chapter or two of that nonsense and your brain starts pleading for mercy, believe me.

Positive Points

On the bright side of things, Chapters 1 and 2 (as well as other parts of the book) do give blow-by-blow accounts of the use of all editing and function keypads which DX7 newcomers will welcome as a supplement to the Owner's Manual and the Dave Bristow booklet supplied free of charge by Yamaha in the UK. And if you aren't averse to the idea of applying a bit of mental perseverance, you'll find that Fukuda passes on any number of worthwhile hints to the would-be DX programmer: some of the information hasn't, as far as I'm aware, appeared in any other form anywhere else, and this makes interesting reading even if you've written a seven-part magazine story on how to get the best out of Yamaha's FM poly...

So we end up pretty much where we came in. Even given the occasional unforgivable error and what I can only describe as a mildly eccentric interpretation of the English language, this book deserves to be on every DX owner's shelf – at least until somebody comes up with something better. And yes, I do intend buying a copy for my own use. ∎

Yamaha DX7 Digital Synthesizer is available direct from Music Sales Mail Order, Newmarket Road, Bury St Edmunds, Suffolk, IP33 3YB.

Powertran
BBC MIDI Interface

A build-it-yourself MIDI interface for Powertran's MCS I sampler and the BBC Micro has now been made available. In the first of a two-part feature, we look at how the hardware is put together and the design thinking that lies behind it. *Jim Grant & Tim Orr*

If you've been following previous articles on the design and construction of the Powertran MCS1 MIDI Controlled Sampler (and if you've read through every episode, congratulations), it will no doubt have sprung to your attention that in amongst the machine's plethora of switches lies one labelled 'BBC'. However, the value of this switch will not become apparent until you make appropriate use of the similarly-labelled socket at the rear of the MCS1, since without some sort of valid connection, pressing the button will have precisely no effect at all on anything. True, there's software within the MCS1 that makes digital communication over a serial data bus a realistic possibility, but first of all, you need a suitable interface.

As luck would have it, the device in the accompanying photograph is just such an interface. It has three functions: to receive MIDI data, to send MIDI data, and to communicate with the MCS1 using non-MIDI codes.

Figures 1 and 2 should give you some idea of how the interface can be used either as standard MIDI-to-computer link or as a device capable also of non-MIDI conversation with the MCS1, which is itself MIDI-controllable, of course.

The first example shows the interface as a stand-alone link between the BBC Micro and the Wonderful World of MIDI. Figure 1(a) shows how the unit can be connected to the hardware on the outside world if circumstances (and finances) allow, while Figure 1(b) shows how the interface is configured internally for this particular application.

In contrast, Figures 2(a) and 2(b) show the external and internal connections necessary for using the interface in conjunction with the MCS1. This arrangement facilitates not only the generating of MIDI codes within the BBC and their subsequent transfer to compatible instruments, but also the saving and loading of sounds (using Powertran's own custom-developed software) between Beeb and MCS1, this data being transferred in a non-MIDI format.

Figure 1(a). Possible system arrangement using the Interface as a stand-alone link unit between the BBC Micro and MIDI instruments.
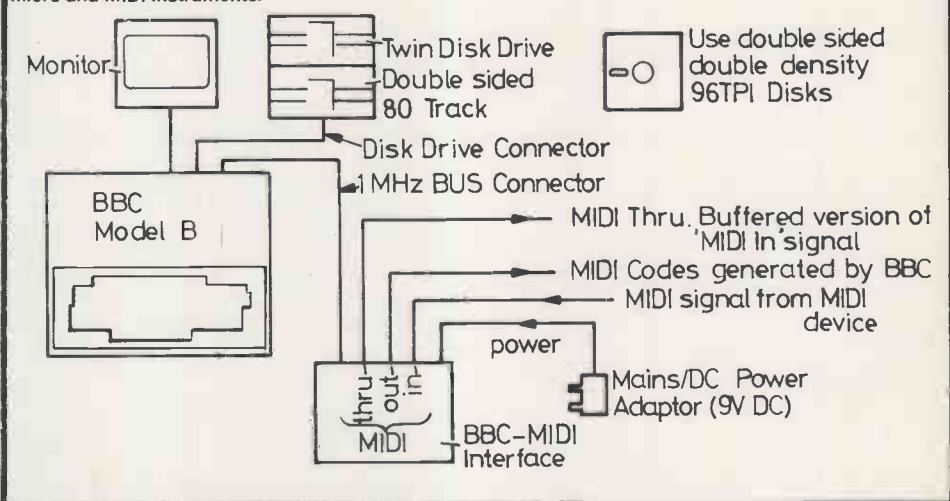
Figure 1(b). Internal configuration for stand-alone operation.
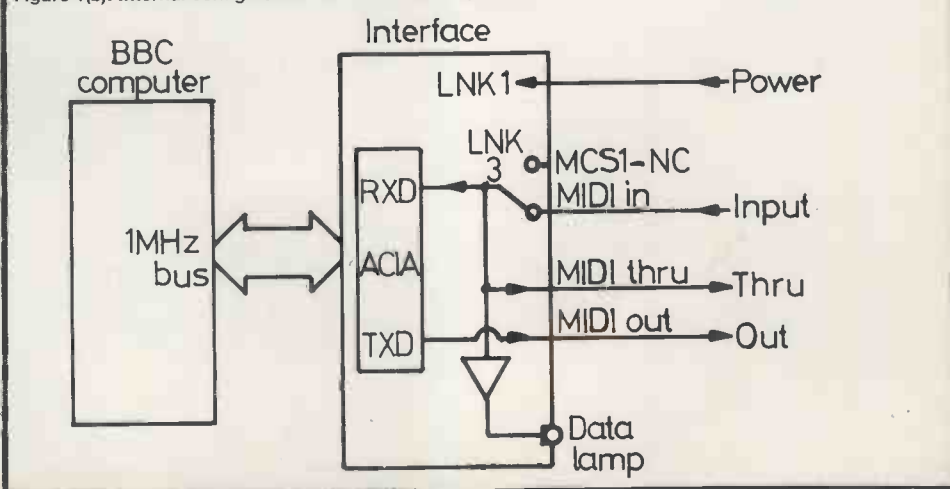
Figure 2(a). Possible system arrangement using the interface as a link for both MIDI and MCS1 data to and from the BBC Micro.

Monitor

Twin Disk Drive
Double sided
80 Track

Use double sided
double density
96TPI Disks

Disk Drive Connector

BBC
Model B

1MHz Bus
Connector

Optional for MIDI test
BBC In/Out

MIDI In

MIDIout
MCS1

BBC-MIDI
Interface
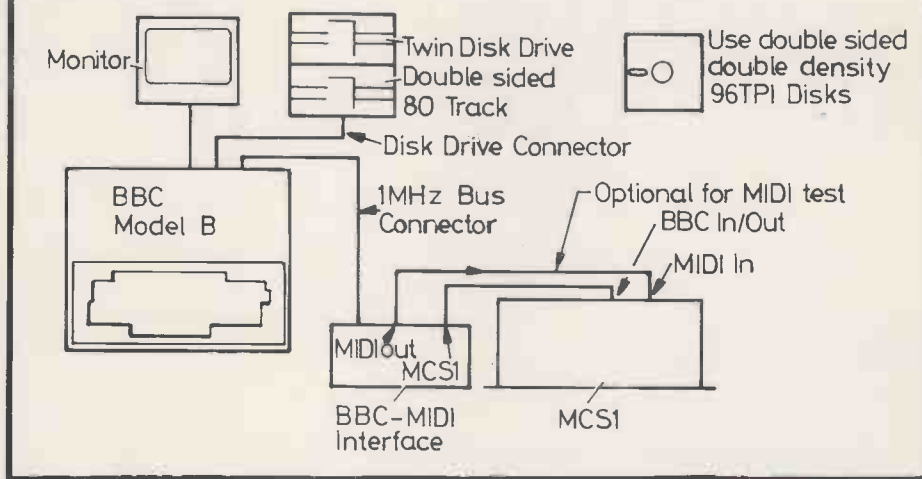
MCS1

Figure 2(b). Internal configuration for MIDI and MCS1 operation.

one cable

BBC
computer

Interface

Power

MCS1

1MHz
bus

ACIA

LNK
3

RXD

MIDI
in – NC

MIDI
thru–NC

TXD

MIDI
out

Data
lamp

D Load
D Save

BBC
in/out

MIDI
in
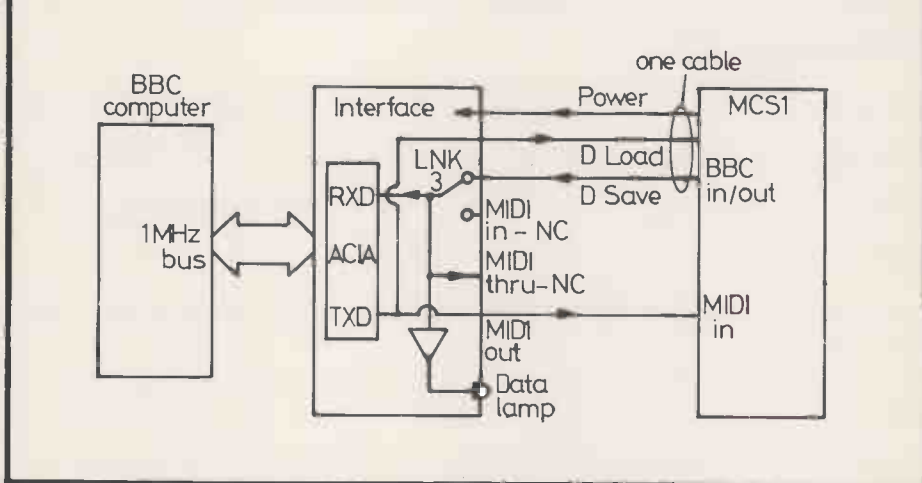
Links & Options

Luckily, the BBC Micro's design makes interfacing fairly easy. There are three pages (each of 256 bytes) called FRED, JIM and SHEILA, in which memory-mapped inputs and outputs can be placed. And the process of decoding memory locations within these pages is simplified by Acorn having provided a Page Select signal for each of them.

Figure 3 shows the circuit diagram for the interface, and a quick glance at it should tell you that the design can be divided into three main sections: the memory decoding, the ACIA that actually generates and receives serial data and the MIDI buffering.

Memory decoding is handled by IC502, and makes use of a signal labelled NPGFD: this is the Page Select for FRED, which lies from FD00 to FDFF in the BBC's memory. Link 5 is provided to map the interface into either JIM or FRED (FRED is preferable). Using NPGFD and address lines A4 to A7, IC502 slices FRED into eight memory location blocks, from which Link 4 selects one: again, the choice is yours, but in the normal course of events it should be in position Yo. If Link 5 is at position B-A (FRED) and Link 4 at position Yo, the ACIA is mapped in the memory of the BBC at two locations. In addition, the RW and RS (Register Select) pins on the ACIA take the number of on-chip registers to four.

Link 2 is an option that lets you use an external clock to synchronise data transfer at high transmission rates. This may well prove useful, but the usual connection is B-C. Finally, Links 1 and 3 determine whether or not MCS1

communication is intended in addition to data transfer along MIDI. If you want to power your interface direct from an MCS1, you should set Link 1 to A-B, while using the interface in stand-alone MIDI configuration will necessitate an external 9V power supply, power being regulated by IC505: Link 1 should be connected B-C.

But you've got to select the data receive line as well as the power routing. Link 3 set at B-A allows the interface to listen to an MCS1, while a B-C connection hooks in the optoisolator circuit for MIDI reception. You'd be right in thinking that all that adds up to an awful lot of connections for such a small circuit, so Table 1 provides a summary of available options.

Construction & Testing

Construction should begin, as always, with the insertion of the links, closely followed by the fitting of the passive components, namely resistors and capacitors: take care checking the polarity of electrolytic capacitors. Bend the legs of the 5V regulator, IC505, and insert it into the PCB so that the fixing hole aligns with the hole in the board. Slide the heatsink under the regulator and bolt both to the PCB before soldering the IC legs: this should ensure a stress-free joint. Check the orientation of both diodes and transistors before insertion, and fit the connectors and remaining large items last of all.

There are a few more points to be made about putting the hardware together, and the first couple concern the interface unit's casing. In addition to referring to Figure 4, you should also bear in mind that the case itself is

asymmetric, and that the PCB and panels will only fit properly if they're positioned the right way round. A simple point, perhaps, but worth making all the same.

Secondly, we come to the problem of the external power supply. Not everyone will need one of these, of course, but if you're one of those that does, reference to Figures 5(a) and 5(b) should help you out when it comes to wiring.

