THE

MUSIC TECHNOLOGY

MAGAZINE

SEPTEMBER 1986 $2.95

GENESIS
The Tony Banks Interview

ENSONIQ ESQ1
The All-in-One MIDI Machine?

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Australia's Hottest Export

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- Recognizes MIDI Song Pointer.
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VOLUME 1, NUMBER 2
SEPTEMBER 1986

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THANKS!

IT'S NEARLY TWO MONTHS NOW since we launched MUSIC TECHNOLOGY with our inaugural issue. In that time, life for the staff in our Californian office has become somewhat hectic.

Why? Because subscription checks have been flooding in (many of them accompanied by flattering letters, some of which are reprinted on page 7) in greater quantities as each day has gone by. And because the phones keep ringing with dealers wanting to stock the magazine in their stores.

In the end, we just didn't have any choice: we *had* to take on more staff to cope with the workload...

And that's not all. As part of the launch schedule for MUSIC TECHNOLOGY, our people attended two major trade events – NAMM at Chicago and the San Francisco Music Fair. On both occasions, the Music Maker stand was inundated with people clamoring for more information. Our writers were besieged by enthusiastic would-be readers. And our business people were being congratulated everywhere they went, by industry figures thanking us for providing a much-needed breath of fresh air.

So why all the fuss?

Well, in addition to individual articles in that first issue, it seems it's the *concept* of the magazine that's been the most popular receiver of readers' compliments. Large numbers of people have commented that MUSIC TECHNOLOGY is already filling a big gap in the music magazine market, and our editorial policy of informative interviews, no-nonsense reviews and easy-to-follow practical features seems to be a hit with everyone.

Overall, we think you like us – and we're flattered.

But as we said in our Editorial on page 2 of that inaugural issue, your criticisms and worries are just as welcome as your congratulations. So far, there haven't been too many messages of complaint (actually, there haven't been any at all).

Yet some of our readers are definitely worried. The big question many have been asking is: can MUSIC TECHNOLOGY keep it up? After all, it's not unusual for a magazine's first issue to be a real stunner. What *is* rare is to find the magazine maintaining the high standards it had previously set.

Well, pardon us if we sound arrogant, but we don't think you'll be disappointed by MUSIC TECHNOLOGY as the months pass. If anything, we're aiming to make subsequent issues of the magazine even better, by listening to what our readers have to say, and implementing popular demands.

Why are we so confident? Well, the main reason is that although MUSIC TECHNOLOGY is a new name Stateside, the sister company of the corporation that owns it has been publishing music magazines for over five years.

When our staff writers put pen to paper in California, they know they have a wealth of experience to call on across the Atlantic. We're used to publishing magazines in a fast-moving, flavor-of-the-month arena like the musical instrument business. We're used to changing the make-up of magazines in the light of changes in the industry they're serving. And most of all, we're used to the idea that no matter what anybody else says, it's a magazine's readers who are the most important people. Always.

So, if you've already subscribed to MUSIC TECHNOLOGY – or are considering doing so – you can rest assured that this is one magazine that puts its readers *first*. And that's a promise. ■

Choosing the right sampling keyboard is an open and shut case.

There are lots of reasons why you see more Mirages on stage and in studios than any other sampling keyboard. Besides being affordable, the Mirage can perform musical tricks* other keyboards can only envy.

The Ensoniq Sound Library now contains over 300 distinct sounds—from grand piano and strings to synthesizer and screaming electric guitar—all on fast-loading standard 3.5″ diskettes. The Mirage can also sample any sound you can get a mike on, opening a whole new universe of sonic possibilities. And for serious samplers there are sound editing programs available for all major personal computers.

If your setup already includes a MIDI keyboard or controller, consider the Mirage Digital Multi-Sampler. It's designed specifically for MIDI performance and studio use and it's everything a Mirage is... without the keyboard.

Visit your authorized Ensoniq dealer today. See and hear why twice as many musicians have chosen the Mirage over all other multi-samplers combined.

**If sawing a waveform in half or pulling an entire string section out of a disk sounds like your kind of magic, a Mirage can make you a wizard. The Mirage Digital Multi-Sampler retails for $1395—the Mirage Digital Sampling Keyboard... $1695.*

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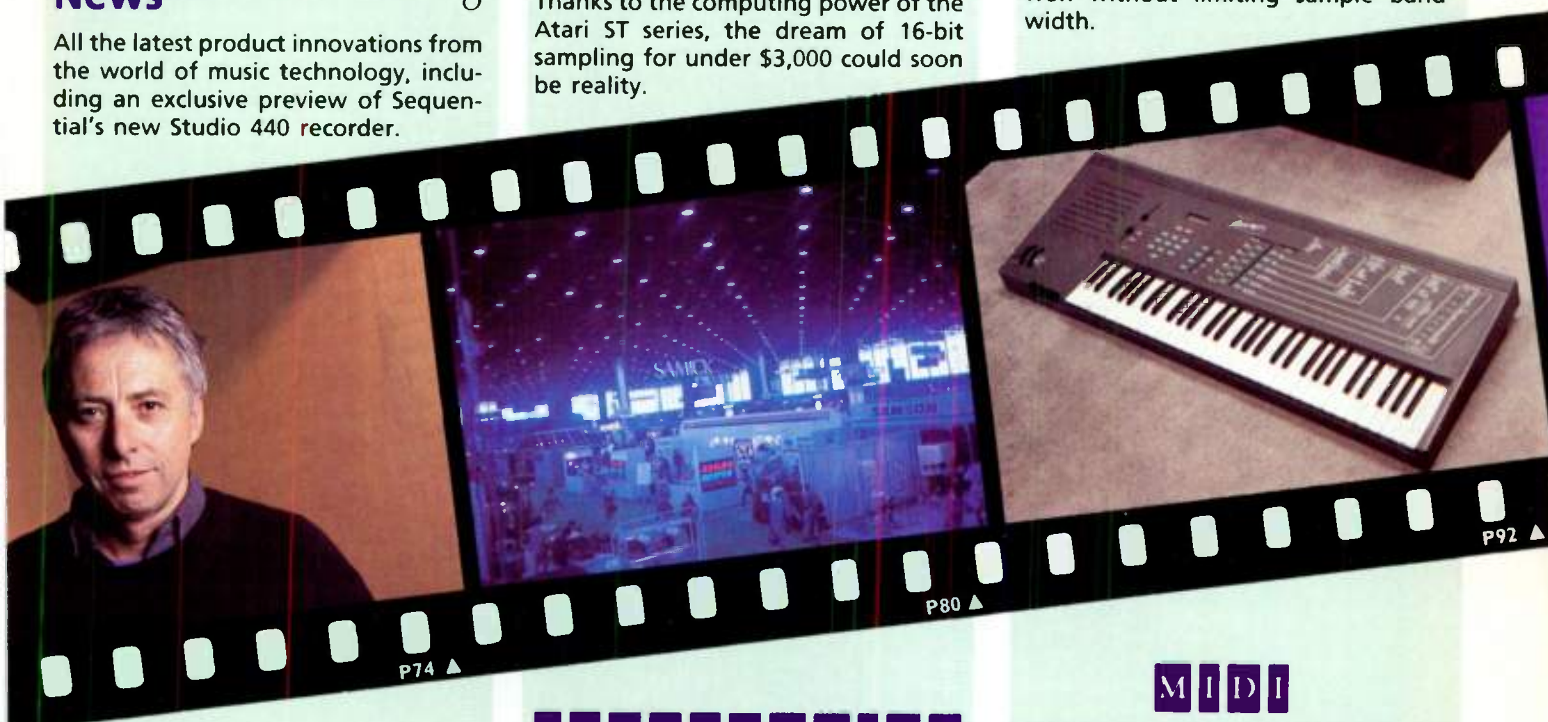


PPG

AMERICA, INC.

2210 Wilshire Blvd., Suite #700
Santa Monica, CA 90403 Tel. (213) 827-0952

Dear Reader,

I am pleased to inform you that PPG Palm Instruments Gmbh, the West Germany based manufacturers of digital synthesizers, have opened an American office to promote sales, improve service and provide training as to the best use of the current Wave System and lower prices! Also this year sees the launch of two new exciting products, The HDU and The Realizer, plus the release of revised software for the Wave 2.3, Wave 2.2, Waveterm and the HDU.

The HDU is a self contained sampler, multi-track recorder and real time audio processor. As a stand alone device the HDU can operate as a 10-track 16-bit recorder with 44.6kHz sampling rate and a total record time of 12 minutes using an 85-megabyte Winchester drive. For recording vocal or instrumental passages each track can be routed to the audio processor for one of the following effects; flanging, echo, phasing and harmonising. For best use of the HDU's memory space and facilities bouncing from track to track adding new effects with each pass is possible without any loss in quality as the signal remains in the 16-bit domain. With MIDI sync or an optional SMPTE interface the HDU is a definite 'must have' for recording studios, film and video production companies as well as MIDI one-man-bands.

The Realizer is a production center including a 16-voice digital synthesizer without oscilators and filters but instead uses digital signal processing and software to achieve Wavetable, FM and MiniMoog™ styles of synthesis which can be combined along with sampling. The Realizer also includes an expanded HDU with 24 minutes total record time. Due for release in Nov/Dec, the Realizer will bring an unparalleled flexibility to synthesists and composers alike.

You may not have even heard of PPG products before, but you can be sure of hearing PPG on the air. David Bowie, Steve Winwood and the Robert Palmer Band all feature PPG sounds on their current singles. If you require further information, please telephone or write to us.

Yours sincerely,

Andrew Thomas

Andrew Thomas - Vice President

WAVE 2.3 WAVETERM EVU PRK-FD HDU REALIZER

LETTERS

Dear Music Technology,

Congratulations on your new magazine, which I received last week and absorbed over the weekend. It's well formatted and well printed, too (I know because the ink didn't come off in my hands, like it does in Keyboard... no kidding!).

Want to know what really knocked my socks off? The illustration on Page 15 that showed the live setups of Tangerine Dream and Saga. I'd love to see similar artwork on Kraftwerk, Yes, Moody Blues, Howard Jones, Stevie Wonder, etc.

One last thing, how about an article on a real high-tech artist who's often abused in other magazines: Laurie Anderson. Nobody seems to like her, but I just saw her live show in Atlanta three weeks ago, and she was truly awesome and totally unique. Her drum suit will kill you...

Here's wishing you continued success...

Jim Eshleman
New Lake Studio
Macon, GA

Dear Music Technology,

I would like to congratulate you on an excellent inaugural issue. It is immediately clear to me that 'Music Technology' seems set upon bringing its readership the same clear and incisive reporting that has characterized 'Electronics & Music Maker' for so many years. You're off to a fine start – keep up the good work!

I would like to bring to your readership's attention the formation of an Oberheim Xpander Users' Group. This Users' Group will serve as a forum for the exchange of patches, tips, questions and answers, and will provide a semi-regular newsletter and contact list (tentatively entitled 'Xpansions') to all group members. This service is and always will be free of charge; the Users' Group requires no membership fees of any kind, and is in no way affiliated with the Oberheim division of ECC Development Corp.

A self-addressed, stamped envelope will speed replies, but is not necessary. Xpander owners who don't want to miss out on this opportunity should write me as soon as possible.

Michael Metlay
MysTech Productions
PO Box 81175
Pittsburgh, PA 15217-0675

Dear Music Technology,

First, thanks for your inaugural issue. It is really refreshing to see 'another' musicians' magazine. But in your case, it is not just another magazine – it's greatly needed, I think, in our incredibly fast-paced, obsolete-tomorrow world of music equipment and technology.

To keep ahead of everything you'd need a MIDI brain that'll correct to all the manufacturer's computers! But you seem to shed a light on an otherwise seemingly mindboggling array of companies dealing in the obviously very profitable world of electronic instruments.

As a drummer who has become subject to Simmons synthesis, I'd like to commend Nigel Lord on his (extremely well-written) review on SDS1000. His vivid description of Dave Simmons was marvelous! Being a pioneer in his field, Simmons opened the doors for countless copies, but like Nigel Lord stated: "Long Live the Simmons Sound". Nothing comes close to that unique quality.

Well, to close, I'd just like to wish you very well indeed, and want to thank you for opening my eyes in so many ways. People like me who blow all their money (and incidentally, hock their typewriter to boot) to get more equipment need concise, intelligent and thoughtful articles to help us get the most out of the greatest gift of all – our music.

Cheers and Best Wishes.

Terry A Miller
Santa Barbara, CA

Dear Music Technology,

I have read your first issue. I probably don't have to tell you it was successful, do I? Your confidence came through as well as your information, which was great.

I'm a songwriter, and when with a band I'm a lead vocalist also. I have an Oberheim Matrix 6 which is what I write on. I'm just like a lot of others in music who turned around one day and realized the necessity, joy, intrigue with MIDI and sampling and all the wonderful things becoming out there in the world. And I'm one of those who said, "Yeah well it's here, but I don't understand how and when it got here". Most of us have come into these innovations in the middle. No-one has previously bothered to go back and give the information on the beginning.

That is, until your first issue. Your DX Explained article was great. How about the same treatment for the Oberheim Matrix? Anyway, you have the right idea. A good approach. Your choice of artist and interview with Peter Gabriel (setting aside the fact that since 1970 I wanted to be Peter Gabriel when I grew up) was very good.

Just remember to keep a few pages around for anyone in the dark trying earnestly to understand the changes. It's simple when it's laid out and explained once in a while. Anyone who wants to know, deserves to know! Thanks for the real steps towards Music Technology progress.

Victoria
Mountain View, CA

Dear Music Technology,

Sometime in June '86 I received this magazine. I don't know how I got it, or where it came from. But I have never, in my 20 years of playing music, seen anything like this.

I'm an electronic percussionist, and constantly seek ways to improve my equipment and art.

This, truly, is one of the best, most informative magazines I've ever read. Keep up the good work, guys.

Raymond G Knights
St Paul, MN

Dear Music Technology,

You sent your mag to my brother, a strict acoustic drummer. I, "the young musician sitting at home with a Portastudio, surrounded by keyboards and drum machines", picked it up and read it cover to cover.

I like long reviews on the latest gear and hardware sequencers, and short reviews on software sequencers. Your publication is distant from 'Keyboard'. Now, I have an alternative.

Gints Klimanis
Poughskeepie, NY

Dear Music Technology,

Enclosed you will find a check for a subscription to your magazine. It is indeed the light at the end of the tunnel of dreary periodicals. I often find myself on the verge of nausea reading similar magazines, because I get the impression the articles were written by moonlighting auto mechanics. Yours offers a fresher look at what's new and what will be new.

What would I like to see in your magazine? Gorgeous women modeling the latest in sound-processing equipment...then I could cancel my subscription to 'Hot Rod'.

Jeff Baudin
Novato, CA

Dear Music Technology,

Congratulations on a very impressive first issue! One thumb thru and I assumed that you had been around for quite a while.

Enclosed is my subscription check. Please don't let me miss a single issue.

The following are some of the areas that I'd like to see covered in future issues:

An explanation of how SMPTE is used to sync tape to film/video, tape to MIDI, etc. and a comparison of various products used for that.

An overview of software-based digital synthesis including how algorithms are developed (more or less) and how they work.

A practical guide to user sampling. How to make your sampling keyboard play what you imagine in your head. How the realities of physics and perhaps unrealities of one's imagination can make things difficult in that area.

I guess that's enough for now. Congratulations again. You're a welcome addition to my studio.

Robert Bolman
San Francisco, CA

NEWS EXCLUSIVE

SEQUENTIAL UNVEIL SAMPLING DRUM MACHINE/MIDI RECORDER

Latest news from Sequential, who weren't exhibiting at this year's NAMM show but who threw a party on a boat instead, are now putting the finishing touches to a studio-standard digital drum machine that also acts as a sampler and MIDI recorder.

The machine is called the Studio 440, and its 512K of memory is capable of holding a total of 32 12-bit sounds, stored as four banks of eight.

User sampling is an integral part of the machine's spec, and can be carried out at three different sample rates: 16kHz, 31kHz and 42kHz, with maximum sample times of 32, 16 and 12.5 seconds respectively.