Finally on the construction front, Figure 6 gives the lowdown on how to assemble the CN505 34-way connecting cable.

Since the interface unit does nothing whatsoever in isolation, it defies most conventional test methods. However, if the unit has been correctly assembled there's actually little scope for things to go wrong, and a few simple tests should shed light on any faults that might exist.

If you're using the interface in conjunction with an MCS1, plug the unit into the MCS using a five-pin DIN cable: this will also power the interface. Check to see that +5V is present on all ICs (see power pin box, Figure 2). Alternatively, if you're powering the unit from an external mains adaptor, check that +9V is available on the input to IC505 and that +5V is generated at its output. The Power LED will illuminate if power is present.

The next step is to connect the interface to the Beeb's 1MHz bus. If the system is operating correctly during a D S A V E (part of the software operating procedure), data will be seen on connections D0 to D7 of the ACIA (IC500). In addition, FRED should go low when reading or writing from the ACIA.

If you're unlucky and a fault condition occurs, the first thing to do is to look for open circuits or shorts between adjacent tracks on the PCB. Short circuits on the PCB often manifest themselves as illegal logic levels (eg. between 0.5V and 2.5V) during testing.

Plugging in a scope should bring a couple of square waves from different pins on IC503, viz a 1MHz square wave at pin 3 and a 500kHz one at pin 2.

Note also that the interface's MIDI Out signal is normally high, going low when active. This can be seen at pin 11 of IC501, and can be tested by playing the f1 to f8 keys on the BBC's keyboard in the context of the software's MIDI Test page. MIDI In, on the other hand, can be tested by injecting a MIDI signal source or digital input: if all is working as it should do, the output of the optoisolator will be an echo of this signal with a delay of a few microseconds. As a quick visual indication that all is well, the built-in Data LED lights up when data from either the MCS1 or the MIDI In connector is being received by the interface.

Data Communication

Let's sort out a couple of bits of theory. The transfer of both MCS1 and MIDI data requires conversion from serial to parallel format and back again in order to take place at all. This has to be done at constant 31.25Kbit/sec rate, with stop and start bits to synchronise transfer: this is the ACIA's function in life, and communication between it and the BBC Micro is via the latter's 1MHz bus. Special timing requirements also have to be met, and these are catered for by IC501 and 503.

Mind you, before the ACIA can either receive or transmit data, it has to be initialised by the BBC sending two bytes to its Control register. The bytes in question are 7&FD00=03 and 7&FD00=&55. The first resets the ACIA and clears the flags in the Status register, while the second performs a number of functions including setting the clock divide to 1/32 and choosing one start bit, one stop bit and eight bits of data.

The Interrupt flag is disabled, but can be

BUILD IT

Figure 3. BBC-MIDI Interface circuit diagram.

Table 1.

LINK SELECTION			DESCRIPTION
	MCS-1 INTERFACE	STAND ALONE MIDI	
LINK 1	A-B	B-C	POWER SOURCE
LINK 2	B-C	B-C	EXT. OR BBC CLOCK
LINK 3	A-B	B-C	RECEIVE DATA. MCS1 OR MIDI
LINK 4	Yo *	Yo *	MAPPING WITHIN
LINK 5	A-B (FCCD) *	A-B (FCCD) *	PAGE SELECT

SHOWN SOLID

* USER VARIABLE

THE 1MHZ BUS

POWER PINS

	+5v	0v
74LS138	16	8
74LS32	14	7
74LS74	14	7

FCCD → NPCFD   FCC0 → FCFF
FCCD → NPCFD   FD00 → FDFF

IC500 6850 ACIA

IC502 74LS138

IC503 74LS74

LAST Nos.
IC 500 - 505
PC 500 - 508
C 500 - 504
T 500 - 501
L 500 - 501
D 500 - 501

IC501 'MIDI THROUGH DATA'

6N138

MCS1/BBC INTERFACE

MIDI OUT

MIDI THROUGH

MIDI IN

OUTSIDE VIEW OF PCB DIN SOCKET. * MIDI PLUG IS MIRROR IMAGE.

PCB →

E&MM   APRIL 1985

▷ enabled simply by sending the second byte as 7&FD00=&95. This causes the IRQ line to be pulled low on receipt of serial data, upon which the BBC whizzes off into an interrupt routine in search of the device that caused that interrupt. It'll ignore the ACIA unless you make specific use of the user interrupt vector at &206 and &207 to patch code that'll service the interrupt request.

With all that safely out of the way, the interface is now in a position to deal with either MIDI or MCS1 data, depending on how its links have been arranged.

Basically, there are two kinds of bytes – Status and Data. Instruments can distinguish between the two by examining the Most Significant Bit, which is set for Status and clear for Data bytes. The easiest way to view the situation is to see Status bytes as commands that tell the instrument what to do with (very often) the help of additional Data bytes. Examples of this sort of operation are MIDI Note On and Note Off commands.

The interface's reception of MIDI data is performed by checking bit 1 of the Status register (reading location &FD00), and if this is set, reading the received serial data at &FD01. This process can only be achieved successfully by an assembler program, as BASIC is too slow to keep pace with MIDI data transmission rates.

MIDI Data

Well, just in case you've been on safari in the Gobi Desert for the past couple of years, the above acronym stands for Musical Instrument Digital Interface, and is rapidly becoming *the* interfacing standard in the world of electronic musical instruments. Physically, it's five wires terminating at each end in a five-pin DIN plug that connects two or more MIDI devices together. Electronically, it's a communication medium that allows machines of varying origins to pass on to each other common information such as pressed keyboard notes, pitch-bend and other parameters. MIDI instruments from the same manufacturer can also communicate more specialised information to each other using what's known as System Exclusive data.

The information itself is sent serially along the connecting cable, with special signalling bits that mark the beginning and end of individual segments of information called bytes. The standard bit format is shown in Figure 7.

Each byte sent has a specific meaning to the receiving machine, which decodes it and then acts accordingly.

Let's take the Note On and Note Off commands mentioned earlier as examples of this process at work. Both commands involve the sending of three bytes, the second of which represents the keyboard key number and is represented as a number between 0 and 127 in both cases: 60 is Middle C. Note On's first byte is Status byte 90 hex where as Note Off's first is 80 hex, and the third byte, similar to the second in that it's also a value of between 0 and 127, represents key-down velocity in the case of the Note On command and key-up velocity in the case of Note Off.

In either case, the first byte sent is a command that tells the receiving MIDI instrument what to expect along the MIDI lines in the (very) near future and what to do about it when it comes. It's worth noting that the Most Significant Bit of this byte is always a 1, whereas all the other bytes have their MSB clear. It's this distinguishing feature that enables the receiving machine to sort out exactly which bytes are commands (or Status bytes) and which are merely supplied data for commands, such as which specific note to play or release.

Figure 4.

Plastic case has 4 solid pegs without central hole. Remove these with sidecutters. cut here

Figure 5(a).

+VE / ground / Plug body
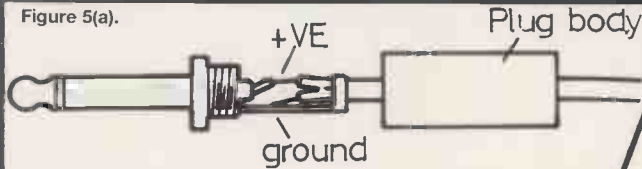
Cut off power supply concentric connector & fit 3.5mm jack plug

Figure 5(b).

+9V 0V
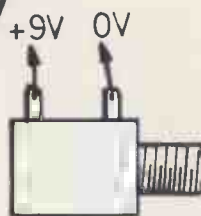
3.5mm jack socket CN500

Figure 6.

34 way 0.1"x0.1" solder transition connector

500mm (cable length)

34 way cable

34 way female transition socket / strain relief clip / polarising tab/bump

Farnell 609 3453
Farnell 145 069
Jermyn 3934 0000T
Verospeed 901 71368E
Verospeed 832 6254C

Roxburgh IS34-A01-83
Farnell 145005
Verospeed 901 71197F

LSB / MSB

D0 D1 D2 D3 D4 D5 D6 D7 Stop Bit

Start Bit

time
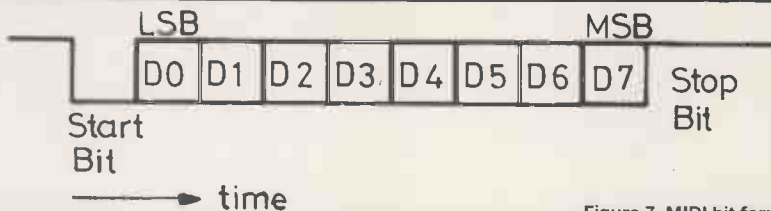
Figure 7. MIDI bit formatting.

Incidentally, you'd be right in thinking that some MIDI synths are incapable of producing keyboard dynamics from MIDI velocity information, but this is still sent in any case as part of the standard MIDI protocol.

One point that's crucial to bear in mind is that although we've so far viewed MIDI as a single two-way stream of data, it does in fact consist of 16 separate channels of information transfer. Any MIDI-compatible synth is capable of sending MIDI commands on any of these 16 channels simply by varying the number held by the four Least Significant Bits of the Status byte. So for instance, while Status byte 90 hex is a Note On command transmitted over MIDI channel 1, 93 hex is the same command sent over MIDI channel 3.

The process of receiving bytes is a little bit more involved, because MIDI machines vary in the way(s) in which they can receive MIDI data. There are in fact three standardised receive modes in general use. The first is Omni mode, in which the synth in question ignores channel information entirely and executes commands whenever they arrive. The second is Poly mode, in which a synth only acts upon MIDI commands if their channel number matches its own. And the third is Mono mode, in which the instrument's voices can be controlled individually due to their having been assigned different MIDI channel numbers.

Unfortunately, hardware manufacturers have been decidedly slow on the uptake when it comes to implementing Mono mode, and to date, only a handful of models (among them most of the polysynths in the SCI range and, apparently, the Casio CZ101 and CZ1000, though Casio themselves don't seem to be aware of the fact) feature this 'multi-timbral' facility...

MCS1 Data

Since MCS1 data is unquestionably a non-MIDI format, it requires its own brief explanation. Its transfer rate and stop/start bits are in fact identical to those of MIDI, but it differs in one respect important enough to earn it its own separate transmission bus. Each sound on the MCS1 occupies 64Kbytes of RAM which, if transferred to a BBC Micro down a 31.25Kbit/sec data bus, would take around 21 seconds to make the journey.

MIDI, remember, only allows data to be sent with the MSB clear, thus reducing the amount of information that can be sent in a single byte. Synth manufacturers overcome this by splitting each data byte into two four-bit nibbles and transmitting them as two separate bytes, but this is clearly not satisfactory for MCS1 data since it would double what's already a fairly lengthy sound dump time.

So that, in a nutshell, is why MCS1 data is transferred along its own communications bus (thus maintaining the eight-bit byte size) rather than down the MIDI bus under a System Exclusive command.

Next month, we'll look at the first piece of commercially-written software for the BBC-MCS1 combination, which allows MCS1 sounds to be dumped onto and retrieved from BBC disk, among other things. So stay tuned.

Further information on both the MCS1 and the BBC-MIDI Interface can be had from Powertran Cybernetics, Portway Industrial Estate, Andover, Hants SP10 3EM. ☎ (0264) 64455.

COMPUTER MUSICIAN

A s Dan Goldstein said in last November's *Comment*, 'our consumer population has taken to the home computer like no other public in the world.' And yet, if you cast your eye over the catalogue of companies going into liquidation and receivership, the UK's micro manufacturing industry has a tendency to insolvency that's also like no other in the world. The latest of these to hit the headlines is, of course, Acorn Computers. And like the big trees that grow from small acorns, who'd have believed that a company with a turnover of £54.9m for the six months up to last December could be so easily toppled?

So what went wrong, and more importantly, what are the lessons to be learned from this catalogue of disasters?

To start with, it's easy enough to look at the micro marketplace in the UK and see a clear two-tier structure developing. On the one hand, there's the business community that moves in the upmarket heights of IBM, Apple, Apricot and DEC. On the other, you've got the general consumerism of home and education, where the favoured machines cover a remarkably narrow band of prices between about £125 and £400. And as far as this is concerned, the general consensus is that the marketplace is now more or less saturated. It's hardly surprising, then, that companies like Oric and Dragon Delta, with machines that to all intents and purposes weren't hugely different to any of the other eight-bit micros on the market, found it tough going.