Eight pressure-sensitive pads are mounted on the Studio 440's top panel, and these can be assigned to any voice. Four 'kits' (ie. sets of voice-to-pad assignments) can also be stored in the machine's memory, which is configured so that, for example, the eight pads can be used to trigger eight different pitch and pan variations of a single sample. Each voice also has a set of 'alternate parameters', which can comprise different settings for pitch, pan position, and dynamics as well as reverse reading of the sample as on Sequential's earlier Tom drum machine.

The digital recording side of Studio 440 is configured so that it can trigger either the machine's internal sounds, or external voices connected to a maximum 32 MIDI channels (via two MIDI Outs), or any combination of the two.

The sequencer's 200K memory is capable of storing between 50,000 and 70,000 events, depending on the kind of music you're recording and the sort of MIDI data you're storing in addition to note values. It's an eight-track affair that's programmable in both real and step time, and in addition to a host of sophisticated programming and editing facilities, it features a novel idea called Record Phrasing, in which the user inputs all the events first, and then programs a rhythm for them.

Studio 440's back panel is a haven for futurists, with Terminal In and Out sockets for the proposed MIDI-SMPTE standard discussed by Chris Meyer in the first issue of MT, and an SCSI (Small Computer System Interface) connector for hooking-up to a hard disk storage unit.

First shipments of the Studio 440 should reach stores sometime around October, though initial models may lack some of the fancier interfacing.

MORE FROM *Sequential, 3051 N. 1st St, San Jose, CA 95134 ✆ (408) 946-5240*

SUITE BEAT ADDS EUROPEAN NEW AGE LABEL

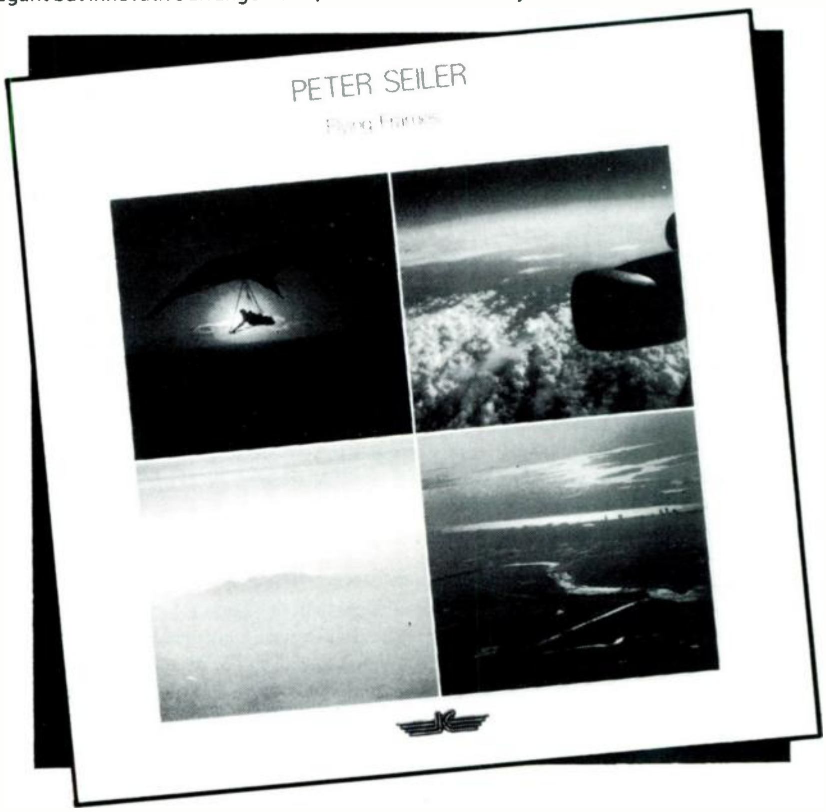
Suite Beat Music is continuing its policy of acquiring new labels to its fast growing distribution network by signing a domestic licensing agreement with Germany's highly esteemed New Age label Innovative Communications.

Formed in 1979 by Tangerine Dream founding member Klaus Shulze, IC has gained international acclaim for its state-of-the-art production, award-winning graphics, quality pressings and most of all, its musicians.

Suite Beat Music claims that: 'IC music combines beautiful, lyrical melodies with elegant but innovative arrangements, colorful instrumentation and advanced production ideas which results in not more "electronic" but more feeling for the music, human warmth and empathy for the songs, instruments and listeners alike.'

In keeping with the tradition that IC has established in Europe, Suite Beat Music Group is committed to preserving the high standards of the label and thus is mastering all albums from digital tapes, and pressing the LPs on virgin vinyl and the cassettes on chromium tape.

The first four IC releases debut July 20 on LP and cassette, with Compact Discs to follow in September. The titles of the first four albums are *Universal Avenue* by Double Fantasy, *Chip Meditation* by Software, *Beam Scape* by Mergener/Weisser and *Flying Frames* by Peter Seiler.



TOO GOOD TO BE FORGOTTEN

If you've lovingly held on to a faithful Korg Polysix, patiently forgiving it its lack of MIDI because of its huge and distinctive character, then now's the time to breathe a sigh of relief. Korg have come to the rescue with a MIDI retrofit kit.

The modification also expands the synth's program memory to 120 sounds without outdating old 32-patch tapes, and includes a sustain pedal jack as a bonus. The price is expected to be around $299 including fitting.

MORE FROM *Korg, 7 Frost St, Westbury, NY 11570. ✆ (516) 333-7100*

GAND MUSICTECH SHOW

Gand Music & Sound of Illinois will be holding the Gand Musictech '86 on September 13 and 14 at the Holiday Inn in the Chicago suburb of Skokie, Illinois. In all, over 40 manufacturers will be exhibiting. Additionally, Musictech guests will be able to attend a variety of seminars, demonstrations, and workshops conducted by top-name musicians and producers that will cover topics such as digital sampling, sound reinforcement, and recording of film soundtracks.

Some of the manufacturers scheduled to appear at Musictech are: 360 Systems, Akai, J L Cooper, E-mu Systems, Ensoniq, Fairlight, Fender, Ibanez, JBL, Lexicon, Oberheim, Otari, Peavey, Roland, Soundtracs, Star Case, Steinberger, Tascam and Ultimate Support Systems.

Scheduled to run from noon until 5.30pm each day, Gand Musictech '86 will feature many items for sale at specially discounted prices, as well as promotional giveaways.

MORE FROM *Gand Music & Sound, 780 Frontage Road, Northfield, IL 60093. ✆ (312) 446-4263 (446-GAND)* ▶

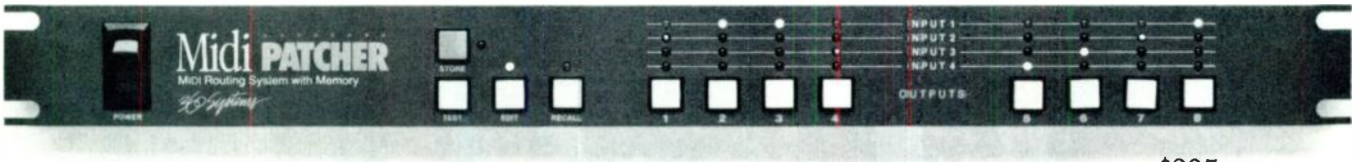
ROLAND SYMPHONY

Symphony Series now introduces for Roland keyboard and drum machine users: the S16 and S64 RAM memory cartridges – with four times the memory storage capability of Roland's own M16C and M64C memory cartridges.

The S16 and S64 RAM cartridges feature four total banks of data memory, 128-voice capability (S16), write-protect switch with red/green LED, one-year replacement warranty and high toggle memory band switching.

MORE FROM *Symphony Series, Inc., 9500 South 500 West, Suite 210, Salt Lake City, Utah 84070. ✆ (801) 566-1683*

Link your Midi instruments



$295.

with MIDI PATCHER—a 4 in, 8 out Midi routing system *with memory*

IF you could connect your master keyboards to different Midi sound sources, at different times—you would have the power to create musical arrangements never before possible. Now, every important link can be remembered, recalled, changed and instantly compared to others.

Midi Patcher does more than organize your wiring—it displays every connection on its front panel with four-color LED's. It even changes routing while you play, either from the front panel or by Midi.

With Midi Patcher, the complexity of large systems becomes manageable under intelligent micro-computer control. Available rack mount or stand-alone, at high-tech keyboard stores everywhere.

360 Systems

18730 Oxnard Street • Tarzana, California 91356

Hear the MIDI BASS sampled sounds on this new cassette

At $399, Midi Bass adds a real bass player to Midi keyboards, guitars, pedals, and sequencers. Now multi-sample digital recordings can be part of your performance. Each sound is a perfect copy of a real bass. Midi Bass even copies dynamics and pitch bends.

NEW ALTERNATE SOUNDS

We've added the best of Rock, Tech-Pop, Funk, and Country to our expanding library. Get the Midi Bass cassette and hear the entire collection of alternate sounds from this remarkable bass player. Each sound is performed by itself, and again with a rhythm section. Included with each cassette is a detailed description of every sound, and a complete library list/order form.

Get the Midi Bass cassette from your dealer, or send $5 cash or money order to 360 Systems, 18730 Oxnard Street, #214, Tarzana, CA 91356.

360 Systems

$399 MIDI BASS with two sounds

Send me the Midi Bass Cassette
Enclosed is $5 for each tape
(Outside U.S. add $1.)

Name ____________________

Street ____________________

City/State/Zip ____________________

Please check the boxes that apply:
☐ I own a Midi Bass. I use it for ☐ recording ☐ live.
☐ I don't have one yet. Send info.

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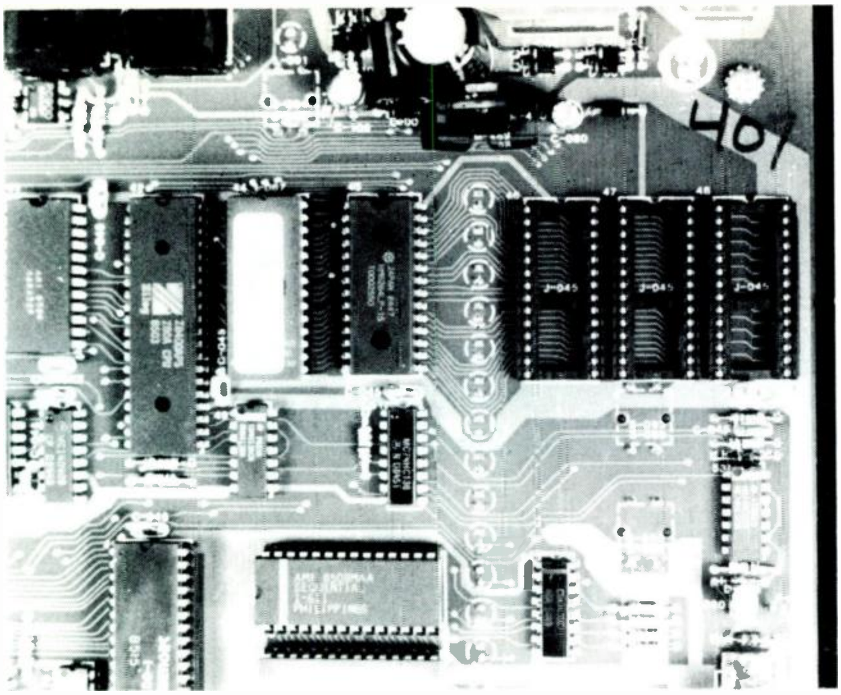

NEW DRUMWARE SOUNDS FOR TOM

Drumware has announced six new cartridges for the Sequential TOM drum machine. These collections of new sounds should come as a pleasant surprise to TOM owners wanting alternatives to Sequential's four cartridges (which appear to be the only ones planned by the manufacturer). The new sound cartridges are Analog Drums (à la Roland TR808), Percussion 1 (includes triangle, fingersnaps, castanets, and ambient congas), Percussion 2 (featuring brakedrum, ceramic logdrum, and slitdrum), Rock Drums ('big' sounds), Atomic Drums (even bigger), and Hippifex 1 (featuring anvil, bursting bulb, and paper bag). Unlike the Sequential cartridges, Drumware's do not come with a mylar overlay for the TOM's front panel, but considering that each cartridge retails for $99, that probably won't put off many TOM owners in need of some new sounds.

MORE FROM *Drumware, 12077 Wilshire Blvd., #515, Los Angeles, CA 90025. ✆ (213) 478-3956*

THE JUST INTONATION NETWORK

The Just Intonation Network has just announced the first issue of a new publication called 1/1 (pronounced one-one). 1/1 is to be the quarterly journal of the organization, which was founded by a group of Bay Area composers for the purpose of encouraging communication and mutual support among composers, musicians, and instrument builders exploring this alternate musical frontier.

In future issues of 1/1, you can expect articles on a comprehensive array of topics, both theoretical and practical, of interest to the just musician. Topics include acoustic, electric, and electronic instrument building and modification; composer interviews; in-depth analyses of specific scales and musical styles; book and record reviews; tutorials for the novice; and discussions of the crucial issues by composers leading the exploration of new intonational realms.

MORE FROM *The Just Information Network, 535 Stevenson Street, San Francisco, CA 94103. ✆ 415 864-8123*

MELLOTRON RETURNS

Under the same name as the keyboard first introduced back in the early sixties (and still used by many artists including MT contributor Patrick Moraz), the new Mellotron ES-2000XT has been announced.

It's an IBM-compatible music system featuring MESA (Music Editing, Scoring, and Arranging – a user-friendly MIDI multitrack recording/sequencing package), a Voice Manager (which stores and retrieves thousands of Yamaha DX7, DX5, and TX sounds direct from hard disk in seconds), and a MIDI Director.

When coupled with the Mellotron MIDI IV-port card, this last item turns the ES-2000XT into a MIDI control base. There are four MIDI inputs and four MIDI outputs that provide the user with full programmable control over all MIDI data, including the ability to re-route channel information or filter unwanted commands. The user can delay the signal on any output, providing complete integrated control over the entire MIDI system, say Mellotron. Current status of all MIDI keyboards can be displayed on the ES-2000XT color CRT screen, eliminating much of the guesswork common when using multiple MIDI keyboard instruments.

The Mellotron ES-2000XT's 20-Megabyte hard disk storage medium provides fast, accurate downloading of instrument sounds, complete song sets, songs and eliminates the need for disk shuffling between songs.

The Mellotron ES-2000XT's standard features include a Princeton EGA Color Monitor (built in), 640K of RAM, 1-5¼ floppy disk drives, 1-20 Meg hard disk, and a QWERTY keyboard. System options offer expansion slot accommodation for the Mellotron MIDI IV-port card, M-DIT modem for new sound and song access to Mellotron Library via touch tone phone, MESA music editor, scorer, arranger, mouse, and perhaps most excitingly (in view of the company's history using real sounds), the Electric Symphony digital sound sampler module.

Retail prices are ES-2000XT $3,995; ES-2000XT MIDI Controller $5995; ES-2000XT with Electric Symphony module $7995

MORE FROM *Mellotron, 36 Main Street, Port Washington, NY 11050. ✆ (516) 944-6789*

STEPPS AHEAD FOR THE ELECTRONIC GUITAR

Across the way from the NAMM show proper, MT spies were lucky enough to get a sneak preview of the Stepp DG1 electronic guitar. The instrument has been developed over a number of years by a small British company, and huge sums of research money have been spent in the process.

At first sight, the DG1 seems to be following in the footsteps of the more expensive SynthAxe: it has a moulded body, and two sets of strings – one for picking and one for fretting.

The biggest difference is that the DG1 contains its own voicing electronics and is therefore a self-contained instrument – just plug it into an amp and away you go. And as the multi-timbral synth voices are controlled directly from the guitar via an interface several times faster than MIDI, there's no perceptible delay between plucking any string and hearing the synthesized sound. There are many nuances of guitar-playing that don't translate too readily via MIDI, and the DG1 system is designed to exploit these, too.

MIDI, however, is an important feature, and is supported not only to control external keyboards or voice units from MIDI Out, but also to allow the DG1 voices to be accessed via MIDI In.