But Acorn were different. With a virtual monopoly in the educational arena as a result of the BBC's licensing agreement and the Government's generous educational discounts, the company was in an extremely cosy position. So much so that when he was asked why Acorn weren't reducing the price of the BBC Micro, Chris Curry said: 'Why should we? We're selling them as fast as we can make them.'

So for a couple of years, Acorn did very nicely, thank you. Unfortunately, just around the corner was an ambush of new machines like the Sinclair QL, promising the Earth and delivering rather less, but still showing rather effectively that £400 could buy you more than 120 chips, 32K of memory, a reasonable BASIC and a 'BBC' tag.

The Beeb was under threat, but the only form of retaliation Acorn had at their immediate disposal was a cheaper micro called the Electron: it sold well but not, it would seem, well enough.

The first message to come out of all this must be that resting on your laurels while accumulating turbo Porsches isn't very bright. Neither is having fisticuffs with your nearest and less than dearest, because any new micro firm has a hell of a job on its hands if it's to grab a share of consumer attention these days, and dents in your own street credibility don't help one iota. What's needed is foresight, imagination, and plenty of capital to invest in the new chip technology and operating system development essential for the new generation of micros. Acorn had the latter, but missed the boat when it came to the other, less financially-dependent qualities.

So far, the lateral-thinking spearhead is coming from the States in the shape of Commodore's Amega and Atari's ST range, machines that cast aside the home versus business argument and just perform impressively – with added benefits for the computer musician. Unfortunately, aside from a lot of hot air from Sinclair, there's no evidence of similarly forward-looking designs coming from the UK in the near future.

Still, once the jigsaw pieces have been put back together, at least Olivetti's virtual takeover should make a few Acorn heads a little less arrogant and a lot less complacent. At this stage, it looks as though the Italian company's plans are to make Acorn's products the worldwide educational norm: hardly forward-looking, I agree, but a pretty safe bet given the Beeb's pre-eminence in this respect.

We can only watch and see what happens next. ∎

David Ellis

Rumblings...

This month's round-up of all that's new in the world of computer music. *David Ellis*

Passport Designs have now updated their MIDI/4 software to 'MIDI/4 plus' status (amazingly logical, these Americans...). And they've also got a brand new package out called 'MIDI/8', which offers double the number of recording channels of MIDI/4. No kidding... Both items of software provide drum, tape, and MIDI sync, plus oodles of editing facilities including auto-correct, punch in/out (the previous version of MIDI/4 required punch-in recordings to be completed to the end of the track, which was hardly the most endearing of features), fast forward, and rewind. As Passport themselves put it, 'a complete recording, editing, and printing system'.

Both packages are available for the Apple II/IIe and Commodore 64, together with the all-important MIDI card (a cool $195 for either micro). Curiously, there's still no sign of Passport's MIDI products really breaking into the UK market. Some of the market resistance is probably not unconnected with the swinging effect of the declining pound on the cost of US imports, but more than any other music market, the UK also seems flooded with MIDI software at present, so perhaps market saturation also has something to do with it. Anyhow, for more info, contact Passport Designs Inc, 625 Miramontes, Half Moon Bay, CA 94019, USA. ☎ 415-726 0280.

Mimetics

No further news on **Mimetics'** efforts at updating the hardware side of the alphaSyntauri to report yet, but at Frankfurt they were splashing around review copies of some nice DX-oriented software for the Commodore 64, IBM PC, and Apple IIe. First off is the cryptically-titled 'Data/7', which 'is designed to expedite cataloguing of (DX7) preset information through the use of a high-speed storage and retrieval system'. Promo hype apart, it all seems very user-friendly (courtesy of plentiful graphics), and allows the user to load or save 32 voices between disk and DX7 in under two seconds.

Mimetics' other product is 'Performance/7', which they claim 'provides access to 288 voices instantly'. Which may or may not thrill you to kingdom come. Both programs cost $125, and also require the relevant MIDI computer interface (Passport MIDI card for the Commodore 64 or Apple II, Roland MPU401 for the IBM PC or Apple II). Mimetics Corp invite enquiries to be directed to PO Box 60238, Sta. 'A', Palo Alto, CA 94306, USA. ☎ 408-741 0117.

Caged Artist

...definitely most bizarre name of the month – the US company behind a software package called (almost equally bizarrely) 'DX-Heaven', which claims to provide DX7 and 9 owners with the ability to store 1000 DX sounds on a single Apple floppy disk. On-screen editing makes life easier when it comes to sorting out which parameters do what, and the whole shooting match then gets sent down to the DX keyboard *via* (you've guessed it) a Passport MIDI Interface. The price of this potentially juicy slice of FM heaven is $59.95, and Caged Artist Productions can be found locked away at 64 Griggs Road, Brookline, MA 02146, USA.

Jackintosh

Rumours turned into reality this week when **Atari's** Jack Tramiel appeared with his high-spec 520ST micro on Channel 4's *4 Computer Buffs* programme. Mind you, unless you'd been forewarned, you might well have switched off before the rotund Tramiel even so much as appeared, what with the general air of ineptitude emanating from interviewers and interviewees alike. Still, at long last, we saw the proof behind the hype that accompanied the Las Vegas Consumer Electronics Show, and evidence that refuted Sinclair's comment that 'Atari doesn't have a computer – it has a box... we remain unconvinced by Atari'. Indeed, if I was in Sinclair's boots, I'd be getting worried. The ST range really does deliver the goods, and it also looks set to be delivered on time – mid-May, in fact.

The projected package deal of the top-of-the-range 520ST (that's the 512K version) plus a colour monitor and 1 mega-byte disk drive is likely to sell for between £800 and £900, and if you bear in mind that the equivalent (though still only monochrome) Apple Macintosh system costs well over £2500, that's exciting by any standards. When you add on the ST's wonderful Mac-type Digital Research GEM operating system, the built-in MIDI port, the MIDI software that companies are already developing for it, and the add-on keyboard and digital synth that Atari have produced for it, there's really not much doubt left in my mind about Tramiel's claim that, 'by 1986, we will have captured 25 per cent of the UK home computer market'.

If you ask me, he deserves it.

Stop Press!

David Ellis, moments before being knocked over by a feather (out of shot, top left).

...just launched: **'DIMPLE'**, standing for Diminished Intelligence Music Production LanguagE, which turns the Acorn Music 500 into an authentic sonic replica of the average British typing pool, complete with Olivetti typewriters, spark-jet printers, grunging disk drives, and Italian tea ladies singing arias from *Il Trovatore*... *Sinclair C5 User* magazine starts an obituary section – Sir Clive launches a DIY embalming and funeral kit... **Prince Harry** astonishes the computer gaming world with his megagame 'Elitism', programmed on his dad's BBC Micro – British Telecom are rumoured to have paid £250,000 into the royal coffers for the royal rights... **Acornsoft** finally publish *Creative Sound on the BBC Micro* – David Ellis is knocked over by a feather... ∎

Datel Sound Sampler

Hardware and Software for Sinclair Spectrum

As sound sampling becomes as popular a buzzword in the computer industry as it is in the world of music, a few enterprising companies are setting about designing sampling packages based round home micros. The Datel is one of those packages. *Simon Trask*

I f you're anything more than a half-regular E&MM reader, you'll be aware that sound sampling is at last about to enter into what we hope will be a meaningful relationship with average musicians, whoever they may be. You might also be aware that the home micro industry is also about to come under the spell of sampling, and it's for this reason that the means by which sampling is made available will take various forms – the dedicated sampling keyboard (eg. Ensoniq's new Mirage), the sampling synth expander (eg. the Powertran MCS1), and the computer-based sampling system, like the one reviewed here.

Datel's Digital Sound Sampler is a modest little system based round the 48K Spectrum. It acts as a reasonable introduction to the world of sound sampling, but dreamers who think £50 and one of Sinclair's babies is going to buy them a Fairlight in miniature had better stop reading – now.

The total package comprises the DSS 'black box', a cassette containing four programs, a small microphone, and three A4 pages' worth of 'instructions'. The DSS box plugs into the Spectrum's expansion port (where else?), and has one input that can be mic or line and one output that'll happily drive an external amplifier or hi-fi system. The bad news here is that the Gain and Feedback controls manifest themselves in the form of tiny screws recessed into the DSS casing – the Feedback control is particularly awkward to access. A Line Test function in the first program feeds the input directly to the output so that you

can set a suitable level, but the microphone wouldn't do justice to an early Osmonds rehearsal session, let alone a sound sampling system – definitely a candidate for replacement.

Software

This consists of four programs packaged on one cassette, and these are titled Effects, Keys, Sequencer, and Games Speak. None takes more than a minute to load, and all loaded reliably from the review copy. Each program is conceived to run independently of the others, so each one contains its own sample record facility. It's a shame there's no option to save/load a sampled sound which could then be transportable between programs, as this would be an elegant way of building up a library of samples for future playback and manipulation.

So what do these four programs allow you to do, exactly? Well, Effects allows you to manipulate the sound you've just sampled, the facilities available being reverse play, riser (in which the sample is played through an ascending pitch sequence), chop play (in which the sample is played in progressively shorter segments), and echo (with adjustable delay: fiddling with the Feedback control alters the amount of the effect). A further option allows four sounds to be sampled and then played on keys 5, 6, 7 and 8 of the Spectrum keyboard: a possible use for this might be to sample four drum-type sounds which could then be played as a mini drum kit. The play and reverse

facilities allow a single sampled sound to be played at various pitches using the 1-9 keys on the Spectrum, while the draw facility simply draws a frequency plot on the monitor screen. Single or repeat playback can be set for either riser, chop play or echo, and maximum sample record time is no less than four seconds.

The Keys program presents the user with an on-screen, one-octave keyboard which can then be 'played' using the upper two rows of the Spectrum keyboard. A nine-octave range is possible, though the currently-selected octave can only be changed by returning to a command level. Still, if you really want to say 'Hello' in tones ranging from a foghorn to a chipmunk, now's your chance.

Sequencing

The Sequencer is entirely monophonic, pitch is selectable over a four-octave range, and 16 note durations are possible: the overall tempo of the sequence can be altered from the menu page. Sad to report, the Sequencer is a rather primitive step-time affair with the ability to store up to 1000 notes – though I can't imagine what twisted mind would glean enjoyment from inputting a sequence of that length using a system as tortuous as this one.

Each note is input as note-name followed by octave number, and this in turn is followed by a duration value from one of those specified. One failing that a lot of people are going to find annoying is the fact that the program doesn't allow for flats, and neither does it accept B# or E# (or at least, the review copy didn't). And although the instructions state that a value of 1.5 represents a dotted semi-quaver, the program will only accept integer values – most odd. The Compose facility only allows note input to start from the first note, and the Edit facility, such as it is, doesn't allow notes to be added to a sequence, so your magnum opus has to be input in one go. Editing has to begin from note 1 as well, so if all you want to change is note 499, you're going to have to be of a very patient disposition.

Frankly, there's no excuse for such a low-grade program as this, especially when the shortcomings listed above would be so easy to sort out.

The final program, Games Speak, is a utility that enables as many as eight

sampled sounds to be incorporated by the user into his or her own programs. Each sound can be up to half a second's duration, or you can dedicate all four seconds of memory time to just one sample, should you so wish. Playback time is of course dependent on playback pitch, as different pitches are created by reading the sample out through the DAC at differing rates (pretty much the standard procedure, this). Datel have provided two sample-manipulating routines, one producing a rising pitch effect and the other a random pitch effect.

Thankfully, full details are given in the accompanying instructions on how to incorporate Games Speak into your own programs, and on how to make use of the sample record and play routines.

Quality

I'll be quite candid with you. The sampling is not of a quality that would make you rush to a Portastudio, let alone a professional multitrack facility.

On the software front, only the I/O routines governing sample record and playback are in machine code: the rest of the programs are written entirely in BASIC. None of the programs is protected, and Datel don't seem to mind if you tinker around with the code and make use of their sampling routines: indeed, the nature of the Games Speak

program seems to have been formulated for this very purpose.

A number of software bugs manifested themselves in the review copy, though Datel assure me that these will have been corrected by the time you read this. The company have apparently had plenty of feedback from users, and are willing to listen to any suggestions/criticisms they might have, which is always a good sign.

Anyway, it's because most of the code is in BASIC that the user interface side of things is a bit sluggish, a fact which I found irritating after a while. The Keys program is particularly badly off in this respect.

Personally, I'd have preferred all the programs (with the possible exception of Games Speak) to be integrated into one, coded in machine code for speed and compactness, and given a more carefully thought-out set of facilities.

Conclusions

I know this is going to be a disappointment to many of you, but I can't really say in all honesty that the Datel Sound Sampler is suitable for serious musical use. As a package, it's crying out for some decent software to do its hardware justice, but sadly it appears that Datel themselves have no plans for updating this area, so you'll have to make any programming improvements yourself.

Luckily, the system as it stands now should allow plenty of scope for such tailoring, but that's not really the point.

What Datel *are* planning is a hardware interface that'll allow (initially, at least) the Keys program to be controlled from any Casiotone keyboard. This sounds to me like a pretty shrewd move, since it'll no doubt save a lot of musicians the duplication of keyboard hardware that would have been necessary had the company opted for a self-contained, dedicated chromatic pitch-controller of their own. The interface shouldn't cost more than about £25 when it becomes available within the next month or two, so it should be a neatly viable option.

Commodore owners can also prepare themselves for the arrival of a 64-based version of the system, with some new software written in machine code, that'll also be available in the near future.

Dedicated computer buffs may well find the DSS' particular combination of facilities attractive, but the system simply isn't user-friendly enough to warrant serious consideration by micro-equipped musicians. And unless you're keen to develop things further by writing your own software, there's not much you can do about it.   ∎

Further information from Datel Electronics, Unit 8, Fenton Industrial Estate, Dewsbury Road, Fenton, Stoke-on-Trent. ☎ *0782 273815*

BeeBMIDI 8

The second of a two-part look at DX7ED, E&MM's own editing program for the Yamaha DX7 and BBC Micro. *Jay Chapman*

What you're about to read (or at least, I hope you are) is a discussion of the DX7ED screen 'pages' and related facilities not covered last month, followed by an explanation of some of the program's procedures. Listings of these procedures are included at strategic points along the way, but the program as a whole is too large to list in full – there has to be some space left in the magazine for other things, after all.

Pages

Figure 1 shows the Main Menu page that appears when the program is first run or whenever one of the major pages is exited. The pages for the LFO, Modulation Controllers, Algorithm Selection and Envelopes were discussed last month.

On selecting the Operator page, you'll come to an intermediate page which shows a diagram of the current algorithm with the current Operator highlighted – this is similar to the display in the centre of the Copy page in Figure 4, in which Operator 4 is highlighted. You can then change the currently-selected Operator.

The current Operator is used by default whenever an Operator-related action is required, so that manipulating Operator parameters on different pages (Operators, Envelopes, Scaling and Copy) affects the same Operator automatically. Once the current Operator has been selected, you can exit back to the main menu or move on to edit the selected Operator's parameters (except envelope and scaling parameters, which each appear on separate specialised pages). The Operator editing page is shown in Figure 2.

The Coarse and Fine pitch parameters result in a Ratio or Fixed pitch display dependent on the Mode setting at the bottom right of the page. In Ratio mode, one more digit is displayed on screen than is displayed on the DX7 itself, so that Ratios such as 0.500 and 0.505, which would both display as 0.50 on the DX, can be seen to be distinct from each other. Note that the Detune bar graphic gives a helpful visual indication of the amount and direction of the detuning taking place.

The Keyboard Scaling page (Figure 3) should help quite a few DX users come to terms with both Level and Rate scaling. The vertical axis of the Level Scaling Curve graphs – rising from the C2 key on the 'DX7 keyboard' shown on the sample page – shows where the Break Point is for level scaling. If the Break Point parameter is edited, then this axis, along with the Left and Right Curves drawn on the

page, move so that the Break Point and Curves are positioned correctly with respect to the keyboard image. If the Left or Right Depths

or Curve types (-LIN, -EXP, +LIN, +EXP) are altered, then the new curves are drawn on screen immediately. You can't get better
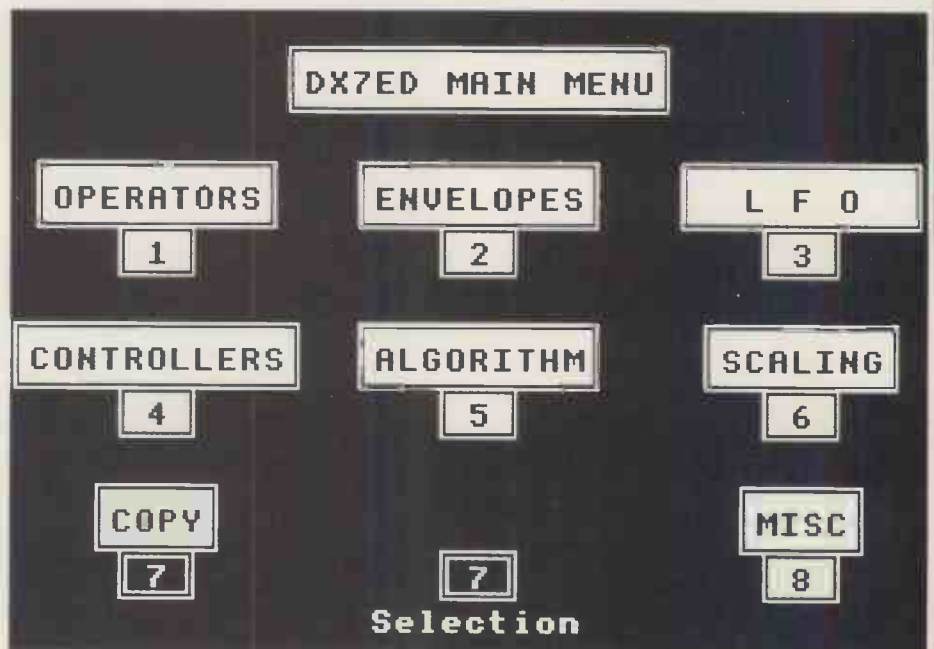
Figure 1.
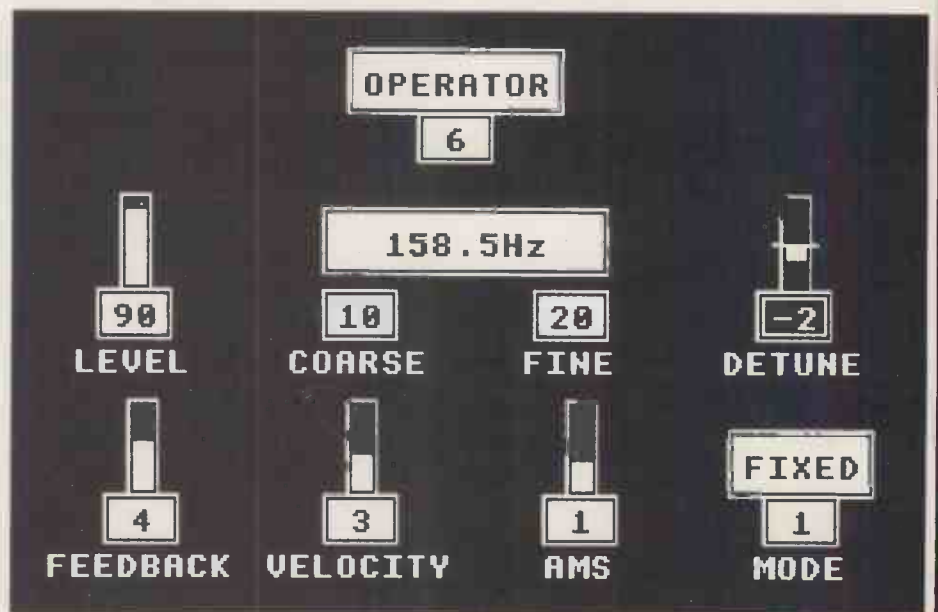
Figure 2.

feedback than that.

In the sample page shown in Figure 3, the effective level of Operator 6 will increase quite quickly below C2 (Left Curve = +LIN), and will fall to nothing with increasing speed from about F3 to the top of the DX7's keyboard (Right Curve = -EXP), as can be seen from the curves shown explicitly on the screen – or in this case, printed with widely-spaced dots. The other dotted line represents the effect of the Keyboard Rate Scaling parameter: the slope of this line corresponds to the severity of the Rate Scaling at any one time.

The Copy page (Figure 4) corresponds to, but is more powerful than, the Envelope Copy facility available on the DX7 itself. Not unexpectedly, this facility allows you to copy one envelope to another, and both source and destination envelopes are selected with reference to the diagram of the current algorithm.

It's also possible to swap two envelopes, or even to copy/swap *all* the parameters of any two Operators. If you can't quite imagine just how mind-blowingly useful this facility could be, here's an example. Let's say you'd set up Operators 1-4 as the basis for a new voice and then decided you needed to modulate Operator 2. With DX7ED, all you have to do is swap OP1 with OP3 and OP2 with OP4 – which takes all of five seconds – and then start work on Operator 5. Simplicity itself...

The last page (Figure 5) has the title Miscellaneous, which, I have to admit, is decidedly unhelpful. However, the page itself is anything but, as it deals with all the DX7 parameters that haven't already found a home elsewhere. Note that editing the first parameter to select Monophonic or Polyphonic automatically updates the display of Portamento Mode, which should make the interaction of these two parameters a little clearer to those still in the dark.

You'll probably have noticed that the Voice Name in the example shown is a little on the weird side. This is quite deliberate: I wanted to emphasise the point that with DX7ED, you're no longer restricted to the characters you can input on the stand-alone synth. This software allows you to include lower case letters and special symbols, and all these will subsequently be displayed on the DX7. Well, everything I've tried so far has, anyway.

You want ergonomic efficiency? You got it. Entering the characters that make up the Voice Name is the *only* DX7ED activity that necessitates you moving both hands. All the other editing functions can be performed using only one hand, leaving the other free to try the sound out on the DX keyboard as you go along. And since you don't even have to enter numeric values when editing parameters, there's no need to take your eyes off the screen. In other words, creating sounds with DX7ED is as quick and painless a task as you'll find in the area of music software.

Procedures

The listings shown here are parts of some of the DX7ED program's routines, the intention being to illustrate how the program works. The routines are written in BBC BASIC, but production versions of the program will be written in Pascal and distributed in an 'executable only' form.

The basic (no pun intended) idea used in the program is that of displaying sets of related parameter values in labelled boxes on the screen for the user's information. If the user selects a particular parameter for editing, its box 'background' is changed so that selection is graphically confirmed: then when the user enters commands to change the value, the content of the box is updated to reflect that change.

Now, if a parameter represents nothing more than a simple numeric value within a range of allowable values – examples are Operator Level and Keyboard Scaling Depth – the program displays a graphic bar above the parameter value box so that the user has immediate feedback on how much of the allowable range is currently being used.

However, if the parameter value represents a piece of non-numeric data, the program has to display some representation of that data *in addition to* the numeric value, otherwise the user will rapidly become lost in a sea of unintelligible numbers...

Examples of the display of non-numeric data range from the simple On or Off of the Portamento Glissando (from the Miscellaneous Page) through the slightly more complex KLS Break Point (from the Keyboard Scaling Page) to such climactic delights as the Keyboard Scaling 'keys and curves' diagram (also from the Keyboard Scaling Page) mentioned above and the six Envelope curves discussed last month.

Heart of the Matter

The heart of DX7ED is the main input routine ▷

KEYBOARD SCALING
Operator 6

+LIN
3
CURVE

-EXP
1
CURVE

60
DEPTH

99
DEPTH

C 2
39
BREAK POINT

5
RATE

Figure 3.

COPY

SWAP | OPERATOR | 2 | WITH | 4
1 | 0 | 2 | | 4

6

5

2 | 4

1 | 3

Algorithm 1

Figure 4.

shown in Listing 1 comprising lines 1000 to 1560. This routine organises the display of the parameter value in a box, with a relevant label and graphic bar if required, at a particular place on the screen. In fact, this stage of the process is common to all parameters (except Voice Name, which has its own input routine) and is accomplished by a procedure known as PROCdisplay. I won't go into this procedure any further since much of it comprises a lot of BBC graphic commands that don't, I'm afraid, make very interesting reading. If the user selects a parameter for editing, PROCinput has to change the parameter value and cause both the parameter box value and any associated diagram to be updated.