The DG1's design incorporates some interesting technological features, particularly in the neck. The frets themselves are made from a semiconductor material which translates the string position into bend information. And it's because pitch information is originated by the fretting system that the sort of tuning problems, delays and mis-tracking that plague pitch-to-voltage guitars are eliminated on the DG1.

The synth voicing circuitry employs digitally-controlled analog elements, forming an architecture which, according to the designers, has been conceived with the guitar-like nature of the instrument in mind – though the potential of the system should be great enough to encompass a broad range of different sounds. So that musicians can start using the DG1 immediately, the voicing section comes with 100 useful preset sounds – though all synth parameters are user-programmable.

The designers have succeeded in keeping the size and weight of the DG1 in line with that of a normal electric guitar by relocating much of the electronics, including the voice generators and power supply, in the instrument's custom-designed (and rather neat) stand. And the guitar, like the stand, looks tasteful and well thought-out.

As soon as we get our hands on a finished example of the DG1, we'll compile an 'official' review. In the meantime, final production work is proceeding apace, with widespread availability a matter of weeks away.

MORE FROM *Stepp, 8 Primrose Mews, Sharpleshall Street, London NW1 8YL, England*

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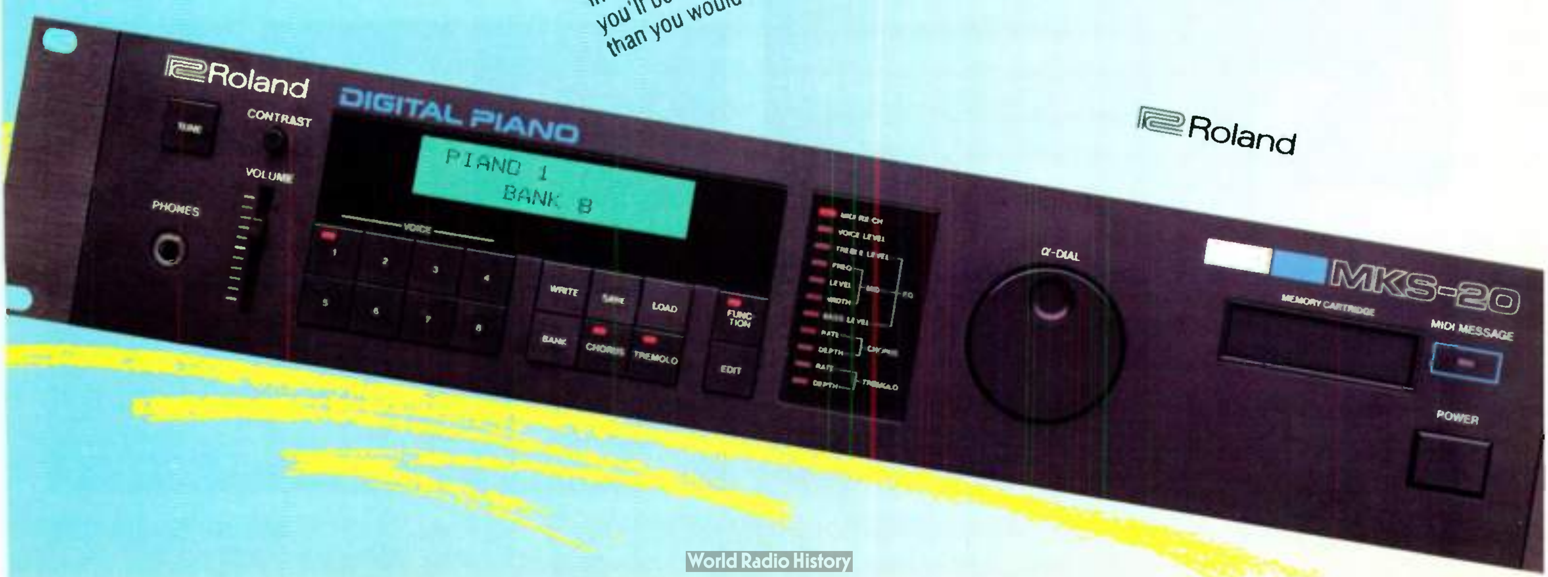
RD-1000/MKS-20 DIGITAL PIANOS

GRAND ILLUSION

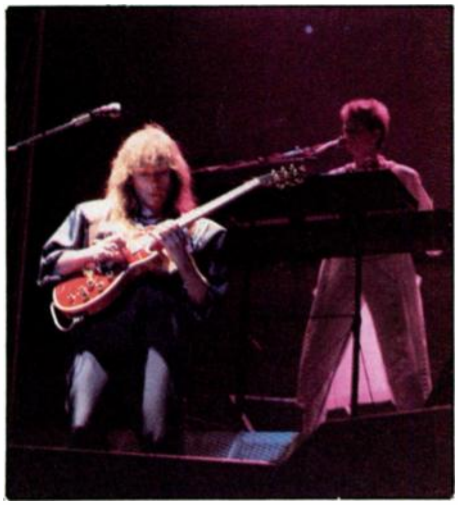
Until now electronic attempts to duplicate, by sampling or synthesis, the grand piano's complex sound have always been disappointing. Not so with Roland's new proprietary digital technology–SA (Structured Adaptive) Sound Synthesis–which neither approximates nor simulates an acoustic piano, but actually recreates the sound. Note for note, nuance for nuance–replicating faithfully the harmonic characteristics and timbre variations across the entire keyboard while responding to playing dynamics with extraordinary accuracy and warmth. Roland's is indeed a grand illusion–but one born of the noblest intentions: to offer musicians a modern, practical and more affordable version of the revered grand piano. Yet Roland has accomplished more: the RD-1000 Digital Piano and the MKS-20 Digital Piano Module share the same polyphonic, 16 voice SA technology that offers eight different velocity sensitive sounds–two classic grand pianos, electric grand, harpsichord, clavinet, vibraphone, and two distinctive electric pianos. Each preset is modifiable by a simple twist of the ergonomically concieved Alpha-Dial–vary numerous parameters including digital parametric eq. settings, tuning, volume, tremolo and chorus rate settings, then store those changes into 56 internal memories or on an M-16C external cartridge. The RD-1000 is a stand alone, velocity sensitive keyboard instrument, with 88 wooden keys and a balanced weighted action making it a powerful MIDI controller for other instruments as well. The compact rack-mountable MKS-20 Digital Piano Module, designed to be an add-on unit, provides the same keyboard voices under the control of any MIDI controller–keyboard synthesizer, sequencer, Roland's Pad-8 Octapad, even a guitar synthesizer. Roland invites you to audition the remarkable RD-1000 and MKS-20 because with any fine instrument, hearing is believing. But even beyond the hearing and feeling we know you'll be amazed by the fact that these instruments cost quite a few grand less than you would expect. And that's the best part of the illusion.

Roland



ON STAGE

M U S I C

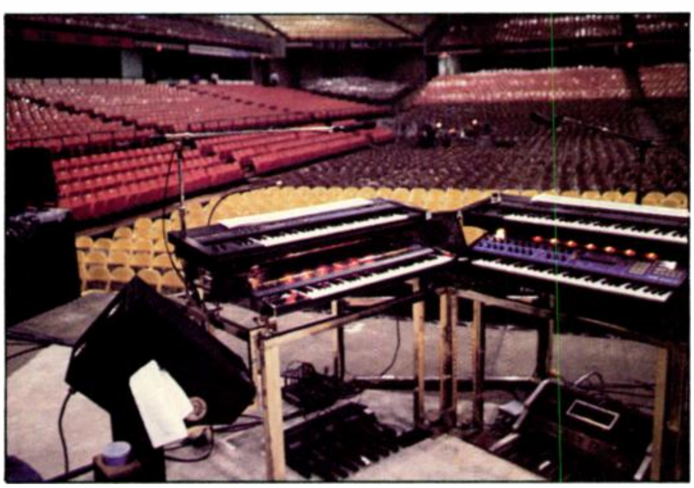


GTR

STEVE HOWE: 'I'm using the Ibanez MIDI system because the delay in triggering is miniscule, some 20ms compared to 60, but I'm still using the actual Roland synth pick-up on a Gibson Les Paul. I sometimes use the straight sound of the Roland guitar with say the OSCar and Super Jupiter to strengthen a lead line. Other guitars I use live are the Gibson 175, the Telecaster, the Sitar Guitar and of course the Pedal Steel.'

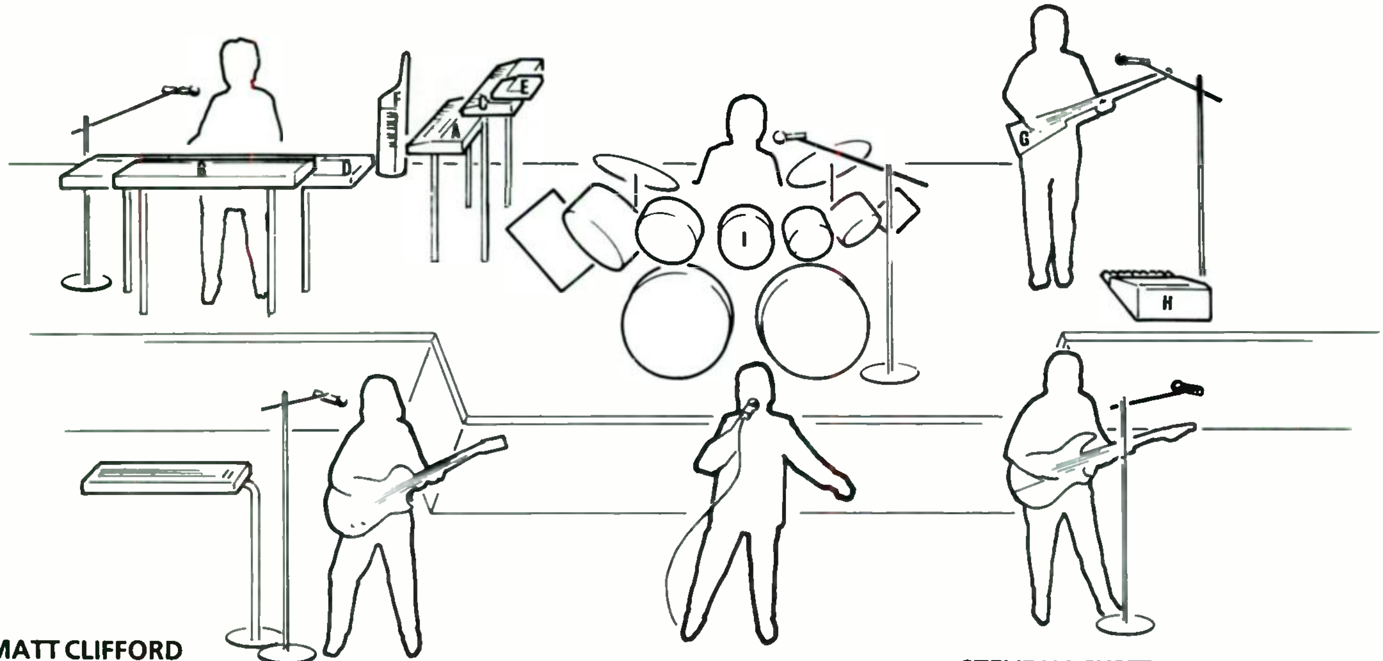
Steve (Hackett) uses both the Roland GR300 and 700 floor units as well as a lot of straight guitar treated through things like the Ibanez Harmonizer (transposed up a Major 3rd in Jekyll and Hyde for example).

MATT CLIFFORD: 'Virtually everything in the set is played live except for the intro to 'Still Get Through' which is sequenced on the QX7, with the TR707 just being used to control the tempo (not for the drum sounds). All my MIDI configurations played on the KX88, are store on the Sycologic routing box which is wonderful. All I have to do is use the remote to call up the patch and it sets up everything.'



Rush

Instead of merely recording music in the studio which they knew they would be able to reproduce live, Rush recorded their new album exactly how they wanted it to sound, and then left it up to Tony Geranios to design the entire stage system to cope with the demands of the studio recordings. In order to keep control of everything that is going on on-stage, Tony uses the J L Cooper MSB16-20 to receive the incoming MIDI signals from Geddy Lee's keyboards and the Korg MPK130 MIDI Bass pedals which both Geddy Lee and Alex Lifeson use and routes them to the DX7 and the Emulator IIs (no less than five). Live, Tony doesn't actually *play* anything at all, but loads up disks and switches all the MIDI routings.



MATT CLIFFORD

A Korg PolySix
B Yamaha DX7
C Yamaha DX100
D Yamaha KX88 Master Keyboard,
E Yamaha QX7 and
F Yamaha KX5 Remote
controlling (off-stage rack):
2 Prophet 2002s
Roland MKS80 Super Jupiter
Roland MKS20 Digital Piano
Korg DVP1
Yamaha TX7
Sycologic M16 MIDI routing box
Yamaha R1000
Yamaha D1500
Roland SD2500

STEVE HOWE

Gibson Les Paul with Roland pick-up, controlling
(off-stage rack):
Ibanez MIDI Guitar Brain
Roland MKS80 Super Jupiter
Yamaha TX7
OSCar
Yamaha SPX90s (2)
Roland SDE3000 (2)
Roland SDE2000 (2) and
Roland SRV2000
(All audio and MIDI signals routed through Quark custom pedalboard which also incorporates various T.C. effects pedals including 2 sustain units, 2 booster/distortion units, chorus/flanger, plus a Korg Mr. Multi auto-wah/flanger)

STEVE HACKETT

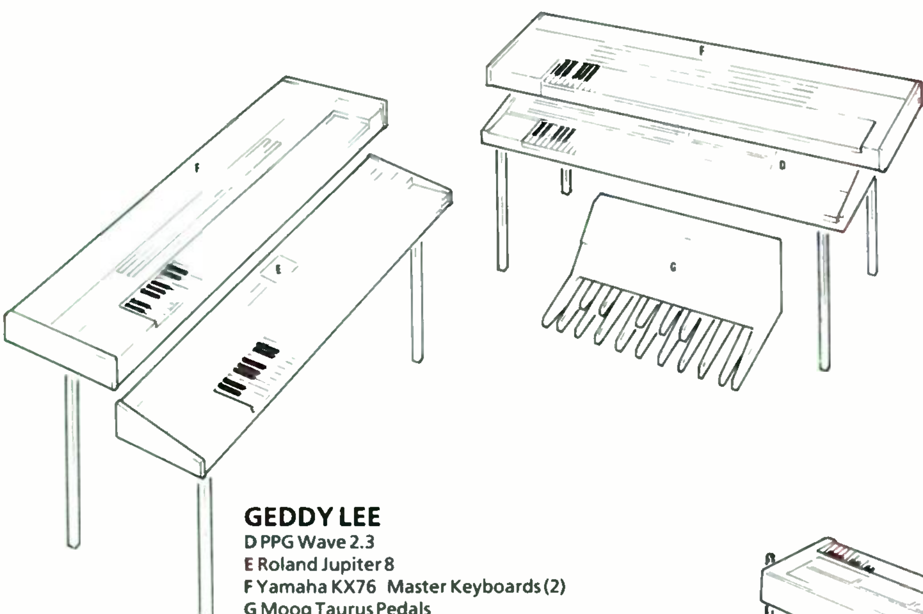
Roland G505 Guitar controller driving:
Roland GR700 Guitar Synth,
Roland GR300 Guitar Synth,
Roland SDE3000 Reverb,
Roland RE501 Chorus/Echo,
Ibanez HD1000 Harmonizer and
Boss Chorus
Pete Cornish pedalboard

PHIL SPALDING

G Steinberger Bass Guitar
H Moog Taurus Pedals

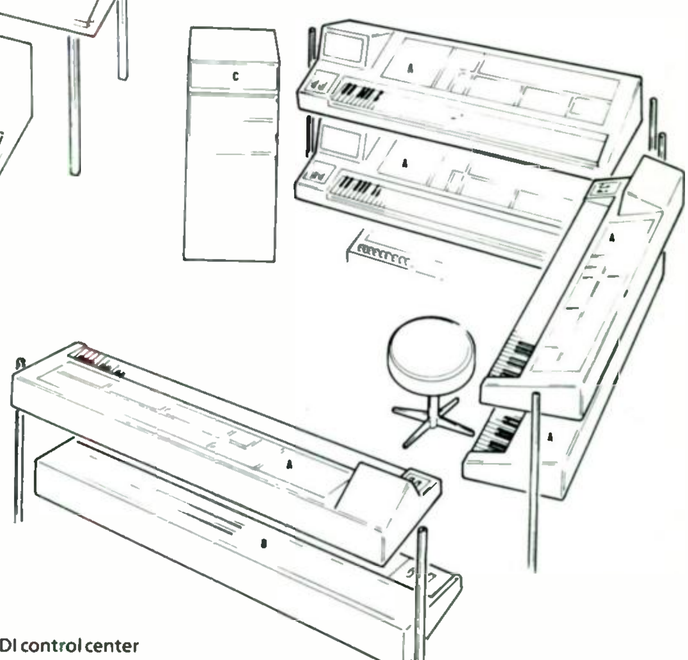
JONATHAN MOVER

I Pearl Drums



GEDDY LEE

D PPG Wave 2.3
E Roland Jupiter 8
F Yamaha KX76 Master Keyboards (2)
G Moog Taurus Pedals



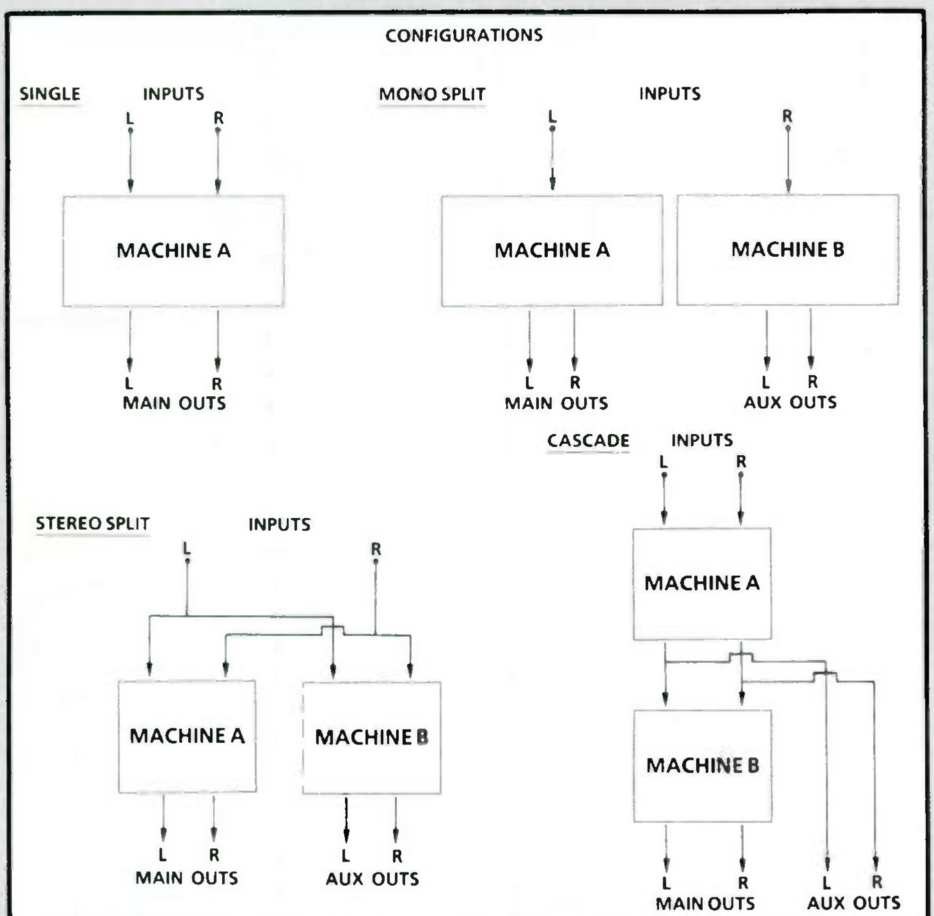
TONY GERANIOS

A Emulator IIs (5)
B Yamaha DX7
C JL Cooper MSB16-20 MIDI control center

Lexicon Model 480L
Digital Effects System

Photos: Rick Davies.

The 224X has become the standard digital reverb for many studios, but now Lexicon has come up with a new unit which not only doubles the processing capacity, but adds a whole host of software-generated effects. *Preview by Paul Wiffen.*



YOU THOUGHT THE LEXICON 224X WAS THE LAST WORD in digital reverb? You thought wrong. Because from what we saw in Bertus Studio in the San Fernando Valley on a scorching hot afternoon recently, it looks like Lexicon has another standard-bearer on their hands, though as it turns out, direct comparisons between the new 480L digital effects system and earlier Lexicon machines are of fairly limited value.

To start with, the 480L has four processors to perform its functions, which start with digital reverb but certainly don't end there. These are grouped into two stereo in/stereo out machines internally (A and B), which can be run independently (giving you the power of, say, two 224Xs) or configured together in several ways (see accompanying diagram) to provide multiple effects. Perhaps the most exciting of these is the Cascade configuration, which actually takes the output of machine A and transfers it to the inputs of Machine B on the digital lines so that there is no need for the signal to be turned back to analog, sent into the next signal processor and reconverted to digital (with the loss in signal quality that entails). And on the same lines, there's a PCM1610-compatible digital I/O (input/output) port which allows signals from digital multitrack machines like the Sony and Mitsubishi models to be processed through the 480L without leaving the digital domain. In other words, they are fed in through the digital audio I/O as undecoded digital data from the tape and then processed and sent back to the tape machine still in ►

480L programs and their locations

Bank 1	Halls	Concertgebouw	Carnegie	Philharmonic	Cathedral	Recital Hall	E.G.Hall	Small Wave	Big Wave	High School	Park it Here
Bank 2	Rooms	Regular Room	Live Room	Roomy	Living Room	Hard Chamber	Small & Dark	Small & Bright	Varoom		
Bank 3	Wild Spaces	Brick Wall	Buckram	Big Bottom	10W–40	20W–50	Metallica	Silica Beads	Inside Out	Ricochet	
Bank 4	Plates	A Plate	Snare Plate	Small Plate	Thin Plate	Fat Plate					
Bank 5	Effects	Illusion	Surfin'	Voc Whispers	Doubler	Back Slap	Rebound	Elinar	Sudden Stop	Wah	
Bank 6	Sampling	Sampling									
Bank 7	So What Else	Twin Delays	The In/Out								

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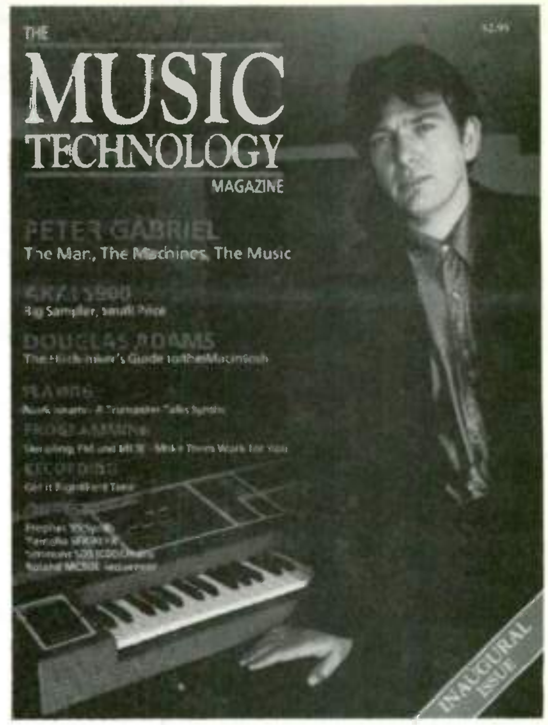

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FROM THE CLOSED ROLL TO THE OPEN MIND

SDE. NEW NOISES FOR MODERN DRUMMERS

GROUP CENTRE INC., 23917 CRAFTSMAN RD., CALABASAS, CA 91302, (818) 884-2653 digital format. This is possible because the Lexicon's sampling rates (48kHz, 44.1kHz or 44.056kHz) can be slaved to an external clock.

The 480L's 16-bit system (18-bit analysis, 16-bit replay, giving true 16-bit perfomance) is equal to (44.1kHz sample rate) or better than (48kHz sample rate) Compact Disc quality. When the machine is used in its sampling application, sample time available is 1.5 seconds per processor, giving a total of six seconds for a mono sample, or three seconds of stereo. Typical dynamic range is 100dB, with a 96dB minimum.

The 480L is controlled by the same remote as the 224X – the LARC (Lexicon Alphanumeric Remote Control). This will be good news to busy studio engineers, who often don't have time to keep learning new programming processes. Both programs and parameters can be accessed and changed using the LARC.

Basically, there are (at the moment) three separate levels of parameters, with complete access between levels possible at any time for each machine. If you want to, you can connect two LARCs to one 480L, allowing two people to timeshare the mainframe. What this jargon means is that, say, in a film studio application, the effects man can control machine A while the music engineer can use machine B, completely independent of one another. This is possible because the 480L has four separate outputs, two for each machine.

Alternatively – if you already have a 224X with a LARC – you can just buy the 480L mainframe and control its two machines, plus the 224X, with just one LARC.

The initial programs stored in the demonstration 480L sounded very impressive. Besides the traditional collections of Halls, Rooms and Chambers, and the more modern gated and reversed Reverbs, there were a host of new signal processing effects which rivalled the Yamaha SPX90 for versatility, but had the extra quality afforded them by higher bandwidth and signal-to-noise specs. And just in case the range of facilities becomes too much for over-eager programmers, the entire program memory of the 480L can be dumped onto a RAM cartridge, allowing a huge library of effects to be built up.

Lexicon has also added their Dynamic MIDI (so successful on the PCM70) to the 480L, which means that in addition to pitch control of samples and program changes from controlling keyboards, you are also able to sequence parameter changes via a MIDI sequencer in an automated-style mix.

And speaking of automation, there's a DE9 port on the back panel which, in a future expansion, will allow direct automation of the 480L. There's also talk of using personal computers such as the Atari STs and the Mac (possibly even Digidesign's Sound Designer package) to store samples saved via the digital I/O port.

A brief glimpse inside the mainframe showed us a very neat plug-in modular system which allows plenty of room for expansion and makes servicing or updates a breeze. In addition, there'll be diagnostic routines controlled and displayed via the LARC, so identifying a possible fault and its location should be quickly and confidently achieved.

By the time of the 480L's official release (September 1), Lexicon plans to have the final basic version of the software finished, which will include new programs in the Single configuration which combines the power of all four programs (from both machines) to achieve powerful, complex audio effects. From there on, regular software updates should continue to expand the range of effects that the 480L can produce.

With this future expandability and a quality and flexibility that have already been proven by the 224X, the Lexicon 480L looks set to take its manufacturers further into unexplored areas of digital technology, and take musicians and studio engineers/producers with them. Talking of which, just wait and see what Lexicon has coming in the field of digital recording at the November AES... ■

PRICE Mainframe 480L $8,200 with LARC remote $9,800

MORE FROM Lexicon Inc., 60 Turner St, Waltham, MA 02154 ✆ (617) 891 6790

480L parameters

Reverb-Variable Parameters

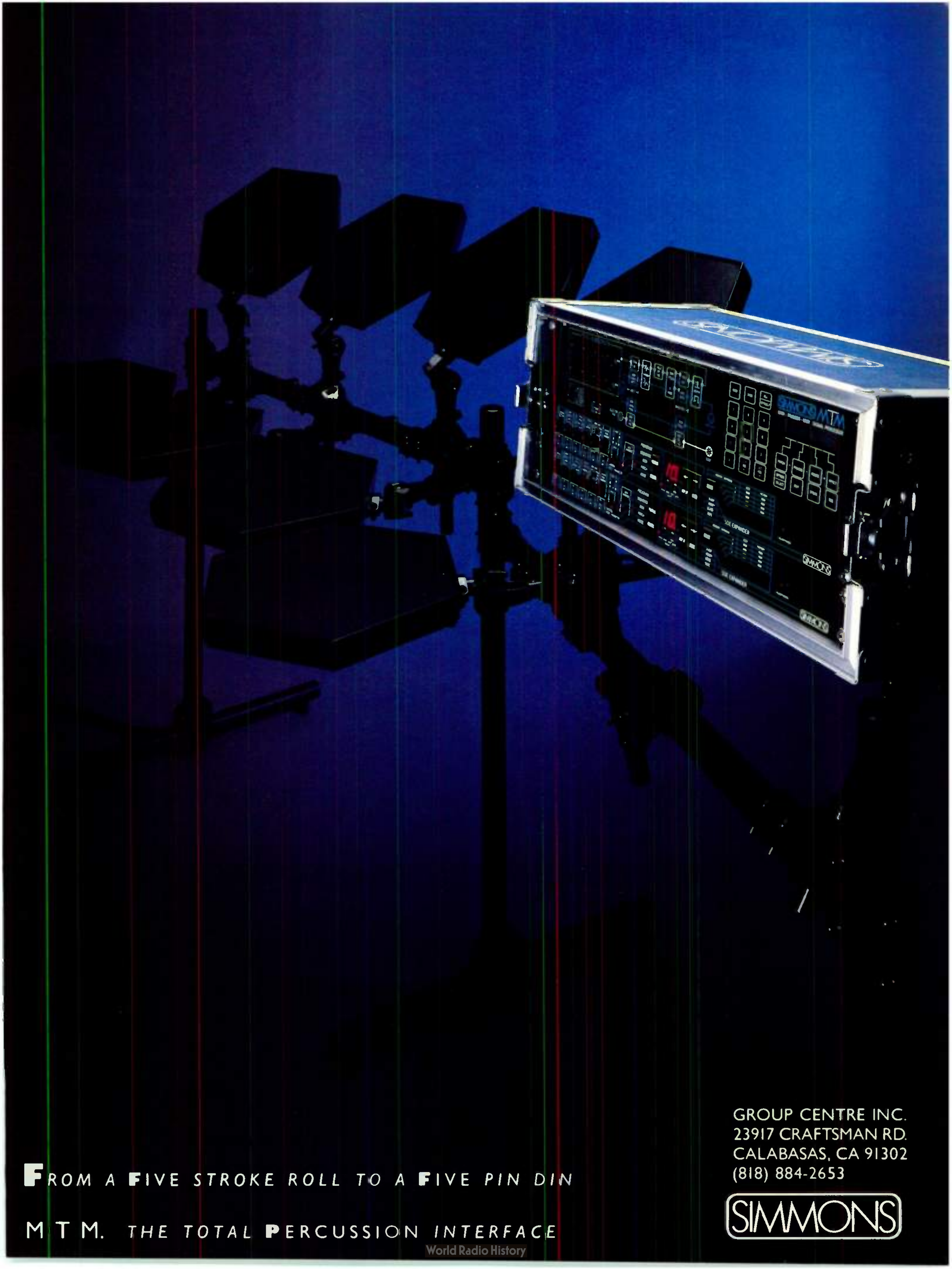
Page	Parameters					
1	MF Reverb Time	Shape	Spread	Size	HF	Predelay Bandwidth
2	Bass Multiply	Crossover	Treb Decay	Diffusion	Decay Opt.	Wet/Dry Mix
3	Pre-echo Level L›L	Pre-echo Level R›R	Pre-echo Level R›L	Pre-echo Level L›R	Pre-echo Level L›L	Pre-echo Level R›R
4	Pre-echo Delay L›L	Pre-echo Delay R›R	Pre-echo Delay R›L	Pre-echo Delay L›R	Pre-echo Delay L›L	Pre-echo Delay R›R

Effects-Variable Parameters

Page	Parameters					
1	Spin	Shape	Spread	Wander	Number	Pre-delay
2	Input Blend	Feedback Level	Feedback Delay	Diffusion Delay	Input Mix	Wet/Dry
3	High Pass Left	High Pass Right				

Sampling-Controls and Parameters

Page	Parameters					
1	Play	Play Repeats	Inactive	Record	Inactive	Check Play
2	Head Trim	Forward Play Time	Play Order	Tail Trim	Reverse Play Time	Cue
3	Forward Play Lev.	Reverse Play Lev.	Inactive	Inactive	Inactive	Cue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AND THEN THERE WAS ONE...