The procedure definition of input is: DEF PROCinput(title$, par-num%, max%, x%, y%, bar%, do-input%). Looks great, doesn't it? Don't worry – all will be explained. The parameter label is passed into PROCinput via title$; par-num% is the index into a byte array (called par) which holds the parameter values read in from – or to

be transmitted to – the DX7 over MIDI; max% is the maximum value the parameter can be and is used by PROCdisplay to work out how much bar to display on the bar graph; x%, y% hold the position on the screen for the parameter box to be drawn at; bar% is a Boolean value w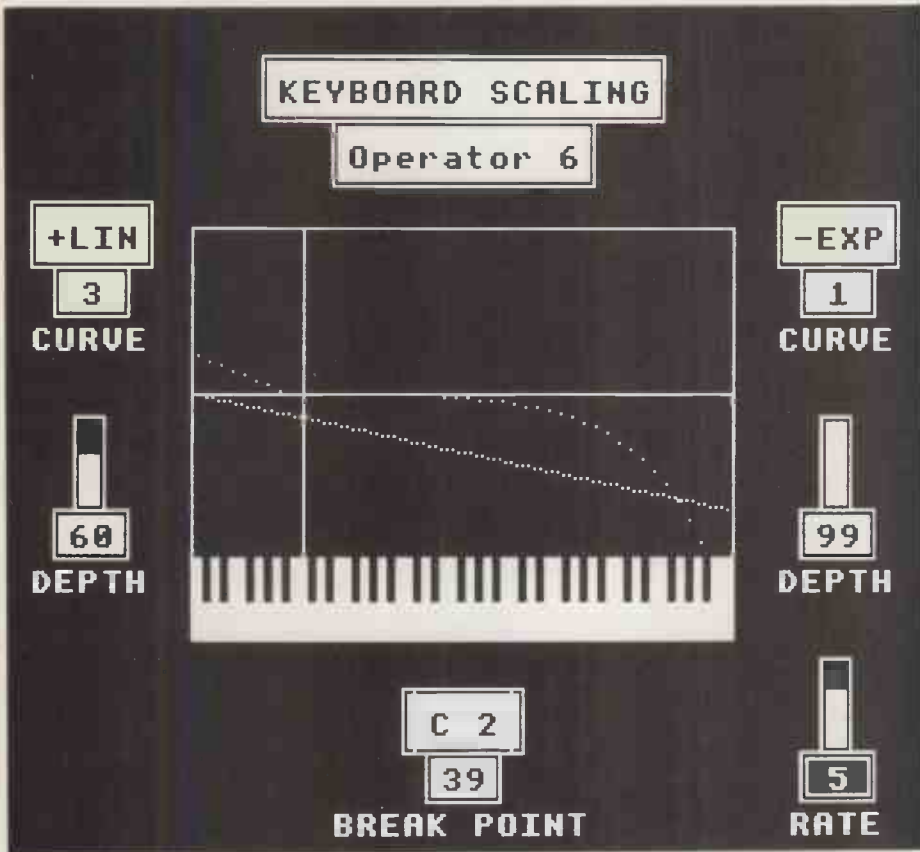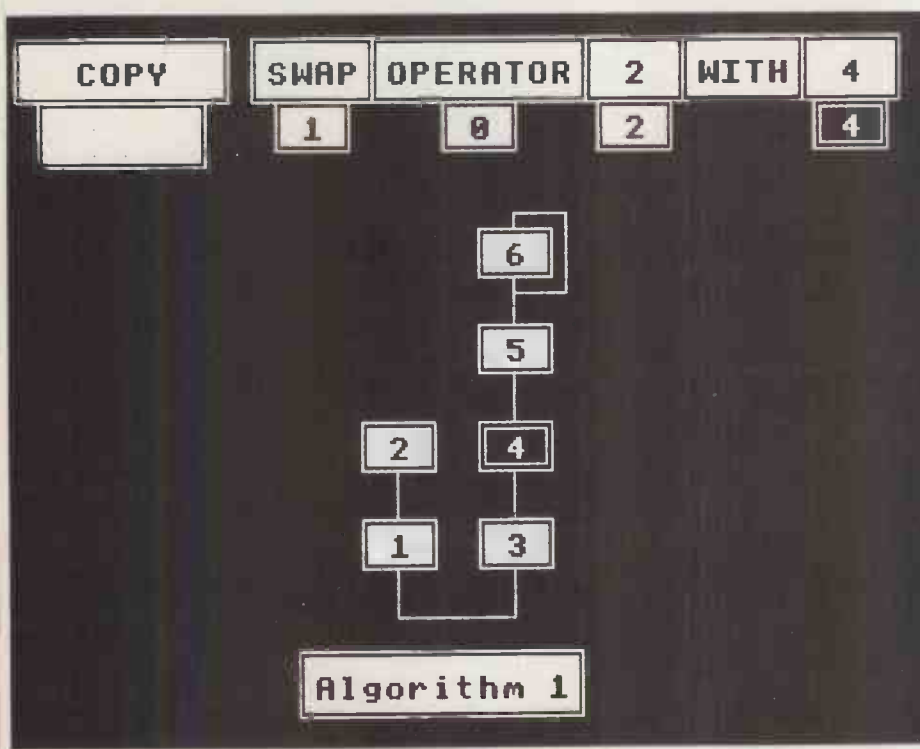hich says whether a bar graph should be drawn (this is usually set TRUE for numeric data and FALSE for parameters that need fancy diagrams); and lastly, do-input% is another Boolean value that specifies whether or not data is being input on this call of PROCinput. When the user selects a Page, all the parameter boxes and their diagrams are drawn onto the screen for his information, and only then is he asked for input affecting the first parameter – and since PROCinput has to 'know' how to draw everything, we let it do the work in both cases.

When PROCinput is called, the first thing it does is pick up a copy of the parameter's current value (line 1020). The procedure then goes into a REPEAT loop (lines 1040 to 1520) which only exits (line 1540)

when the user inputs one of the following characters: 'm', '/', 'M', or '?'. I'll describe the meaning of these characters later on.

PROCdisplay is called (line 1060) to put the current state of the parameter onto the screen at the position given in x% and y%. Note that bar% is passed straight through to PROCdisplay – the TRUE just before bar% in the PROCdisplay tells said procedure to draw the parameter box as selected, since we are inputting data.

Moving on, lines 1070 to 1370 are responsible for organising the drawing of non-numeric data representations. Take lines 1140 and 1150 as examples. The IF statement in both lines tests first to see if data is being input for a parameter whose data needs to be represented by 'off' or 'on'. The three parameter types tested for are Oscillator Sync, LFO Sync and Portamento Glissando. If the parameter type proves to be one of the three tested for and the parameter value% is currently zero, the AND condition of line 1140 is satisfied and 'off' is drawn above the parameter box on the screen. In much the same way, line 1150 draws an 'on' if the value is a 1.

Once all that drawing has been completed, the procedure enters the inner REPEAT loop (lines 1390 to 1410), which gets an input character from the user. In fact, line 1400 cheats (!) if no data has not yet been input by simulating the user inputting a '/' character, which means 'move on to the next parameter'. If data is being input, then one of the abovementioned characters is input. We can now see exactly what these input commands do by looking at lines 1430 to 1480.

The characters are used as follows. The comma and full stop (along with their respective Shift functions) are all connected with changing parameter values. Thus, a comma means subtract 1, a stop means add 1, a '<' means subtract 10 and a '>' means add 10. All pretty logical, really. Change of parameter is governed by 'm', which means move backward, and '/', which means move forward. Finally, screen-to-screen movement comes under the domain of 'M', which means move to previous screen, and '?', which means exit present screen or sub-screen.
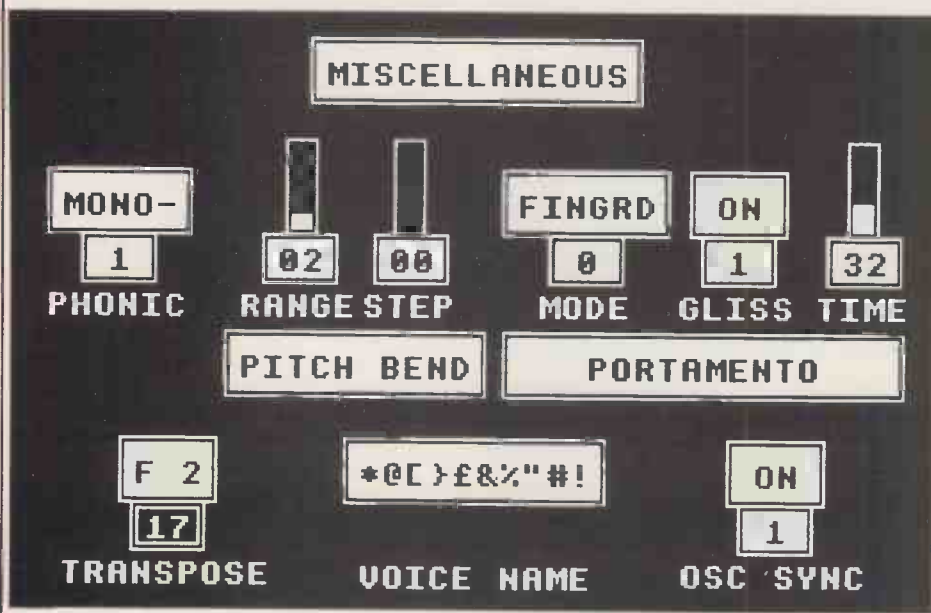

Figure 5.

```
1000 DEF PROCinput(title$,par_num%,max%,x%,y%,bar%,do_input%)
1010
1020 LOCAL value%:value%=par?par_num%
1030
1040 REPEAT
1050   IF ADVAL(-1)>0 THEN 1390
1060   PROCdisplay(title$,value%,max%,x%,y%,TRUE,bar%)
1070   IF par_num%=menu% THEN PROCdisplay_menu(value%)
1080   IF par_num%=wave% THEN PROCwave(value%)
1090   IF par_num%=copy% AND value%=0 THEN PROCstring("OPERATOR")
1100   IF par_num%=copy% AND value%=1 THEN PROCstring("ENVELOPE")
1110   IF par_num%=swap% AND value%=0 THEN PROCstring("FROM"):PLOT 0,
588,-84:PROCstring(" TO ")
1120   IF par_num%=swap% AND value%=1 THEN PROCstring("SWAP"):PLOT 0,
588,-84:PROCstring("WITH")
1130   IF par_num%=copy_from% OR par_num%=copy_to% THEN PROCstring(ST
R$(value%+1))
1140   IF (par_num%=osc_sync% OR par_num%=sync% OR par_num%=port_glis
s%) AND value%=0 THEN PROCstring("OFF")
1150   IF (par_num%=osc_sync% OR par_num%=sync% OR par_num%=port_glis
s%) AND value%=1 THEN PROCstring("ON")
1160   IF par_num%=poly_mono% AND value%=0 THEN PROCstring("POLY-")
1170   IF par_num%=poly_mono% AND value%=1 THEN PROCstring("MONO-")
1180   IF (par_num%=port_mode% OR par_num%=poly_mono%) AND par?poly_m
ono%=0 AND par?port_mode%=0 THEN MOVE 740,600:PROCstring("RETAIN")
1190   IF (par_num%=port_mode% OR par_num%=poly_mono%) AND par?poly_m
ono%=0 AND par?port_mode%=1 THEN MOVE 740,600:PROCstring("FOLLOW")
1200   IF (par_num%=port_mode% OR par_num%=poly_mono%) AND par?poly_m
ono%=1 AND par?port_mode%=0 THEN MOVE 740,600:PROCstring("FINGRD")
1210   IF (par_num%=port_mode% OR par_num%=poly_mono%) AND par?poly_m
ono%=1 AND par?port_mode%=1 THEN MOVE 740,600:PROCstring("F-TIME")
1220   IF par_num%<126 AND par_num% MOD 21=17 AND value%=0 THEN PROCs
tring("RATIO")
1230   IF par_num%<126 AND par_num% MOD 21>=11  AND par_num% MOD 21<=
12 THEN PROCstring(MID$(curves$,par?par_num%*4+1,4))
1240   IF par_num%<126 AND (do_input% AND par_num% MOD 21>=8  AND par
_num% MOD 21<= 13) AND (NOT do_input% OR value%=0 OR value%=max% OR ke
y$="," OR key$="." OR key$="<" OR key$=">") THEN PROCdisplay_scaling_g
raph
1250   IF par_num%<126 AND par_num% MOD 21=17 AND value%=1 THEN PROCs
tring("FIXED")
1260   IF par_num%=transpose% THEN PROCstring(MID$(notes$, (value%+27
) MOD 12 + 1, 1)+MID$(sharps$, (value%+27) MOD 12 + 1, 1)+STR$((value%
+12) DIV 12))
1270   IF par_num%<126 THEN IF (do_input% AND (par_num% MOD 21=17 OR
par_num% MOD 21=18)) OR par_num% MOD 21=19 THEN MOVE 600,520:PROCfrequ
ency( par_num% )
```

```
1280   IF NOT( par_num%=wheel%+1 OR par_num%=foot%+1 OR par_num%=brea
th%+1 OR par_num%=after%+1) THEN GOTO 1320
1290   IF value% AND 1 THEN PROCstring(" PITCH ") ELSE PROCstring("
")
1300   IF value% AND 2 THEN PROCstring(" AMPLI ") ELSE PROCstring("
")
1310   IF value% AND 4 THEN PROCstring("EG BIAS") ELSE PROCstring("
")
1320   IF par_num%=envelope% AND value%=0 THEN PLOT 0,0,80:PROCstring
("PITCH")
1330   IF par_num%=envelope% AND value%>0 THEN PLOT 0,-50,144:GCOL 0,
0:PLOT 1,200,0:PLOT 81,-200,80:PLOT 81,200,0:GCOL 0,1
1340   IF par_num%=algorithm% THEN PROCdraw_algorithm( value%+1, 0 )
1350   IF par_num%=copy_from% OR (do_input% AND (par_num%=operator% O
R par_num%=copy_to%)) THEN PROCdraw_algorithm( par?algorithm% + 1, val
ue%+1 )
1360   IF par_num%=envelope% THEN PROCdraw_axes(par?envelope%=0)
1370   IF par_num%=envelope% OR (par_num%<134 AND (par_num% MOD 21=4
OR (do_input% AND par_num% MOD 21<8))) THEN PROCplot_env(-(6-(par_num%
DIV 21))*(par_num%<>envelope%)-par?envelope%*(par_num%=envelope%))
1380
1390   REPEAT
1400     IF do_input% THEN key$=GET$ ELSE key$="/"
1410     UNTIL INSTR("mM,.<.>/?", key$)
1420
1430   IF key$="<" AND value%>9 THEN value%=value%-10 ELSE IF key$="<
" THEN key$=","
1440   IF key$=">" AND value%<=ABS(max%)-10 THEN value%=value%+10 ELS
E IF key$=">" THEN key$="."
1450   IF from%<>0 AND key$="," AND value%=0 THEN value%=max%:GOTO 15
00
1460   IF key$="," AND value%>0 THEN value%=value%-1 ELSE IF key$=","
THEN 1390
1470   IF from%<>0 AND key$="." AND value%=max% THEN value%=0:GOTO 15
00
1480   IF key$="." AND value%<ABS(max%) THEN value%=value%+1 ELSE IF
key$="." THEN 1390
1490
1500   par?par_num%=value%
1510
1520   UNTIL key$="m" OR key$="M" OR key$="/" OR key$="?"
1530
1540 PROCdisplay(title$,value%-from%,max%,x%,y%,FALSE,bar%)
1550
1560 ENDPROC
```

Listing 1.

```
2000 DEF PROCinput_LFO
2010 CLS:MOVE 590,800:PROCstring("LFO - Low Frequency Oscillator")
2020 do_input%=FALSE:GOTO 2040
2030 do_input%=TRUE
2040 PROCinput("WAVE",wave%,5,300,500,FALSE,do_input%):IF key$="m" TH
EN 2100 ELSE IF key$="M" OR key$="?" ENDPROC
2050 PROCinput("SPEED",speed%,99,600,500,TRUE,do_input%):IF key$="m"
THEN 2040 ELSE IF key$="M" OR key$="?" ENDPROC
2060 PROCinput("DELAY",delay%,99,900,500,TRUE,do_input%):IF key$="m"
THEN 2050 ELSE IF key$="M" OR key$="?" ENDPROC
```
```
2070 PROCinput("PMD",pmd%,7,150,200,TRUE,do_input%):IF key$="m" THEN
2060 ELSE IF key$="M" OR key$="?" ENDPROC
2080 PROCinput("PMS",pms%,7,450,200,TRUE,do_input%):IF key$="m" THEN
2070 ELSE IF key$="M" OR key$="?" ENDPROC
2090 PROCinput("AMD",amd%,7,750,200,TRUE,do_input%):IF key$="m" THEN
2080 ELSE IF key$="M" OR key$="?" ENDPROC
2100 PROCinput("SYNC",sync%,1,1050,200,FALSE,do_input%):IF key$="m" T
HEN 2090 ELSE IF key$="M" OR key$="?" ENDPROC
2110 GOTO 2030
```
Listing 2.

```
3000 DEF PROCdraw_algorithm( number%, select% )
3010
3020 LOCAL x%,y%,start%,dx%,dy%,byte_no%,byte%,drawn%:drawn%=0
3030
3040 IF select%=0 THEN MOVE 200,300:GCOL 0,0:PLOT 1,900,0:PLOT 81,-90
0,500:PLOT 81,900,0:GCOL 0,1
3050 start%=(number%-1)*12:x%=665-160*(alg?start% DIV 8 MOD 8)/2:y%=5
40-45*((alg?start% DIV 64 MOD 4)+1):dx%=0:dy%=0
3060
3070 byte_no%=start%+1:byte%=alg?byte_no%
3080
3090 REPEAT
3100    IF byte%=255 THEN 3200
3110    IF byte%>0 THEN PROCdisplay("",byte% MOD 8, 6, x%+dx%, y%+dy%
, byte% MOD 8 = select%, FALSE)
3120    IF byte% AND 128 THEN MOVE x%+dx%+50,y%+dy%:PLOT 1,0,-60
3130    IF byte% AND 64 THEN MOVE x%+dx%,y%+dy%:PLOT 1,-60,-60
3140    IF byte% AND 32 THEN MOVE x%+dx%+100,y%+dy%:PLOT 1,60,-60
3150    IF dy%=0 AND drawn%>0 THEN MOVE x%+dx%+50,y%-60:PLOT 1,-160,0
3160    IF byte% AND 8 THEN PROCdraw_feedback(x%+dx%,y%+dy%,alg?start%
MOD 8)
3170    IF byte%>0 THEN drawn%=drawn%+1
3180    dx%=dx%+160:IF byte% AND 16 THEN dx%=0:dy%=dy%+120
3190    byte_no%=byte_no%+1:byte%=alg?byte_no%
3200    UNTIL byte_no%-start%>11 OR byte%=255
3210
3220 ENDPROC
3230
3240 REM Coded Algorithms:
3250 DATA 209,129,147,130,148,  0,149,  0,142,255,255,255
3260 DATA 208,129,147,130,148,  0,149,  0,134,255,255,255
3270 DATA 145,129,148,130,149,131,158,255,255,255,255,255
3280 DATA 147,129,148,130,149,131,158,255,255,255,255,255
3290 DATA  89,129,131,149,130,132,142,255,255,255,255,255
3300 DATA  90,129,131,149,130,132,142,255,255,255,255,255
3310 DATA 153,129,147,130,132, 85,  0,  0,142,255,255,255
3320 DATA 152,129,147,130,140, 85,  0,  0,134,255,255,255
3330 DATA 152,129,147,130,138,132, 85,  0,  0,134,255,255
3340 DATA 153,  0,132,145, 37,134,146,  0,  0,139,255,255
```
Listing 3.

If the parameter value is to be changed, lines 1430 to 1480 check that the change is legal and then make the change if it is. The parameter is actually updated at line 1500. Naturally, if the requested change is not legal, the command is ignored by the program.

Conversely, if the input is a 'move to' command as opposed to a parameter change, PROCinput exits after redrawing the current state of the parameter in line 1540, with FALSE set for the 'select' in the PROCdisplay call. The routine that called PROCinput in the first place must then look at the character in the key$ to decide which parameter or page to route to next.

Incidentally, the ADVAL(-1) in line 1050 that I managed to ignore in passing checks to see whether there's a typed-ahead character in the micro's keyboard buffer. If there is, the special drawing routines are skipped, since (a) the new input probably makes what would have been drawn out of date, and (b) some of the routines take around a second to redraw and would therefore slow the editing process a little.

LFO Procedure

PROCinput-LFO is the routine (Listing 2, comprising lines 2000 to 2110) that displays the LFO page that appeared in last month's BeeBMIDI instalment. The general method used applies to all of the pages, however: only the details such as parameter label, number and on-screen position change.

When the routine is entered, the screen is cleared and the page heading is printed (line 2010).

In line 2020, do-input% is set FALSE so that the parameter boxes and display graphics are shown on the screen by the first pass through lines 2040 to 2110 but no data is input. If this weren't the case, no parameter would appear on-screen until some appropriate data was input. And then the program wouldn't be of much use to anyone, would it?

After the first pass, the routine is sent to line 2030, where do-input is set to TRUE. At all subsequent passes through lines 2040 to 2100, the user can alter each parameter to his heart's content and move onto the next when he's finished.

In the case of PROCinput-LFO, as the user exits from a PROCinput call with either 'M' or '?', he also exits from the LFO page and ends up at the Main Menu. This isn't always the case with some of the other routines, however. For instance, if the user exits forward from the intermediate Operator Select page mentioned earlier, he goes on to edit the Operator parameters, whereas exiting backwards returns him to the Main Menu.

Drawing Routines

The PROCdraw-algorithm routine (Listing 3, comprising lines 3000 to 3220) is included as an example of one of the more complex special drawing routines. DX7ED would be somewhat lacking if the synth's algorithms weren't displayed in full on the screen, since an awful lot of DX editing activities depend on the user selecting an Operator or envelope on the basis of its 'connections' within the current algorithm.

Frankly, I came up against something of a problem here in that I didn't particularly want to write 32 different routines – one for each algorithm. It would have been time-consuming, extremely tedious and, in all likelihood counter-productive: it would have eaten up vast quantities of RAM. What I've done instead is code the algorithms up into a maximum of 12 bytes each, leaving PROCdraw-algorithm to unravel the code and follow any instructions it might find. Lines 3240 to 3340 contain the codes for the first 10 algorithms.

Luckily, seeing as I'm in a particularly generous mood, I'm going to 'decode' the coding system used in DX7ED, so here goes.

The eight bits of the code's first byte are split into three fields as follows: hhwwwnnn. The hh (values 0 to 3) represents the height in Operators minus one (to give values 1 to 4) of the algorithm, while the www represents the width (values 1 to 6). These two values are used in line 3050 to work out where the first Operator should be drawn so that the whole algorithm is centred neatly on the page. The last three bits, nnn (values 0 to 6), represent which of the seven possible shapes of feedback loop should be drawn: this is passed onto PROCdraw-feedback in line 3160, just after the relevant Operator has been drawn.

The rest of the bytes in the code tell PROCdraw-algorithm what to do. At the end of dealing with each byte, the drawing position moves by one Operator to the right (start of line 3180) unless otherwise instructed. Thus a zero byte says simply 'don't draw anything, just move to the right'. When the horizontal row of Operators is reached, the drawing position is moved up a row and hard left (rest of line 3180). It's a bit like the carriage return on a typewriter, except that we move up on the screen rather than down it. As each Operator is drawn, it's connected to those in the row below it, as specified in the code.

As for the code in detail, this is pretty much how it reads. Bit 7 (value 128) means draw a connection to the Operator vertically below; bit 6 (value 64) means draw a connection below left; bit 5 (value 32) means draw a connection below right; bit 4 (value 16) means move up a row and hard over to the left; and bit 3 (value 8) indicates that the Operator currently selected is the feedback Operator. Finally, bits 2, 1 and 0 form the Operator number between 1 and 6.

It's left to lines 3110 to 3160 to check which bits of the code are on and perform the associated drawing operations accordingly.

Conclusions

It's my humble opinion that the DX7ED program provides an extremely user-friendly method of programming what's already become a notoriously unfriendly synthesiser. The idea behind this feature – and the one that came before it – was to describe the package's facilities in some detail and, perhaps more importantly in the long run, give some insight into how the program goes about doing its job.

In fact, the routines described above are coded in a different manner to those in the production version of the program, simply because it's easier to illustrate things this way.

Budding programmers interested in writing their own version of DX7ED (and why not? – only took me the best part of six months, after all) might well be better off making the program entirely table-driven, in the interests of saving space and simplifying the routing control in each of the page modules. The extensive use I've made of named constants (eg. menu%, wave%) in the BBC BASIC program from which the example procedures were taken would be better suited to a compiled language: the enumerated type available in Pascal would be almost ideal.

Technical trivia aside, a production version of DX7ED is now available on cassette (and will be transferable to disk) from EmmSoft at the ludicrously low price of £24.95, inclusive of postage, VAT and all that nonsense: the only proviso is that you be prepared to hang on a maximum of 28 days before delivery. As an added bonus, DX Owners' Club members will be able to claim a discount, full details in a forthcoming newsletter.

And if the demand is there, we'll consider making the program available for other micros, too, so if you've got a specific requirement that's not currently being met, drop me a line care of E&MM and I'll see what I can do. ■

EMMSOFT

The latest from EmmSoft, the hardware/software marketing division of Electronics & Music Maker.