Nearly 20 years of rock keyboard-playing have made Tony Banks a unique figure: a virtuoso musician whose enthusiasm for new sounds and new musical styles has never been dulled, and whose influence over fellow artists has been immense. As his 16th album with Genesis is released, Banks has plenty to say. *Interview by Tim Goodyer*

SONGWRITING, CULT SUCCESS, POP STARDOM AND FILM-SCORING have made Tony Banks' name one of the best-known and respected among rock keyboard players.

It all began in 1969, when an album titled *From Genesis to Revelation*, produced by Jonathan King, was unleashed on an unsuspecting and largely unappreciative record-buying public. The album contained the songwriting endeavors of five young hopefuls, among them Banks, Mike Rutherford and Peter Gabriel.

Since those humble beginnings, much has happened to both the music and the line-up of the band, but Genesis have enjoyed one of the most consistently successful careers of any in rock music. I spoke to Banks at Genesis' own Surrey studio on the eve of the release of their 16th album, *Invisible Touch*. It's perhaps the band's most overtly commercial offering to date, with many of its pieces sounding more like out-takes from a Phil Collins album than anything else. There are some surprises, though, notably the beautifully textured 'Tonight, Tonight, Tonight', and an avant-garde instrumental titled 'The Brazilian', which brings the disc to a close.

Coincidentally, Banks has a new solo offering, *Soundtracks*, in the record stores, too. It showcases Banks' most recent composing, programming and performing work, and abounds with the sort of hot-paced sequencer patterns so beloved of the film industry for chase scenes. With Banks at the controls, though, the music goes through a number of dramatic structural and melodic shifts, unexpected but never so contrived as to be unwelcome, as Banks explains:

'All the stuff I did for the soundtracks album I'd actually done before the Genesis album. People would say "what have you been doing while Phil's been making money and Mike's been doing so well?" so I just wanted to get these things out.

'There were a couple of films that I'd done the soundtracks for over the last two years. One of them, *Quicksilver*, came out in America and didn't go down very well over there, so they're not even bothering to release it here. It's a shame because I didn't think the film was that bad, though quite honestly, I thought the rest of the music for it was. I tend to be difficult to please but I felt that it was everybody's second best. So I thought I'd take my part out and put it together with the music I'd done for another film called *Lorca and the Outlaws*.'

Banks' own efforts at singing have taken a back seat to the talents of Toyah and Jim Diamond; Toyah's 'Lion of Symmetry' and Diamond's 'You Call This Victory' are both taken from *Lorca and the Outlaws*. But curiously, it's Marillion's Fish that takes the lead in 'Shortcut to Somewhere', from *Quicksilver*. Marillion frequently stand accused of being no more than a poor man's Genesis, so the association is as intriguing as it is unlikely.

'The idea of a collaboration with Fish was like walking into the problem, which was quite nice', says Banks. 'It simply amused me, really, because they've always been compared with us. I like the approach of Marillion's music more than the result. There's no doubt that Fish's voice does bear an uncanny resemblance to Peter Gabriel's in the old days, but the way he gets to that kind of sound is completely different. There's no contrived way of doing it, it's just the way he ends up sounding.

'The song we've done together is quite strong, and if it gets a chance to be released as a single, it'll do quite well. But with a joint thing, the problem is always getting permission from the record companies to release it.'

In 1986, the output from an ageing Genesis consortium is prolific. Aside from Banks' *Soundtracks*, it's responsible for Rutherford's successful Mike & the Mechanics project, and for Phil Collins' even more successful solo work. In the light of all this activity, I wonder (needlessly, as it turns out) how 'Invisible Touch' the single will fare. My concern is anticipated by a perceptive Banks.

'There is a danger of saturating the market', he concedes. 'I'm a bit terrified, suddenly seeing the competition from our own buddies around like Peter's 'Sledgehammer', Mike & the Mechanics, and obviously Phil who doesn't stop selling. But, for me, Genesis is a combination of the three of us. I probably take it the furthest from the mainstream, and I suppose Phil brings it closest.

'I'm proud of every song on this album. I feel very strongly that all the songs are products of the combination of the three of us being in the same room at the same time. It's what I've had brought out of me when I'm working with Mike and vice versa, and I tend to make Phil do things that perhaps he wouldn't do on his own.

'I think it works better because we're doing different things on our own. When we get back together it's like friends getting together again, and things seem to happen a little bit differently. It's terribly difficult to find out *why* things work. It's worked for us for a long time whereas most other groups don't manage to stay together for any length of time.

'We're a fairly unique group in that we're all involved in the writing of the music. In every other band it's one or two people that look after the writing. When all of you are involved and one of you goes, it changes the whole.'

How does this three-way writing team work?

'It's improvization, really, just like you'd do on your own, but with the three of us. When I'm writing a song I'll sit down and play the piano for hours and things will evolve that I'll try to develop. We do it the same with the three of us. We've been playing together for so long and know each other so well that there are assumptions you can make. I don't think you can get just any three people, put them in the same room and expect them to come up with the right songs. A thing only works if you think the result works.'

This level of maturity obviously makes songwriting a civilized process, though it wasn't always that way, as Banks recalls.

'The three of us have actually been in the same group since 1969. In that time the emphasis has shifted around within the band quite a few times. Ever since we first became a five-piece, we've tried to maintain a situation where we'd all be writing together and trying to listen to each other. With the five of us there was a tendency at certain times for those who shouted the most to get their own way. I tended to be quite a loud shouter, as did Peter. We used to have quite a few arguments in those days and shedding the extra members made things a lot easier.'

A controlled amount of friction between songwriting partners can be a useful factor in bringing out the best ideas from those involved...

'I think friction is the wrong word because we all used to care a lot about everything. We'd argue about one bar or something and people would storm out of the room. I don't think it's an unhealthy thing, but I don't think it's essential either. I think it's something that can help, certainly, but we used to argue about the most stupid things. We argue much less now. We probably avoid the arguments because we know where

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they're going to come. I know that if I start using diminished chords I'll get "a look". That doesn't mean I can never use them again; I can slip them in without anyone noticing, but I know how hard to push and whether or not to really go for something.'

In 1974, the line-up that put Genesis on the back of a thousand denim jackets included guitarist Steve Hackett and frontman Peter Gabriel. To many, Genesis *was* Gabriel, and his departure marked the end of an era. Twelve years on, Banks reflects on the split.

'Peter and I were very close friends and we both felt very strongly about things. We got on very well and when he decided he wanted to go, I did try very hard to persuade him not to. I think it was a necessary thing and, as it's turned out, it's worked very well. We had to lose somebody and Peter was the only one who had any chance of a successful solo career. The rest of us had been completely in his shadow at that stage...but we had the self-confidence that if we lost one member, we could still produce music just as well, as long as we could get over the problem of who sang.'

Auditions were held to fill the post, yet it was Collins who subsequently stepped into the singing spotlight. But Gabriel's departure deprived Genesis of more than just its voice.

'Peter's very good with sound – he always was. He was always the one that would like to do something that was a bit bizarre, even for the sake of it. At times I find that he almost goes too far that way, and I get a bit irritated by it. I love his last three albums in particular – I think they're tremendous. But with the fourth album, one month I thought I loved it and the next month I thought: no, this is too much, I can't take it. It was overworked. My favorite is definitely the third one – all those drum sounds and everything.'

Curiously, the percussion sounds that have marked one facet of Gabriel's innovation turn out to have their roots back in the Genesis days.

'It stemmed from the new technology combined with Peter's insistence to try not to use cymbals. But it's something we used to discuss in the old days: cymbals occupy an awful lot of the sound spectrum within a song. As soon as you stop using cymbals, you can start to use the resonance of the drums to a much greater extent because you can actually hear them. As soon as you've got that ambient quality back, you can start compressing them and lifting them up.

'A drummer will keep on hitting cymbals as he's going along, at rather random intervals. They get all these cymbals that are supposed to sound different but, to me, there are only three: little ones, big ones and dustbin lids, and the dustbin lids are the ones you've really got to avoid. If you use them as a sort of punctuation mark they add such a lot when they come in, but if they're there all the time, you can't add to them.

'As a keyboard player, when you're trying to produce nice wide sounds that have got sparkle to them, a cymbal will make it all appear dull. You lose all the top and the keyboard becomes mellow. And when you *do* want a mellow sound on a piece with cymbals, you've got to brighten it up so much that on its own it sounds so bright it'll still show through.'

Banks' reputation as a musician is based firmly in his classical background and use of the piano. Yet conversely, the Genesis sound we've come to know over the last decade relies heavily on synthesizers and sequencers. Have we seen the last of the piano and its disciplines?

'I don't actually practice piano at all now, but I still like playing it. One of the nicest things I recently acquired was a piano-to-MIDI interface. It means I can use my CP70 and play any other instrument from that. Unfortunately I was only able to use it on a couple of things on the album because it came so late, but I think it's the most exciting thing that's happened for a long time.

'I don't care what people do with all their keyboard touches and things, they just don't feel like a piano to me. I used to hate the feel of a Rhodes, for example, but I liked the sound, so I occasionally tried to play it. With the DX7 now, you can get that sound with a reasonable touch, but it will be even better to have the DX7 sounds played from an ordinary piano keyboard.

'Most of my synthesizer sounds these days come from either the Emulator II or the DX7, but I use a lot of other things like Prophets, Synclavier and a Super Jupiter – anything that's lying around really.

'A lot of the sound qualities are created by using effects, though, particularly the Yamaha REV7 reverb. I find it better than the AMS for keyboards because you've got a wider variety of possibilities. It makes an instrument find its space. On its own the DX7 is a bit crude and naked, it needs something else and it responds to help better than almost anything else. The REV7's perfect for providing that.'

The sophistication of contemporary digital technology marks a far cry from Banks' early days, hunched over a grand piano, Hammond organ, Mellotron, and perhaps an ARP Pro Soloist preset synth. But even then, new sounds were a prime consideration.

'When I had the Hammond I tried to get as many sounds out of it as I possibly could', affirms Banks. 'With fuzz-boxes and things it sounded as if we had synthesizers before we actually did.

'Now you've really got to take time to explore instruments. Say you get something like a DX7, which is really a pretty simple instrument. There's no way you can begin to explore all its possibilities, there's just too much variety. And if you get into samplers, obviously the scope is even wider. Sometimes it's easier when you've got a more restricted format because you know where you stand. When you've got totally open possibilities, things can get a bit frightening.

'But you can just stumble across things. With the Emulator, in particular, a lot of the sounds I use are ones I've stumbled across. I find the Emulator a useful tool for composition, too. What I often do is switch it on while we're improvizing, and I get 17 seconds of everybody doing their thing and not even listening to each other. Then I play through it and sometimes there's something there. You edit out a few seconds and you've got something you can work with. On the new Genesis album there's a number called 'The Brazilian' that's got what sounds like a sequence pattern going through, which was done like that.

'On 'That's All' on the last album I got the main riff that way. When I played two notes of one of these samples at the same time, this riff evolved out of nowhere that didn't seem to be in either note individually. It was played over an octave so the two parts were related by half-speed, but the effect was a riff.

'Another time I was trying to sample a cello sound off a disc and I sampled four notes. By playing them all together, they interweaved and I got this repeating pattern. You can use an Emulator in hundreds of ways, and I haven't *touched* the sequencer yet.'

The subject of sequencing is often a sore point with the classically trained, but Banks offers an objective view.

'With every instrument you buy these days, you're paying for so much guff that you're never going to need. Synthesizers all have sequencers built into them, but you can only use one of them at a time and it's often easier to have that as a piece of outboard gear anyhow. It wouldn't be that expensive when you consider you've paid for the thing ten times over already. I'd prefer to put more money into sample length or something.

'I've always hated sequencers but I've always been fond of using bass patterns so you can build up from there. The trouble is that they're so abused in pop music – that's what makes half of it so dull to me. You've got your rhythm machine, then you've got your bassline and the interplay between them is exciting for about ten seconds, then it goes on like that with an adequate voice and chords on top and that's it – that's your song. If you consider a drum machine to be a specialized sequencer, then I think it's got a lot to answer for. What was that album, *Oxygene*? That's a long time ago now, but I found it boring then.

'The secret of using sequencing well is incorporation. There's another track on the album called 'Land of Confusion' where I use a whole sequenced bassline. Originally it was an addition to the song but it ended up being one of the major aspects of it. I find that quite exciting, I must admit.

'I'm not prejudiced against sequencers any more, but I think they're a dangerous tool.'

Strangely, for a musician with classical roots, one thing that's always characterized Banks' use of sounds is distortion. Even the FM trend of clinically-clean synth sounds gets its share of mistreatment this way.

'I've always liked fuzz-boxes. Get an expensive instrument and put it through a fuzz-box and it sounds as cheap as all the others. The advantage of fuzz is that it gives you a limit. A clean sound can get louder and louder and you never reach that point, whereas with something that gets to a distortion point, you know when you're hitting it. You just don't get that excitement with a clean sound.'

As Banks starts to recount the metamor-

phoses his keyboard setup has undergone over the years, he speaks quickly and with a rashness that belies his private-school education. At times he seems afraid there isn't enough time to say all he feels he has to, and his conversation flits from instrument to instrument with disconcerting ease. With a typical disregard for convention, Banks begins his tale with the old Pro Soloist.

'The greatest thing about the Pro Soloist was the touch-sensitivity on it. It was years before anyone else came up with a system that really improved on that. You had to replace the pressure pads every so often because they got compressed, but the fact that you could control the vibrato manually rather than using delayed vibrato was great. If you were playing an oboe part, you could bring it in when you wanted it. The Pro Soloist oboe sound with vibrato and echo would have fooled anybody at the time, I reckon.'

ARP's preset monophonic was followed by an ARP 2600 system and later a Quadra in Banks' ever multiplying array of gear. What was ARP's secret at the time?

'I think that was a matter of recognizing the initials. It's a terribly arbitrary way a lot of people buy instruments – particularly now. I liked the Pro Soloist a lot and then David Hentschel, our producer at the time, had access to a 2500. Then he got a 2600 which I used a couple of times. I found the way of synthesizing from basics quite easy to do, and I learnt about synthesizers using that machine.

'But poor old Quadra. He was great because he was MIDI before there was MIDI. It had four sections though I could only find a use for two – it was a combination of the poly with the lead. I used to like sending the poly through a fuzz-box and then playing lead with it, so you got the aggressiveness of the fuzz with the definition of the lead. On 'Abacab', which was all done using the Quadra, it gives a very positive sound.

'I also had a Prophet 5 which I switched to a Prophet 10 and still use. The 10 had the advantage that you could get big sounds out of it by combining four oscillators at the same time. You can get organ-type sounds. I know it's easier to use an organ, but I was trying to keep the number of instruments on stage to a minimum, and you could get organ sounds as well as big synthesizer sounds out of it.

'I like the Drone setting on it, too, something Sequential didn't put on the T8 for some reason. I used it on things like 'Mama', where you've got a bass note going all the way through. It meant you only had five notes for each chord. You have to be careful to take your hands off all the time, because if you touch a note in Drone the noise is so awful, or if you play too many notes at the same time, suddenly it'll just go mad on you.

'I find that a useful technique on MIDI too. On 'Tonight, Tonight, Tonight' it's all a MIDI thing with a marimba all the way through, but at the same time I was playing the Prophet 10 on Drone so the chord would float over from one part to the next – so you're never quite sure where one stops and another starts. It's so big, though, that's why I'm using the Super Jupiter a bit now.

'I've had a basic Synclavier system for a long time. I bought it instead of the Fairlight originally because it was promised that they would have the sampling section out in a couple of months, and I thought the basic synthesizer part was better than the Fairlight's. As it transpired, it didn't come out till about four years later and when it did you had to mortgage your house in order to put a down-payment on it, so I avoided it.