EmmSoft has taken under its wing both past micro-based projects and will include future E&MM software developments. The following guide summarises the EmmSoft projects for which printed circuit boards and software packages are available, and this will be updated every other month. All prices quoted are inclusive of VAT and postage and packing – please allow 28 days for delivery. Send your order, with payment in sterling cheque, postal order or bankers' draft payable to Music Maker Publications, to **EmmSoft, E&MM, Alexander House, 1 Milton Road, Cambridge, CB4 1UY.**

MicroMIDI   *May 83*

A single-board serial interface that will link any MIDI synth to the Sinclair Spectrum microcomputer. Features include three parallel I/O ports, crystal-controlled data transfer, and opto-coupled output.

The PCB is available from EmmSoft at £4.25.

MicroMIDI II   *July 84*

A revised, simpler version of MicroMIDI was published subsequently which incorporated the same facilities with the exception of the three parallel I/O ports.

The PCB is available from EmmSoft at £4.25.

As part of the 'Spectrum MIDI' article in E&MM July 84, two software programs were published – a SixTrak Patch Dump and DX7 MIDI Dump – both of which will run on either version of MicroMIDI.

A cassette containing an expanded version of Steve Parr's DX7 MIDI Dump program (including a short sequencing routine) can be obtained from SDS, 18 Cambalt Road, London, SW15 6EW, for £5.95 including postage and VAT.

OMDAC   *June 83*

The OMDAC, when used in conjunction with a Z80-based microprocessor, will provide eight sets of gate, trigger and control voltages compatible with most one-volt-per-octave synthesisers.

A 'Patch Change' program for the Spectrum was published in E&MM September 84, while the second OMDAC Update (E&MM October 84) enables the hardware to be modified to run on the BBC microcomputer. Further OMDAC software is in the pipeline (see also 'Drum Sequencer', E&MM November 84).

BeeBMIDI   *June & July 84*

A MIDI interface for the BBC Model B microcomputer, Part 1 of BeeBMIDI contained the technological and constructional details, while Part 2 continued with a full parts list and some MIDI software routines. The PCB is available from EmmSoft at £4.95.

BeeBMIDI Software   *August 84*

A full listing of a comprehensive dump program written in BBC BASIC and 6502 Assembler for the Yamaha DX7, with the software also available on cassette (for the sore-fingered) from EmmSoft, price £3.95 (reduced from £7.95).

January 85

BeeBMIDI Part 6 features a program that allows you to dump Roland Juno 106 voices onto disk or tape, and which can be adapted to suit other micros and MIDI synths. The program is available on cassette from EmmSoft, price £3.95.

Further software for BeeBMIDI is

currently under development.

Please note that the opto-isolator with the right specification is available from EmmSoft at £2.75. The Maplin component was quoted in the BeeBMIDI parts list in order to ease the problem of purchasing the correct opto-isolator, but this subsequently proved to be unsuitable.

February & March 85

BeeBMIDI 7 & 8 feature the DX7ED software package, a comprehensive DX7 editing program which receives voice parameter information from the DX7 and displays it clearly in useful related sets on screen, allowing the amplitude/time graphs of all six Operators' envelope generators to be viewed at the same time, and complete sets of envelope generator and operator parameters to be altered and/or swapped about at will. DX7ED is available from EmmSoft on cassette, price £24.95, but members of the DX7 Owners' Club can take advantage of a special offer – further details from the Club.

Drum Sequencer   *November 84*

Some new software for the BBC Micro that allows E&MM's electronic percussion modules to be sequenced using either the OMDAC or the user port of the BBC B.

Available on 40- or 80-track disk (please state preference) from EmmSoft at £8.95.

THE FAIRLIGHT
EXPLAINED

When a CMI page is as helpful a source of music graphics as Page 6, you can bet your life it needs a lot of explaining – so this month's episode is dedicated entirely to it. *Jim Grant*

f you're trying to create a new sound on the Fairlight, whether it's from existing sampled data using Page 8 or harmonic profiles generated on Pages 4 and 5, you can be sure that Page 6 will be referred to over and over again.

At its simplest level, Page 6 is a static 'oscilloscope' containing one segment of the waveform RAM. Like Page 4, the large black area is a reverse video image and represents the region that can be hit by the now-infamous lightpen. But unlike an oscilloscope, which acts only like a window on the information, Page 6, in conjunction with its commands and lightpen, behaves more like a door through which we can directly access the waveform data.

The Manual Approach

Regular readers of this series will probably recall that each segment consists of 128 bytes, and that each byte can hold a range of values from -128 to 127, giving a grand total of 256 possible values. Now, we can highlight any particular byte by positioning a narrow vertical window over the display area. The byte in question, shown in Figure 1, has its position indicated by P O I N T and its value by L E V E L: these can be changed by typing in different values from the alphanumeric keyboard. So, theoretically at least, you could type in 128 levels for each byte shown in Page 6, and repeat the process for all 128 Page 6 displays to cover the entire sound, giving 16,384 levels in all. In practice though, such a feat would take an unbelievable amount of manual labour (exactly what much of the Fairlight's software was designed to eliminate) to achieve, and this has led me to wonder whether this feature will be dropped when the Series III Fairlight, complete with waveform RAM of megabyte dimensions, makes its appearance later this year.

A more practical approach is, of course, to make use of the lightpen. This is as simple as drawing on the back of a bus ticket, but just in case even that is beyond the user's artistic capabilities, the J O I N and P L O T functions also present on Page 4 are available here, too, to help out with the drawing of geometric shapes.

In fact, trying out a few sample sketches soon reveals that waveforms which look drastically different may *sound* surprisingly similar. Scientifically speaking, this is probably due to the fact that as a
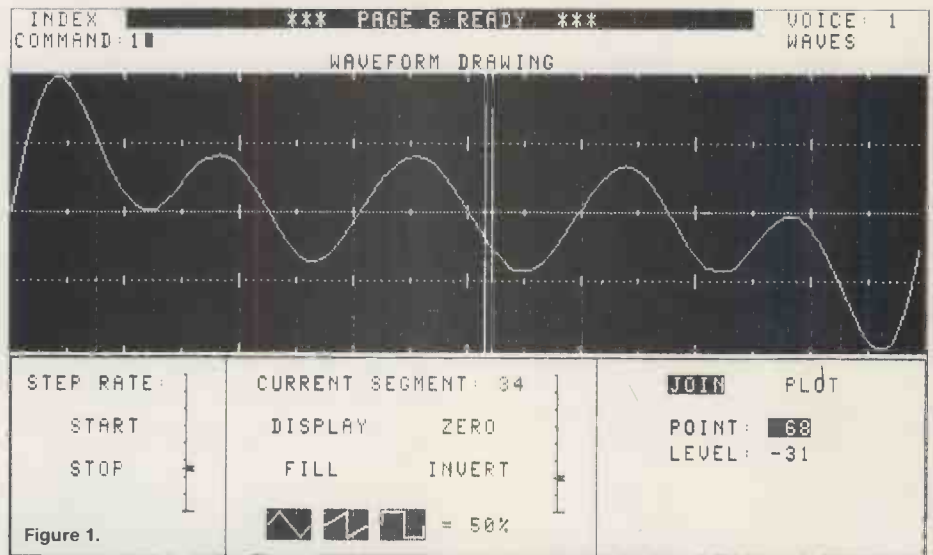
Figure 1.
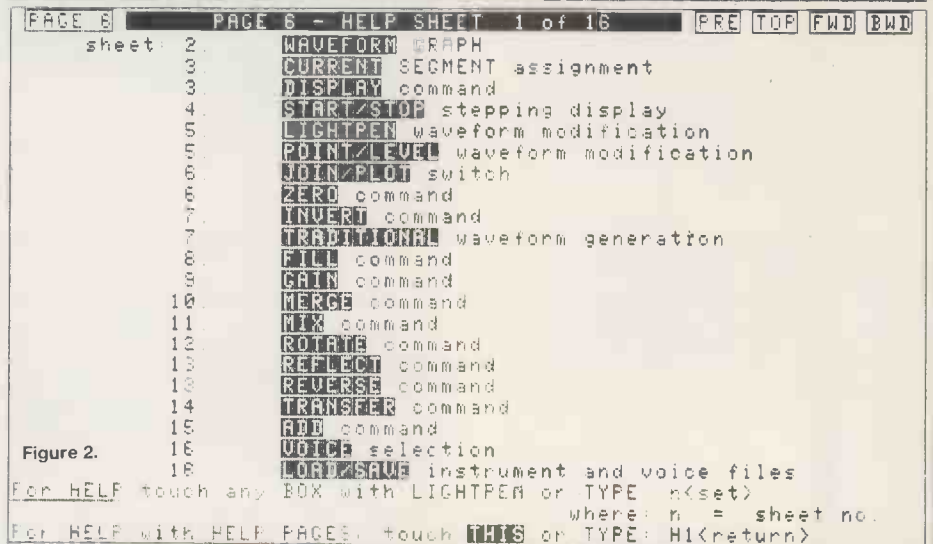
Figure 2.
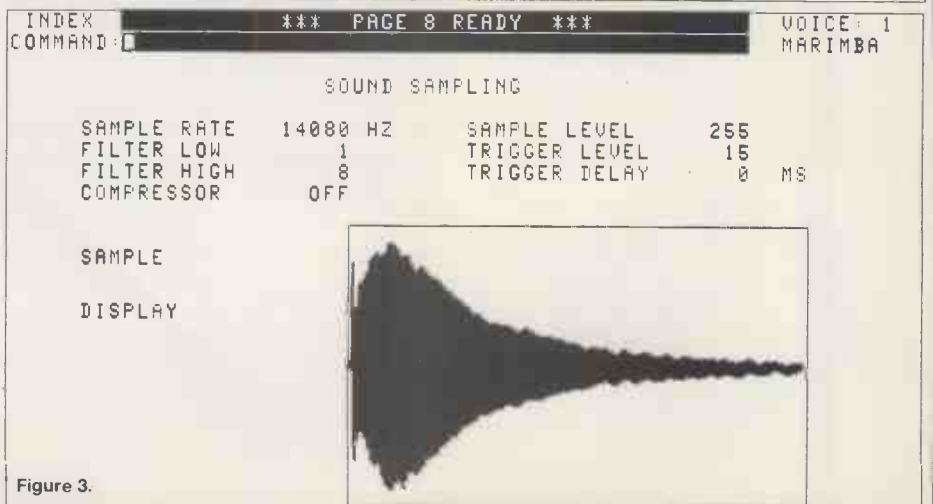
Figure 3.

species, we have little prior experience gained from our senses that relates visual time domain waveforms to what we perceive in terms of harmonics or partials.

But Page 6 waveform-drawing is of enormous help, first because it gives the user extensive and direct control of waveform RAM with one sweep of the hand, and second because the ability to bring about such drastic sonic change so simply and visually (almost artistically, some might say) is a practicality light years away from the sophistication and complexity of the underlying machine and its software. Or to put it another way, it makes the CMI seem less of a beast and more of a pet.
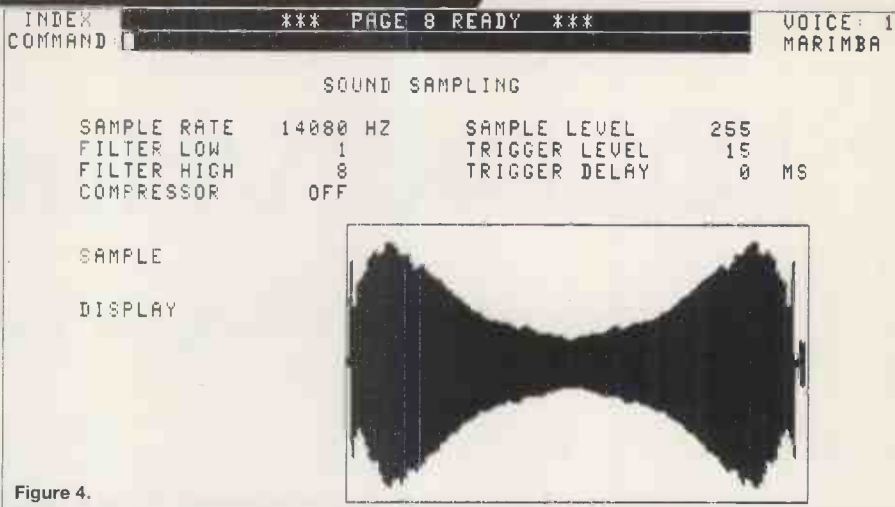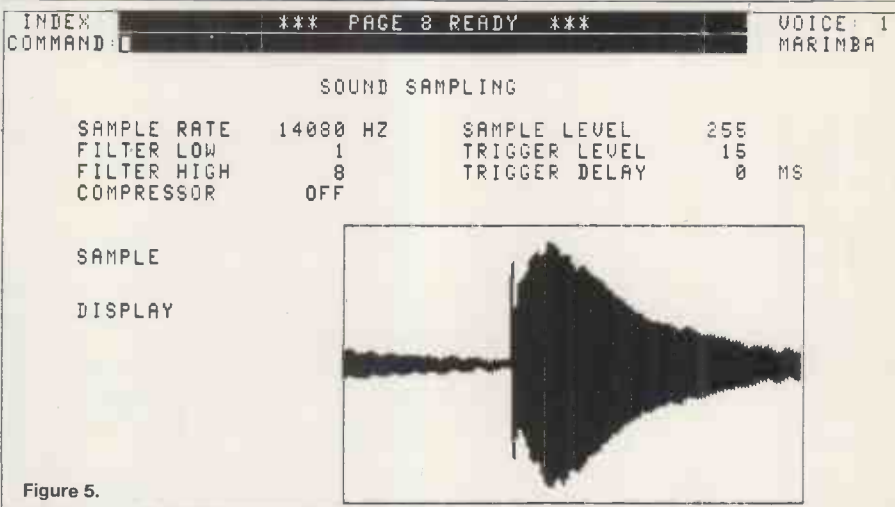
The Practical Approach

Alteration of the display image is not reflected in the waveform RAM until the FILL command is used. This allows the reverse video field to be used as a scratchpad area in which waveshapes can be developed before they are finally committed to RAM.

Looking again at Figure 1, notice there are two 'slider' controls similar to those on Page 5. These control the segment number to be displayed and the stepping rate through the entire memory instigated when the command Step is issued. Also shown are three classic waveforms – triangle, sawtooth, and square – which deposit a perfectly fitting single cycle wave into the segment. The pulse width of the square can be varied from 1% to 99% by changing the value held next to the square wave symbol.

Auxiliary Commands

In addition to the graphic capabilities of Page 6, there are a number of commands that can only be entered from the Fairlight's alphanumeric keyboard. Figure 2 shows the range of functions available, as presented by the HELP sheet menu.

First off are a few utility-type commands. GAIN scales the displayed data by a specified percentage: if the rescaled waveform is about to exceed the amplitude range of the system, the CMI will ask you whether or not you want to proceed and thereby induce clipping. If you reply in the affirmative, the command will be duly executed. Meanwhile, the INVERT command inverts the phase of the waveform: this is useful as a prelude to some of the other functions such as MIX, MERGE and ADD.

A particularly neat little command is ZERO, which allows us to create a null voice in preparation for the ADD command. The entire waveform RAM can be

Figure 4.

Figure 5.

turned end on end by using REVERSE, and this results in the backwards sounds that have become familiar to pop music followers the world over.

REFLECT is a less commonly-used but if anything more interesting variation on this theme. It allows us to place an imaginary mirror in the sound and reflect every part of the waveform in front of this mirror to waveform RAM behind it. Take a look at Figures 3 and 4 for before and after views: the 'mirror' is at segment 64.

Now on to another of the seemingly insurmountable problems brought on by the onset of new musical technology. Very often, when a sound has been sampled, the beginning of the captured data does not occur at exactly the start of the RAM, perhaps due to some extraneous noise pretriggering the ADC. However, if the effect isn't too severe, a convenient method of correction is to ROTATE the sound within the waveform RAM to bring the start in line with byte 1. This has the (often desirable) side-effect of shifting the first part of RAM to the end. See Figures 5 and 3 for another revealing before and after picture.

I imagine NOISE will be fairly self-explanatory to even the least clued-up of this column's readers: it fills sections of the RAM with the output of a random number generator hidden inside the software. Funnily enough, the GAIN command is also used to tailor the amplitude of the noise to that of the rest of the sound...

Now, BLEND is a strangely out of place command. Personally, I think it should be on Page 4, since its role is to help smooth out glitches caused by imperfect looping points. It may be that to find a good loop it makes extensive use of extrapolation techniques, and that seeing as this is a central feature of the MIX and MERGE commands, Fairlight decided to bung it on Page 6 alongside them.

However, rather than discuss the remaining Page 6 commands in any detail right now, I think it's probably best for all concerned if I demonstrate their power in the context of a typical edit session in which a sound is created from scratch, ie. without any sampled data. So, now you know what I'll be talking about next month.
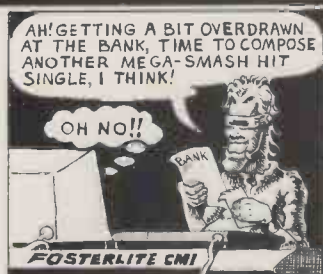
SDS DX7 Voice Editor

Software for Sinclair Spectrum

E&MM's own DX7ED might be the ideal DX editing program if you've got a BBC Micro, but what about Spectrum owners who want the same sort of facilities? This SDS package might be the answer. *Gerry Queen*

I know I'm going over well-trodden ground, but when Yamaha introduced the DX7 in 1983, they produced a world-beater that had incredible voicing capabilities but a tiny liquid crystal display on which owners had to attempt to see all its various parameters. Not to put too fine a point on it, trying to look at the parameter settings for any voice, let alone actually altering any of them, involves more tiresome button-pushing on the DX7 than just about any other synth you care to name.

Luckily, the DX's designers also saw fit to include a decent implementation of MIDI, which allowed all of the parameters to be displayed and edited using a computer with suitable hardware/software. For many, the ultimate DX-oriented computer set-up has to be Yamaha's own CX5M computer and appropriate editing software, but seeing as an awful lot of musicians took out second mortgages to buy their DX7s in the first place, it's by no means a universal cure to the DX's accessibility ills. Now a more cost-effective alternative is available in the form of the Sinclair Spectrum, the suit-able MIDI interface, and Sound Design Studio's DX7 Editor software.

Using the Program

All you need to get going is a DX7, a 48K Sinclair Spectrum (now *very* cheap secondhand, incidentally), a suitable MIDI interface, a cassette recorder, and a couple of MIDI cables. The MIDI interface can be any of the following: E&MM, EMR, JMS, XRI Micon, or Siel. I tested the program with E&MM's own MicroMIDI board and had no problems at all.

Once you've connected everything together (make sure you get the MIDI Ins and Outs the right way round) and switched on, you simply type in the usual LOAD" ", start the tape recorder, check that you've got the title, and then go away and make a cup of coffee (the program takes a while to load).

The first option you're given is to designate which MIDI interface you're using. Simply press the appropriate key and the program will run, giving a Menu of options that includes invoking of the Editor, saving or loading a single voice (or bank of 32 voices) to or from the DX7/tape. Before progressing any further, you must first set up the DX7 by selecting the Internal Memory Protect off and setting 'Systems Info Available' using Button 8 on the DX.

Dump Program

You can now use the Dump side of the program to save those precious voices you have in the synth's internal memory. This is a simple operation that involves setting the computer to receive mode and then transmitting the voices from the DX7 using 'MIDI Transmit'. The actual transfer takes about a second, and the computer tells you when 32 voices have been received. In fact, said voices are placed in a Voice Buffer from which they can be saved to tape using another function of the program. This Voice Buffer can contain either a single voice *or* a bank of 32, so beware – you'll erase an entire bank if you subsequently transfer a single
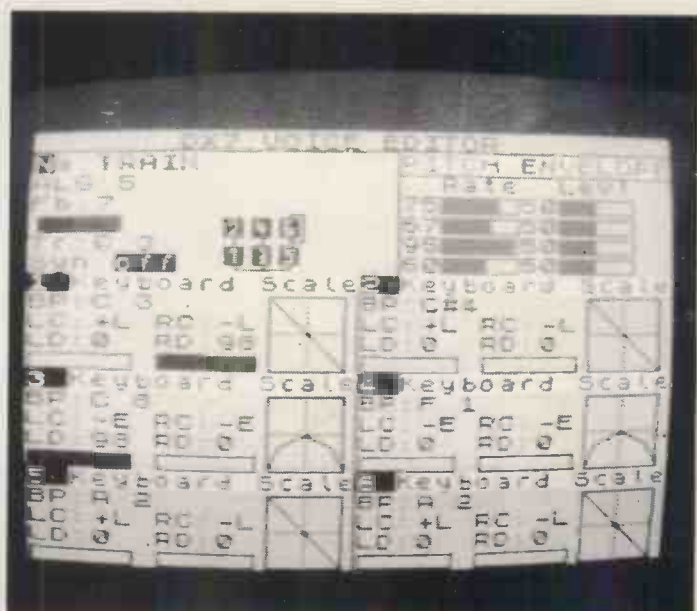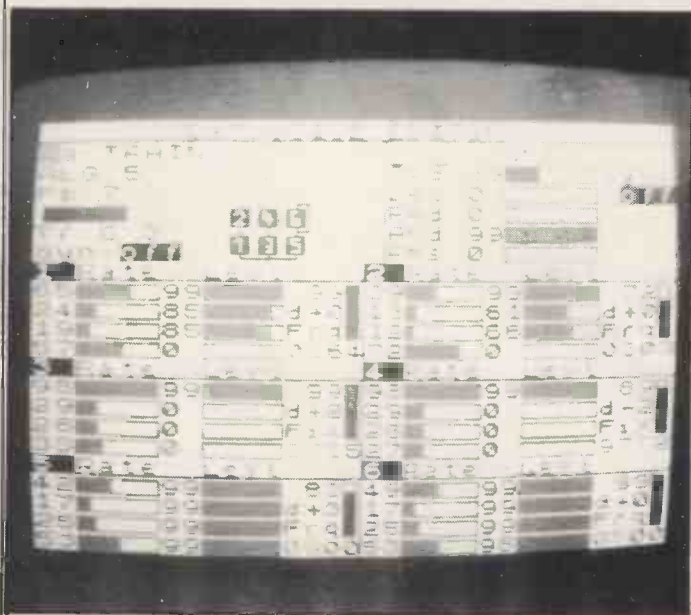
voice from the DX. The Tape Dump works well, and building up a library on a single cassette couldn't really be more straightforward.

The Editor

To invoke the Editor, you've got to load the Voice Buffer with a single sound using the Single Voice Transfer function.

Your television screen then undergoes a dramatic metamorphosis, as a tastefully-coloured display of all main parameter values (in both numeric and bar graph form) comes suddenly into view. The display is remarkably close in design to that of the CX5M's Editor, including as it does a graphic representation of the algorithm in use and detailed information on just about everything else, as we'll see.

The display is split into eight sections. The first contains Voice Name, Algorithm, Feedback, Key Transpose, and Key Sync, the second contains the LFO details, while sections 3-8 contain individual operator details, EG Rates and Levels, Output Level, Detune, and Frequency data.

A flashing cursor is used to show which part of the screen is being edited at any one time, and this can be moved (using the cursor keys, naturally enough) to any of the eight sections, while the Enter key is used to move the cursor within a

section and update information. And once you've altered the value for one parameter, the cursor moves to the next

> 'Once you've altered the value for one parameter, the cursor moves on to the next one automatically, so key-pushing is reduced to a minimum.'

automatically, so QWERTY key-pushing is reduced to a minimum.

As the last value within a section is changed, the cursor moves back to the start of the section and your editing is transmitted to the DX7. This means you can now hear the results. Now, while this may not be quite as good as that on the CX5M, which transmits as each value is changed, the SDS system's way of working is by no means annoying.

And as a bonus, pressing the 'E' key after setting EG rates and levels for a

particular Operator gives you a graph of current values – useful if you want to know what each Operator is doing.

Finally, once you've finished the first screenful of parameters, you can call up two further screens to edit the remaining functions, viz Pitch EG, Level Scaling, Sensitivities, and so on.

Conclusions

It's clear to me that the amount of thought SDS have put into the development of this software – and particularly its wonderful, if plagiaristic, set of screen displays – has paid dividends.

For the reasons I mentioned at the start, DX7 owners have good reason to seek the assistance of computers when it comes to getting the best from their synth, and this package will certainly help them do that. A comprehensive set of instructions is included in what's already a very reasonable asking price: I certainly can't think of a competing package that's more affordable than a Spectrum, a MicroMIDI board and the SDS software.

For many, then, the key that opens the door to trouble-free DX programming. ∎

DX7 & DX9 Voice Creator/Editor programs are available at £24.99 each from Sound Design Studio, 88 Dean Street, London W1.