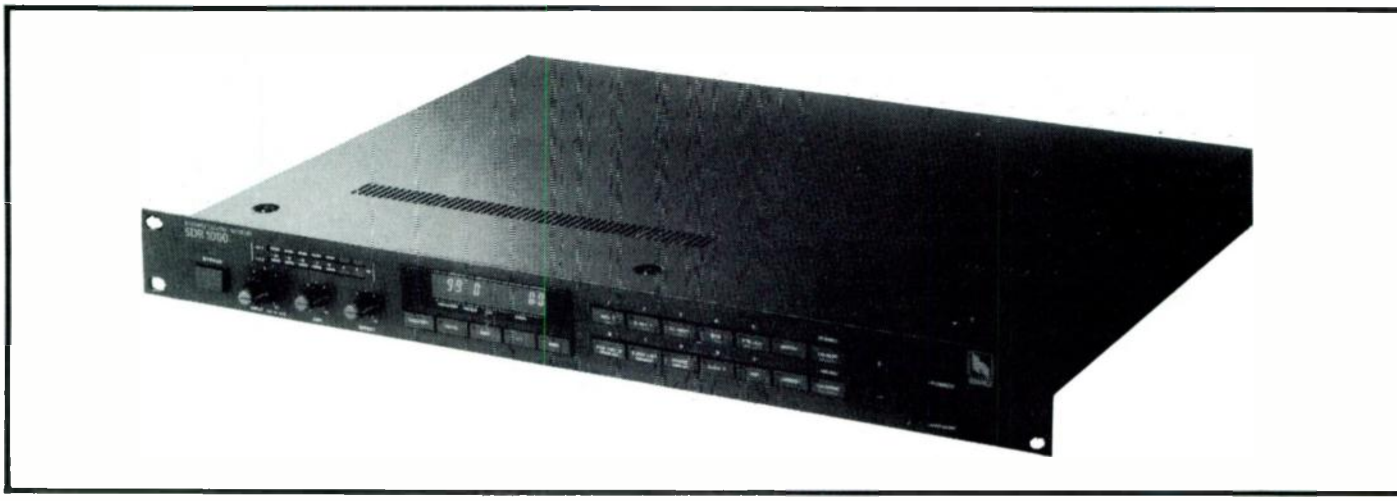
'At the same time E-mu brought out the Emulator I which was nice 'n' cheap by comparison and was pretty good. I had such trouble with the Synclavier – it was about two years before I could really use it properly. There was lots wrong with it and I couldn't get anything done about it. I got extremely angry: it was depressing, having an instrument that represented such an incredible outlay lying there useless. Since then, I've always thought that the more expensive a piece of equipment is, the more likely it is to go wrong.

'The Mellotron was bicycle chains and vacuum cleaners, but it produced a sound that was totally unlike anything else. They'd cornered the market for about ten years if only they'd realized it, and they never made the most of it. That instrument had the potential to be stunning. Yet even the basic sounds weren't good enough, apart from the strings and I suppose the choirs.

'You can still use a Mellotron in such a way that nobody knows what it is. It's very difficult to distinguish half these instruments these days. I think people will come back to using real choirs and I think there's a lot to be said for that, too.'

Banks closes with a considered word of warning: 'Keyboards are fascinating but one mustn't get too much into the technicalities of them. What's important is what you can *do* with these things.' ■

Ibanez SRD1000 Digital Reverb



Does the world really need another digital reverb? Ibanez think it does, and after checking out their new SDR1000, we think they may have a point. *Preview by Paul White.*

REVERB IS ONE OF THE MOST IMPORTANT EFFECTS available today, and it's easy to see why. In real life, nearly everything we hear is reflected sound, so it stands to reason that sound engineers must have full control in this area if they're going to do their job properly.

To help them get that control, musical instrument manufacturers have developed digital reverb systems, which have been flooding the market for some while now.

The Ibanez SDR1000 is a high-quality digital reverberator conforming to the 19" rack-mounting format, and styled along typical Japanese lines. It has a clearly laid-out front panel with an informative alpha-numeric plasma display, and all its functions can be addressed with just a handful of buttons. It's not aimed at the bottom end of the home market like the popular Alesis MIDIverb, for example, but it still costs far less than a Lexicon, AMS, Klark Teknik or Yamaha REV1.

What sets the Ibanez apart from the competition, aside from its sound quality, is the fact that it can operate in true stereo; the channels can be used independently if required, and they don't even have to have the same reverb settings.

The effects the SRD1000 can produce are arranged as 30 factory presets, with the capability to store a further 70 treatments of the user's own devising. These effects are divided into eight modes are: Hall, Room, Plate, Gated Reverb, Reverse Reverb, Dual Delay, Auto Pan and Dual Reverb. Most of these are recognizable as standard reverb effects, but the selection also includes delay programs and an interesting auto-pan facility. Also included within the programmable section is a four-band equalizer which can be applied to any effect, and the user-friendly operating system lets you compare any patch you've just modified with the original before you commit it to memory.

User-variable parameters include Reverb Time, Pre-Delay, Early Reflection Time and Early Reflection Level. Additionaly, you can alter the effect using a Room Size parameter, and in total, there are more than a dozen user variables, so there should be plenty of scope for experimentation.

It almost goes without saying that the SDR1000 has MIDI to enable programs to be selected remotely, but an optional MIDI foot control unit means that this can be accomplished conveniently live as well as in the studio. It's possible to step through the programs using a regular footswitch, too, but the MIDI foot controller sounds more useful to me.

Before going on to enthuse about how good this machine sounds, it's worth taking a brief look at its specification.

The inputs and outputs are on regular phone jacks, but Ibanez has thoughtfully included RCA-type pin jacks as well to give the user a choice. The effect bandwidth is 10kHz – which sounds plenty bright enough in practise – and the 16-bit linear sampling gives a high resolution with little noise or distortion. In fact, the dynamic range is quoted as being greater than 90dB with distortion being under 0.03%, which is mighty impressive.

Subjectively, it's sometimes easier to judge a good reverb by what it doesn't do than by what it does do, because it's all too easy for outboard machines like this to impose their character on the signals you put through them. This one, though, seems to score fairly heavily in all areas. It doesn't impart a metalic ringing to the program input (not even on percussive sounds), and the decay tail is smooth all the way, just as it should be.

The Ibanez isn't at all difficult to program, and the range of reverb and delay treatments it offers is comprehensive enough not to be limiting, even if you have wild production ideas.

Crucially, the SRD1000 seems to have got the early reflection part of the reverb treatment just right. This is the part that simulates those first few echoes that occur in a room before the density builds up to a dense clutter, and as it's these few echoes that pass on information which our brains interpret as room character and so they're hardly unimportant.

From the smallest room to the largest of halls, the Ibanez remains convincing, conveying an impressive sense of stereo perspective and depth. It's also a neat trick to be able to set the pre-delay so that it's slightly different on each channel, as this further enhances the stereo illusion.

As for the delay effects – which include chorus and flanging – and the panner, these are really to be considered as a bonus, and work faultlessly.

I've made a point of listening to most of the digital reverbs currently available and this one compares with the best. There *are* better reverbs, but only at the very top end of what is becoming a very tall (in price terms) market tree. We'll be looking at this machine in depth just as soon as we've had one long enough to assess it thoroughly, but in the meantime, I can say that initial impressions are definitely favorable. ■

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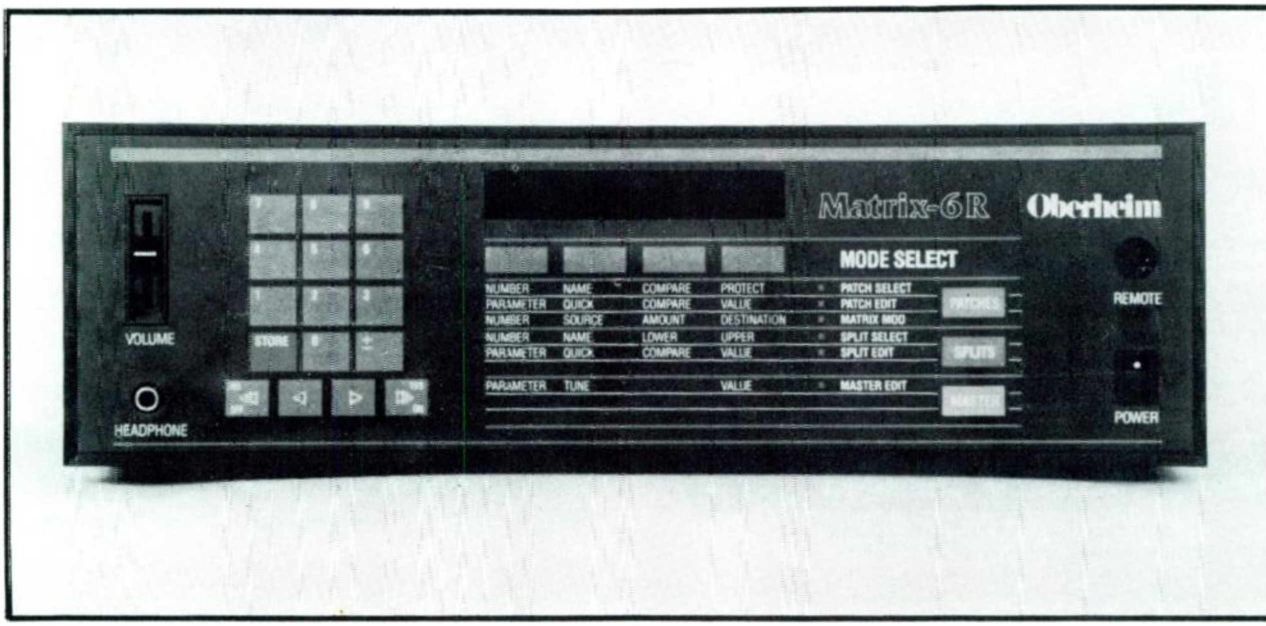
Oberheim Matrix 6R Expander



One of the world's most sophisticated polyphonic synthesizers is now available in a modular, rack-mounting package. *Review by Simon Trask.*

WHILE SOME MANUFACTURERS MAKE A SPECIALITY out of bringing sound synthesis to the masses, Oberheim has been busy turning out instruments which are out of this world – but also out of most musicians' financial grasp. However, the introduction of the six-voice Matrix 6 polysynth saw the company moving into a more affordable price range, though it retained many of the features which make Oberheim's version of the modern synthesizer so special. Now the company has brought out a rack-mounting version of the 6 which retails for less than $1000

For the uninitiated, the Matrix 6 and 6R are cut-down versions of the Matrix 12 and Xpander. What's surprising is how few economies have been made where it matters most: the noise the instrument makes.

Some voice components have been lessened in number (though there were so many in the first place, this may actually be an advantage), and the VCOs of the earlier instruments have been replaced by DCOs. But you've still got 99 voice parameters, 54 matrix modulation parameters, 56 master edit parameters, and eight split edit parameters to mess around with. The Matrix 6 handily lists these on its front panel, but of necessity, they're relegated to the top of the 6R – which is fine for tall people, so long as they don't stick another unit on top of the Oberheim in a rack.

Internally, each voice has two DCOs, a VCF, two VCAs, three five-stage Envelope Generators, one FM Modulator and one Tracking Generator, two Ramp Generators and a Portamento controller.

And if that doesn't leave you breathless, there's Oberheim's software-implemented Matrix Modulation system, which allows up to 20 sources to modulate up to 32 destinations – a total of 640 possible combinations. For each patch, you can select from 18 'hard-wired' modulations and create up to 10 of your own. Modulation sources include the three envelope generators, two LFOs, vibrato, the two ramp generators and the tracking generator. Destinations include DCO pitch and waveshape, the DCO1-DCO2 mix, VCF frequency and resonance, VCA volume and the ADSR parameters, plus overall amplitude of the envelope generators.

Along with internal modulations, it's in the Matrix Modulation section that you define the effects of keyboard dynamics (including release velocity – though few keyboards implement this) and controllers on your voices. Oberheim's approach gives you a degree of performance control over your sounds perhaps unmatched by any other synth. Such flexibility means you have to keep your aural wits about you when programming sounds – which is no bad thing.

The 6R allows you to store 100 single patches and 50 multipatches onboard. The latter are split and dual combinations of two patches which can also be partially overlapped. External storage is to cassette or over MIDI – though as always, you'll need a computer and appropriate software or something like the J L Cooper MIDIdisk to accomplish the latter.

Considering its abundant sonic resources, the Matrix 6 was given a rather uninspiring collection of factory presets when it first came out. Fortunately, the 6R's presets sound as though a lot more care has gone into constructing them. And the multipatch combinations are similarly more usable.

Oberheim has also put plenty of thought into the MIDI aspect of the Matrix 6R – not least in giving it the ability to receive in Mono Mode 4. Apart from the usefulness of this mode in sequencing, it also has an essential application in guitar synthesis – and with the increasing number of MIDI guitar systems now appearing, that's a powerful reason for its inclusion on voice expanders like the 6R.

The rack Matrix also has a 'patch mapping' facility, which allows you to assign patches to incoming patch numbers, removing the problem of aligning patches from different MIDI instruments which count their memories in different ways. You can also select another patch number to be sent by the 6R on its MIDI Out to a third instrument – a handy feature. The 6R's MIDI Out can act as a 'mix' output, combining data from your master instrument with MIDI data generated by the 6R. Along with patch changes, this can include controller information – the 6R has inputs for a footswitch and a footpedal, each of which can be set to any MIDI controller code.

The end result is a powerful and responsive instrument which sacrifices little in terms of voice quality over its more expensive predecessors. Well worth checking out. ■

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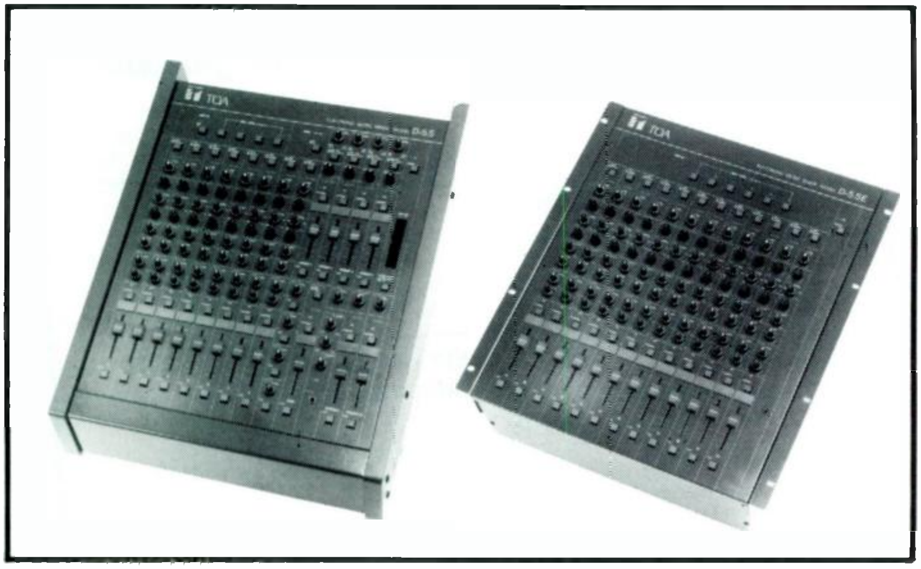
I·N B·R·I·E·F

Toa D-5.5 Electronic Music Mixer

Flexibility and performance are two of the strongest features of this desk which deals not only with audio signals but with MIDI too. *By Rick Davies.*



IT SEEMS THAT WITH THE ONSLAUGHT OF MIDI and the number of affordable multiple-synthesizer setups it has made possible, a lot of electronic music makers are now having to look for something to mix all those synths. And typically, as soon as they find a mixer which meets one set of requirements, their desire for something that can do a little more leads, soon enough, to some mighty expensive boards.

Having spent a life's savings on the latest sampler or signal processing gear, it often becomes a case of double or nothing, so to speak. It can be difficult, selecting a mixer which has the right combination of features for producing electronic music. That's precisely where Toa's D-5.5 mixer comes into the picture.

The D-5.5 is a sophisticated eight-input mixer which can be expanded (in units of 12) to 32 inputs with Toa's model D-5.5E (which connect via multi-pin cables). Both models are designed for console use or mounting in standard 19" racks. The D-5.5 provides four group, two stereo, one mono sum, and two cue outputs – making this board flexible enough for many recording or live applications. In addition to this, the D-5.5 features reverb, effects, and two auxiliary busses with extensive routing capabilities, and even features a simple two-in, four-out MIDI switcher with pushbutton controls on the front panel

Each channel features a source/tape switch and Pad/Trim controls for tailoring the incoming audio signal for optimum signal-to-noise performance. Channel EQ consists of high and low shelving, and mid-range peak with variable center frequency controls. Sends to the four busses (reverb, effects, and two aux) are controlled by two sets of concentric pots, the innermost setting the level of the send signal, the outermost setting the balance between the two aux or reverb and effects busses. The four send levels are controlled from two sets of concentric pots in the output section of the mixer. Unless the Reverb Send output is used, the reverb send signal goes to the D-5.5's internal spring reverb, which has high and low EQ controls.

Two similarly-configured controls are employed to route the input to any of the four mix groups (more on this later). Each input's channel switch and slider (with corresponding LED 'signal present'/clipping indicator) determine the signal's initial level, while a Cue switch allows monitoring any post-EQ, pre-fader input signal. Certainly not features typical of most eight-channel mixers.

XLR, ¼" phone, and RCA (for tape and direct out) jacks are provided on each input, as well as pre- and post-EQ RCA patch points for inserting effects into the signal path. Above each slider is a writing block for labeling the inputs with grease pencil. The features mentioned thus far are common to the D-5.5E.

In the output section are the four mix group controls. Each group has a selection switch for receiving either the reverb/effects returns, or the two aux returns. If having to select between these two return groups seems limiting, take comfort in knowing that the reverb and effects sends can be mixed into either or both aux send signals. Concentric knobs set the level and balance between the reverb/effects or aux1/aux2 returns to each group. Both ¼" phone and RCA jacks are provided for each group output, catering to various applications. Each group also has a pan-pot for positioning in the final stereo mix.

The stereo mixing section features similar controls to the group controls: Cue and Channel (on/off) switches, a slider, and writing block for the left and right outputs. Even the four return busses feature Cue switches in addition to individual concentric level and pan knobs. Great attention to detail all the way around this area.

The monophonic summed output features the same controls as the stereo output controls, with an added pre/post-left/right slider switch. Concentric knobs are used to adjust the cue and headphone output levels. A dual bar-graph LED display follows the left and right, or sum and cue output levels, depending on the position of the Meter Select switch.

Then there's the MIDI section. Two MIDI inputs may be routed to any combination of four outputs, but unlike many stand-alone MIDI switchers, this one allows only one of the two MIDI inputs to drive the outputs at any given time. Still, the inclusion of those six DIN jacks is certainly not going to offend anyone. Adding the D-5.5E adds six more MIDI Outs to the switcher, so if two MIDI chains need to operate separately, an additional switcher may be necessary.

Overall, it appears that Toa have done far more than simply cover all bases. The routing capabilities of this system are such that the board will fit comfortably into many environments, and the back panel jacks are sure to meet most realistic requirements. Flexibility and attention to detail at all levels of operation are definitely the D-5.5's strong points. Retailing for $1999.50 ($2129.50 for the 12-channel D-5.5E) the D-5.5 may not appear to be terribly cheap, but then, the D-5.5 is not simply another eight-channel mixer. ■

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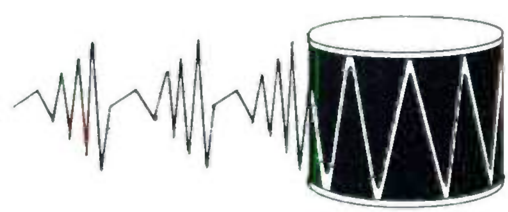

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ENSONIQ ESQ1

REVIEW

Photos: Matthew Vosburgh.

Digital Synthesizer/Sequencer

It's little more than a year since the Mirage first made the authenticity of sampled sounds available to the musician on a limited budget. Now Ensoniq has combined a sequencer and a digital synthesizer into one versatile package and priced it even more competitively. *By Paul Wiffen.*

IT CERTAINLY LOOKS LIKE ENSONIQ plan to stay one jump ahead of the rest of the market. When everybody was looking to the established manufacturers to come out with a sampling keyboard within range of the average musician, they came out of nowhere with the Mirage. And now that all the other major names have finally gotten their samplers out, the guys from Pennsylvania are already moving into other areas. Specifically, their latest machines cater for both the traditional keyboard player (with the weighted keyboard and sampled accuracy of the Ensoniq Piano) and the thoroughly modern synthesist via the instrument under scrutiny here.

The ESQ1 is an eight-voice digital synthesizer which also encompasses a versatile multitrack sequencer able to sequence not only internal voices, but also a considerable number of external devices via MIDI. In fact, so complete is its MIDI implementation, you could use the ESQ1 as a master keyboard, making it the center of a complete MIDI system.

The Synthesizer

The first thing in any player's mind when buying a synthesizer will always be the sound of the instrument, so let's take a look at the ESQ1's voicing to start with.

Each of the eight voices has no less than three oscillators – more than any other synth under $2000.

Waveforms for the ESQ1 are drawn from a wider range than previously possible on a conventional synthesizer. Because in addition to the conventional waveforms bequeathed to us by the analog synth (ramp, triangle, square, pulse), there are waveforms created by digital calculation to contain certain specific harmonic series. And more excitingly, there are waveforms which have samples of real sounds as their source. Not a new idea, this, but on the ESQ1, the waveforms are derived from multi-samples at the point where the sample is brightest, so you have the maximum harmonic content that appears in the sample. This section is then looped so it'll sustain indefinitely, as waveforms normally do on conventional synths.

Now, one of the problems musicians and programmers experience with samplers is that you can't sustain a sample indefinitely unless you manage to get a good loop on it. With the ESQ1, these problems disappear, as Ensoniq has already looped the machine's multi-sample waveforms for you.

All you need to do then is adjust the filter and amplifiers to imitate the changes in loudness and brightness that characterize the original instrument. Even here, Ensoniq has done some of the work for you. The ESQ1 has a waveform entitled 'Piano', plus a patch of the same name which uses that waveform filtered and enveloped in such a way as to recreate the timbre of a piano. Using a footpedal, you can sustain this sound exactly as you would with a conventional piano. So, what we have is the realism of a sampled piano sound with the immediacy of a synth preset (no waiting for disks to load here). What's more, the multi-sampled piano waveform is still available to be used singly, doubled, or in conjunction with up to two other waveforms in the other patches on the machine.

Intriguing though this sampling-meets-synthesis concept is, it shouldn't detract from those ESQ1 waveforms which are entirely synthetic, rather than taken from acoustic sources. Because as it turns out, many of the more exciting patches are made entirely from these waveforms.

Ensoniq divide their waveforms into five categories. First come 'Classic Synth Waveforms', which include sawtooth, sine, square, pulse, and three noise waveforms. Second are the sampled waveforms such as bass, piano, reed, organ, two voices and kick. The third section is created by additive synthesis, and the manual actually tells you which harmonics are present in each of the three waveforms in this group. Formant waveforms come next. These are multi-sampled with a resonant peak which stays around the same frequency (given in the manual) wherever you play on the keyboard. Last come the sound limited waveforms, which are versions of some of the other waveforms with limited bandwidth.

Personal Wiffen favorites are 'Pulse' among the analog (though you can't modulate the pulse width), 'Bass' among the samples, which has a sharp, realistic top end while retaining plenty of power down low, and *all* the Formants (I couldn't decide which one of these five was best for vocal sounds, they all sounded good). The band-limited patches are great for beefing up the lower end of sounds, while those created are additive synthesis are certainly unusual, and especially interesting played through a slowly closing filter.

BUT THE BEST THING ABOUT THE ESQ1 is the scope it gives you for experimenting with different combinations of waveforms from different origins. After a few minutes twiddling, you can discover that, for example, an analog waveform can make the piano waveform sound more authentic, or that a sampled bass waveform can be the basis for a great synth sound. Fascinating stuff!

Anyhow, once you've decided on your combination of analog, digital and sampled waveforms for each of the Ensoniq's three oscillators, you can decide how you want to process them.

Before getting into the filters and envelopes, there are several modulation facilities available. These include a feature I always figured had faded away with the last of the great analog synths – Sync. For those of you too young to remember the Prophet 5 or Moog Prodigy, Sync is a means of forcing a synth oscillator to restart its cycle (ie. the oscillation that produces the pitch and harmonic content) each time the controlling oscillator begins a new cycle.

This can produce some excellent beefy, distorted sounds, especially if the oscillator being controlled is shifted in pitch by an envelope or glide function as well.

On the ESQ1, the ability to use this facility with multi-sampled waveforms (which have never been 'Syncable' before) kept me occupied for some considerable while, and gave rise to some great, distinctive sounds.

Another interesting modulation option is Amplitude Modulation. This doesn't use the same process as Yamaha's FM, but the sideband frequencies Ensoniq's process produces are similar, and the sounds that result have much in common with the electric piano/bell-type sounds that the DX range is so good at generating. And as you're starting with more complex waveforms than you would on a DX, you build up more complex sounds very quickly.

Once you've finished messing around with the harmonic content of the oscillators, you can set a different loudness envelope for each one, if that's what you want. The ESQ1 actually gives you control of the relative mix in real-time, so you can start with, say, just the first oscillator with sharp attack, bring the second in more slowly, and then just as both are dying away, bring up the third oscillator to replace them. The envelopes available to you are more versatile than standard ADSRs, too, so there's plenty of flexibility in the way you combine the different sounds emanating from the oscillators. Imagine, for example, a brassy attack (from a ramp wave) becoming a sharp digital sustain fading into a piano decay... And all this timbre change possible before you even start to use the filter and its associated envelope.

The filter itself (one for each voice) is the classic four-pole device which has remained little changed from the days of the earliest analog synths. Besides the standard Cutoff Frequency and Q-controls, you have Keyboard Tracking and Envelope Amount.

The filter can be controlled by the same versatile type of envelope mentioned above – the sort of envelope which really deserves something of a closer look. Briefly, the ESQ1's envelopes use the Time and Level parameters which are becoming more and more common these days, as they allow complex envelope shapes (in addition to the more standard ADSR shapes) to be created. Each phase of the envelope is programmed by setting a Level, and then a Time which the envelope takes to reach that level.

In ADSR terms, level 1 would be the peak and Level 3 the sustain level. As the envelope has to return to zero to finish, Level 4 must always be zero. The four Time parameters could be roughly described as Attack (Time 1), Initial Delay (Time 2), Second Delay (Time 3), and Release (Time 4). However, it's possible for these parameters to be set in such a way that they go beyond these descriptions. If you set Level 3 above Level 2, Time 3 becomes more like a Second Attack than a Second Delay.

Add to this the fact that all values can be negative as well as positive (useful for controlling the filter, for example), and you begin to see the potential of this system.

All envelopes can also be affected by velocity, which can control not only their peak level (and all other levels in proportion) but also the attack time (ie. how much Time 1 varies from the programmable amount). In addition to this, a Keyboard Delay Scaling feature allows you to shorten the overall time the envelope lasts as you go further up the keyboard. This imitates the nature of plucked or hammered acoustic keyboard sounds, which tend to be of shorter duration the higher in pitch they are.

There are actually four envelopes per voice on the ESQ1. The first three are each assigned to their respective oscillators, and the fourth to the overall output level. The filter can be controlled by any of the four envelopes, but sadly, it has no separate envelope of its own.

INCLUDING THE FOUR ENVELOPES, THERE ARE 15 possible modulation sources on the ESQ1. These include three LFOs, various velocity and keyboard tracking curves, plus wheels, pedal or any MIDI controller (including pressure). These 15 sources are always available to modify the pitch and level of the oscillators, the filter cutoff or the stereo pan position.

Those modulation sources include the most versatile LFOs I've ever seen, with two levels of effect and delay (rise time), plus the ability for the LFO to be further modulated by another source – even itself, if you're feeling experimental. And in case all this flexibility leaves you feeling a little confused, the manual shows you how to create standard modulations like wheel control of vibrato.

Staying with the manual, it's worth pointing out that unlike the documentation that came with initial Mirages, the manual for the ESQ1 covers all aspects of the machine in great detail, with some fine explanatory sections just in case anything isn't immediately obvious. In this respect, the latest Ensoniq manual is more like the Advanced Sampling Guide for the Mirage, but for the impatient, it also contains two brief sections titled 'Getting Started' and 'Getting at the Sounds'.

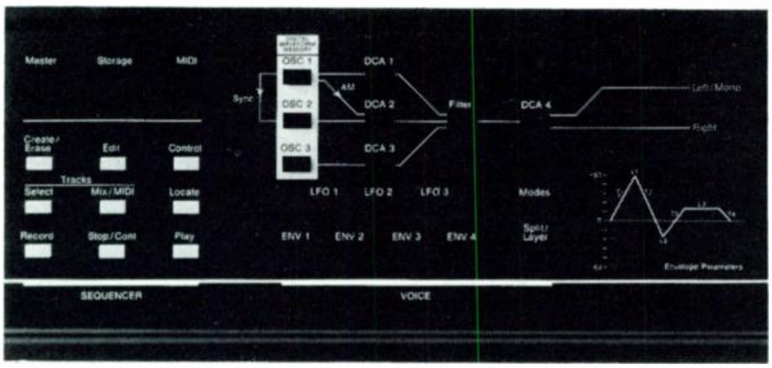
But the manual is something you may not need to look at for a while after you've bought your ESQ1. Because unlike so many modern synths, the new Ensoniq has a programming layout that's so easy to get to grips with, it almost *invites* you to delve deeper.

Whole groups of parameters – refered to as Pages – are called up into a large 80-character LED display, and can then be quickly accessed by pressing the closest button. The values then shown can be changed using either a slider or up/down buttons – depending on whether you want drastic change or fine control. So each page allows easy access between the related ▶

parameters, and saves darting about between individual parameters 'blind'. It's not as useful as good ol' knobs and switches, but it's a step in the right direction.

The Ensoniq's display is one reason for the machine's ease of use, mostly because of the sheer *amount* of information it can convey at any one time. One display, for example, caters for an entire Page-worth of standard MIDI features such as Channel Assignment, Mode, Controller, and Program Change Enable, plus additional features such as Overflow Mode and Multi-Mode. Both of the latter are excellent facilities, but neither is quite the innovation Ensoniq claim. Overflow Mode (where any extra notes played on the ESQ1's keyboard or sequencer the internal voicing can't cope with are 're-quested' via MIDI so you can hook up a second ESQ1 and have a 16-voice system) has been available on the Prophet 2000 for nine months now, while Multi-Mode is similar to the way the Casio CZ5000 sequencer assigns various MIDI channels to its internal sounds and external slave synths. Still, it's good to see companies like Ensoniq continuously looking for ways to increase the flexibility of MIDI, rather than just implementing the basics and leaving it at that.

Another Page worth its weight in gold is Split/Layer. This allows you to save within the program the fact that two patches are used, which can then be either doubled to create a layered effect or arranged so you have two different patches on each side of the assigned split point.



THE ESQ1 HOLDS 40 DIFFERENT PROGRAMS internally, arranged in four separate banks of 10 each. Ten named programs can be scanned in the display at any one time, and each can be chosen in the same way parameters are selected when programming. Another 80 programs can be accessed (in two sets of 40) on the E^2PROM Cartridge, and these can be similarly displayed in banks of 10 before selection. So for a live performance work, you've got instant access to 120 programs – if a cartridge is in the slot.

The Sequencer

You can use the ESQ1's sequencer in two different ways. The first option allows you to sequence just internal voices multi-timbrally. In other words, each track can have a different sound on it, so entire pieces of music with up to eight parts can be recorded. The second option allows you to sequence eight external synthesizers to the extent of their polyphonic capability, by assigning each of them to a different MIDI channel.

As a third option, though, you can actually use both alternatives in conjunction with each other, using both internal and external synth voices. For example, you could make tracks 1 and 2 play just internal programs, track 3 an external synth, 4 and 5 doubled on both the ESQ1 and other keyboards, and so on. You're limited to eight voices on the ESQ1 itself, but because of the fine dynamic allocation implemented on the machine, those eight voices are always available to play any sound, so provided they're not being used at that instant anywhere else, you can have up to eight notes on each internal track.

Because of its portability, it's conceivable many musicians will use the ESQ1 as a songwriting tool, and then augment it with other synths in the recording studio to put their original performance on tape, while retaining complete freedom over the sounds they are using. Luckily, the sequencer's assignment potential is wide enough to make this possible.

The sequencer can store up to ten songs, each identified by a name, and these can be made up from 30 sequences. Now, you may find, once you start to put your entire live set into the machine, that the Ensoniq's internal memory of 2400 notes doesn't get you past the third song. Don't worry. A cheap cartridge is easily inserted (and held in place by screws) to expand this capacity to over 10,000 notes...

Operating the ESQ1'S sequencer is simple enough. All you do is enter the track you want while the required sound is currently selected. This automatically assigns that sound to the track, and you're then free to record your part. If you select a cartridge sound, its name is displayed in the track space until you remove the cartridge, when it'll tell you that you need to re-insert the cartridge before that track will play. Alternatively, you can just go to the MIDI/Mix page and assign that track to both Record and Playback on a particular MIDI channel, simply by entering the number so you can perform that part on the keyboard assigned to that channel.

Once a track is recorded, you can edit it in several ways. First there are standard MIDI sequencer functions like Transposition and Quantization (often known as Auto Correct). The Quantization is particularly good, allowing you to select a resolution between quarter-notes and 32nd-note triplets, listen to the corrected part, and then decide if you want to keep the new version or the original. This way of quantizing 'after the event' rather than during recording is definitely worthwhile, as it allows you to keep the human element in a performance without having to put up with human fallibility (which I specialize in).

Step-time recording – so often neglected by US manufacturers – is also available on the ESQ1's sequencer. This method allows you to be analytical in your composition, and also to program things you find tricky (or just plain impossible) to play. The step size can be from quarter-notes to 32nd-note triplets, as with the quantization, and the readout shows the bar number, the beat number and the clock number you're currently on, so you don't get lost.

If you prefer a more tape-like way of recording difficult sections and correcting mistakes, the sequencer also features Punch In and Punch Out facilities, enabling you to 'drop-in' small sections in your own good time.

There are some more advanced editing functions available, too. For example, you can use Remove Controllers to get rid of any unwanted MIDI controller data. This is particularly useful in conjunction with something like a DX7, which sends out pressure data whether you're using it or not. This would normally use up even the 32K of expanded sequencer memory ridiculously quickly.

Other useful edit functions include Merging Tracks (what we'd call 'bouncing together' in the recording studio) and Copying Tracks, which allow you to move recorded parts around and arrange them in the best format. To build sequences into longer sections (thereby freeing sequence locations), you can use Append, which tacks one sequence onto the end of another. Alternately, you can extend or truncate a sequence to make room for extra bars, or lose some that are less than perfect. And the MIDI/Mix page allows you to adjust the relative levels of your tracks.

When you have your sequences together, you can start to build them up into songs. Each step of the song can be accessed in the Song Edit page, and each one can be Transposed, Repeated, or Deleted. You can insert extra sequences at any time, and you can move backward and forward within the song to facilitate this.

You can also go straight to any point in a song using the Song Locate page, and see what tempo and time signature have been used. The 'Goto' parameter jumps automatically to the step you want.

Synchronization to the outside world is possible via a variety of options. There's tape sync for the recording studio, and MIDI for drum machines and other relative MIDI devices or even SMPTE-to-MIDI syncing. The last facility is particularly useful, as the ESQ1 both sends and receives song position pointers via MIDI. Using the autolocate controls, you can start playback of all MIDI devices with song pointers implemented from any point in the song. Even more importantly, the ESQ1 (in conjunction with a SMPTE-to-MIDI converter like the Roland

SBX80 or Fostex 4050) can be made to autolocate alongside a tape machine to start playback automatically at the same point as the tape. The great advantage of this is that parts recorded on the ESQ1's sequencer needn't be recorded onto multitrack, but played back from the sequencer into the final mix. This frees tracks on tape for other instruments, vocals, and so on which can't be sequenced.

The Master Keyboard

The sequencer can also serve a different role. By using the fact that it'll store patch numbers and transmit them via MIDI *even if a sequence has not been recorded*, you can hit a sequence number and have up to eight different MIDI patch changes sent to different synths and other MIDI devices like signal processors, setting them up for the next song even if you don't intend to sequence them. Or you could record these program changes as the end of a song in readiness for the next piece. And why not have the ESQ1 change programs for you on the keyboards you're playing, or while the others are being sequenced?

This facility, together with the flexible MIDI implementation we looked at earlier, means that the ESQ1 can also function as the central controller of your entire MIDI system, especially if you're on the sort of budget where a master MIDI keyboard looks expensive for something that makes no noise itself.

If you own a DX7, you might do your playing and sequence recording from that, to take advantage of the key pressure which the ESQ1 can record and playback – even though it can't generate pressure itself.

If you own a Mirage, you can save and load ESQ1 programs and sequences to Mirage disks via MIDI. This is faster and easier than the tape storage procedure that the ESQ1 also provides, but a bigger advantage with Mirage saving is that you can keep all your ESQ1 programs, sequences and MIDI controller information on one disk.

All in all, it's difficult for me to fault the ESQ1. Yes, I wish they'd put either a MIDI Thru socket on the back, or at least the switch between Out and Thru that the Mirage has, just in case you don't want to use the ESQ1 as your MIDI instrument. And yes, an entirely separate envelope for the filter would have been nice.

But these criticisms seem trivial stacked alongside the ESQ1's good points.

As a synthesizer, it's able to recreate a wide range of 'standard' synth sounds, and create new ones by the innovative way it combines analog, digital and sampled waveform. Its voice processing is nothing if not comprehensive, and its programming system is one of the modern synth industry's most friendly.

As a sequencer, it's encouragingly quick and easy to use, and most important, it doesn't do anything which could destroy unrepeatable sequences.

And as the center of a modern electronic music system – live or in the studio – it has a MIDI implementation so flexible that, whether you're playing your other MIDI instruments directly or sequencing them, it's difficult to imagine a setup that the ESQ1 couldn't cope with.

If this was the latest instrument from an established synthesizer company, it would be worthy of high praise. But as the first synth from a company that hasn't been in existence for more than a couple of years, the ESQ1 is a revelation, and a hugely heartening one at that. Really, an outstanding bargain. ■

DATAFILE

Ensoniq ESQ1 Digital Synthesizer/Sequencer

Keyboard *61 note C-to-C, velocity-sensitive*
Sound Source *3 oscillators per voice using analog, digital and sampled waveforms*
Voicing *8-voice polyphonic, Single or Split/Layer modes; monophonic option*
Memory *40 RAM memories for user-programmed sounds, shipped containing factory presets. 2400 notes (8K) of sequencer storage*
External Storage *E^2PROM Cartridge holds 80 programs in two master banks, VCP Voice Cartridge holds 80 factory-programmed sounds, SQX10 Sequencer Expander cartridge expands memory to over 10,000 notes (32K)*
Display *80-character fluorescent with 10 'zones'*
Interfacing *Left (Mono) and Right stereo Audio Outs, CV/Pedal In for SW1 optional footswitch, Tape Storage In and Out, MIDI In and Out*
PRICE *$1395*
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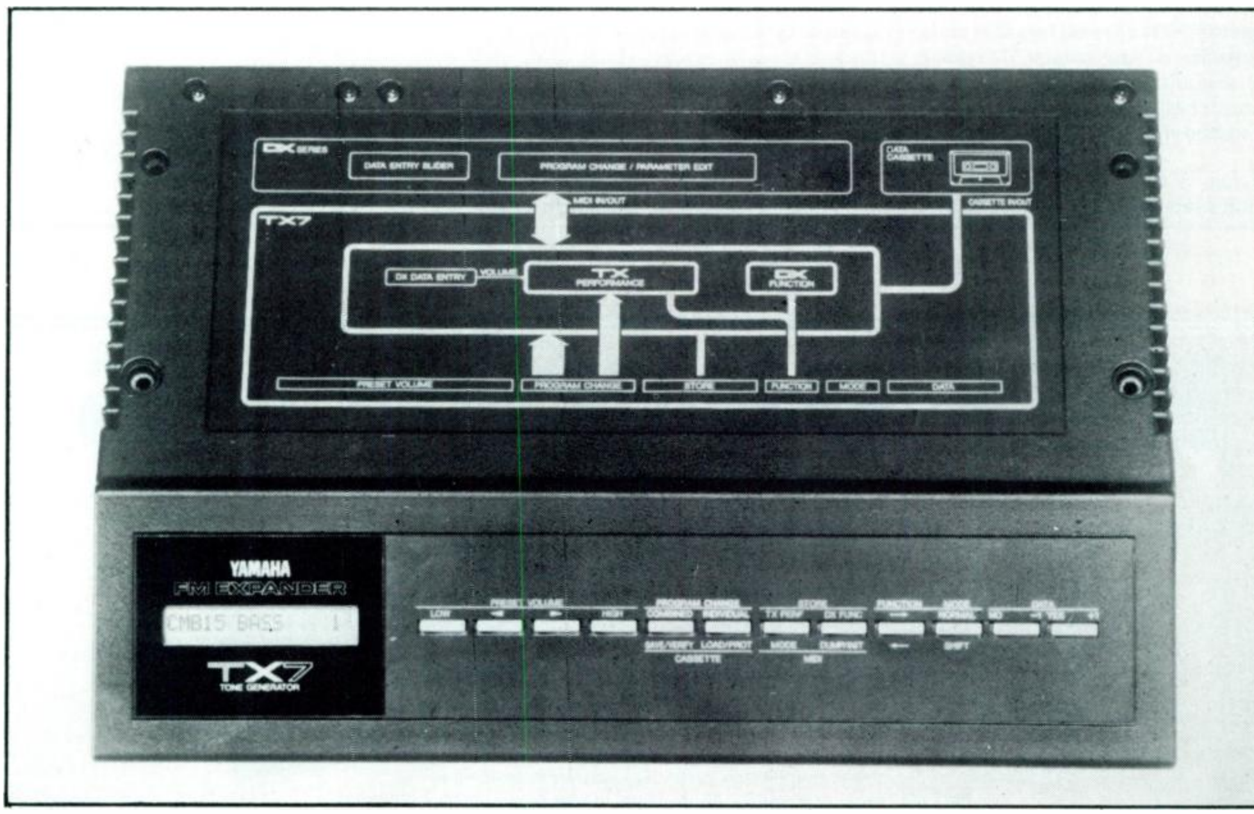
The DX Explained



After our look at the theory and simple practise of frequency modulation in our first issue, we now come to programming simple voices from scratch. *By Jay Chapman.*

THE TIME HAS COME TO GO OUT ON OUR OWN, and begin creating original FM sounds. Both the following voices have been kept simple so that we can build on what we talked about in the first article in this series. While these voices were created on the DX7 (which of course translate directly onto the voice parameters of the DX5 and DX1), the owners of DX9s, DX21s, DX27s and DX100s will be pleased to hear that they will be able to program the same voices on their instruments by using the instructions in brackets (where they differ from the DX7 – if no alternate instructions are given then the parameters are the same as on the DX7).

Only two Operators are used to create the voices. This will of course mean that they are fairly simple sounds. More complex programming will follow in later articles – it's best to learn to walk before you can run...

The first thing you have to do is initialize a voice. Press the brown FUNCTION keypad then the green VOICE INIT (INIT VOICE) keypad and finally the green YES keypad twice. You now have the DX in about as basic a state as it can be.

Before we actually start to enter any parameters on the DX, let's have a look at ▶

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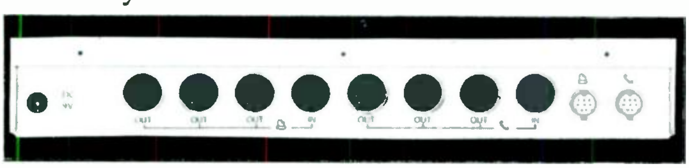

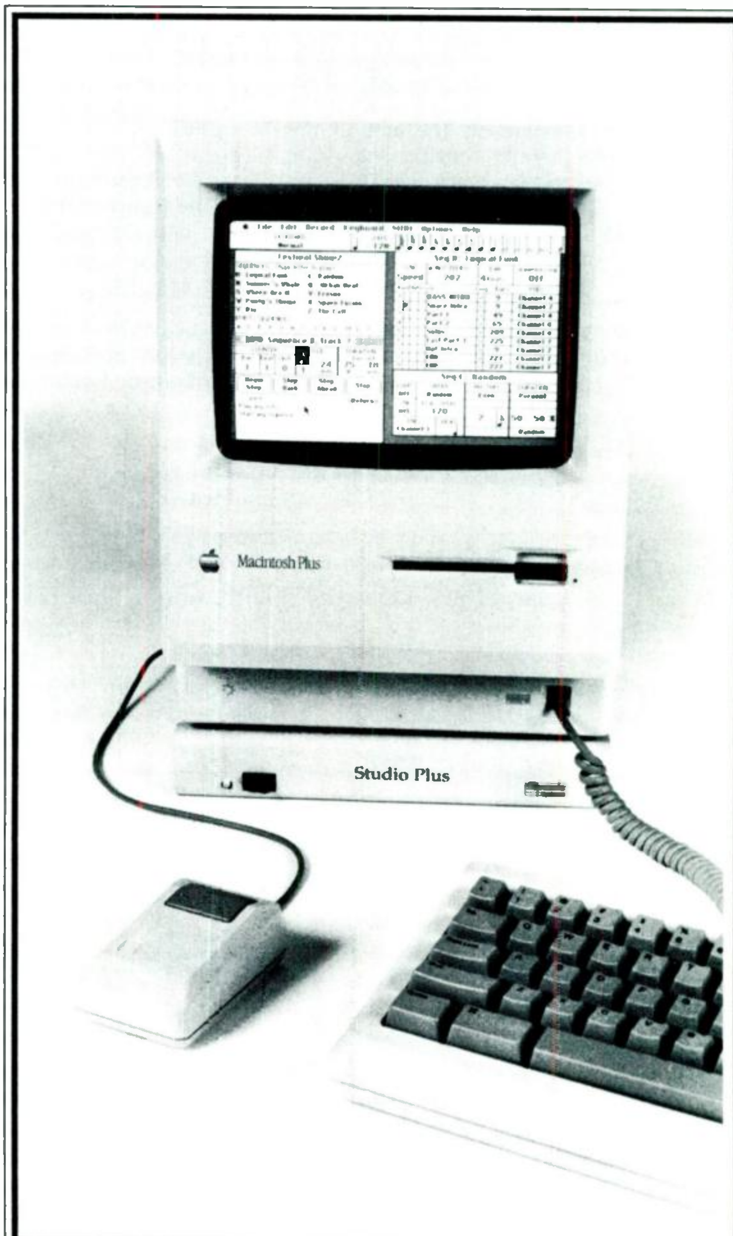

Sequencer 2.5

Did you ever move text or pictures or spreadsheet data from one Macintosh program to another but wonder why you couldn't do the same thing with your music? No problem! In the latest version of our popular Sequencer, you can print out your music files using Electronic Arts' *Deluxe Music Construction Set 2.0* or Mark of the Unicorn's *Professional Composer*. **Sequencer 2.5** also has MIDI files so you can move your sequence data from one program to another. Watch for many other programs to support MIDI Files in the near future.

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what we are going to attempt. Let's start by imitating the basic configuration of a cheap analog synth. Figure 1 shows the state of affairs that we want to imitate – some of you may even be able to work out which cheap synth we're thinking of. The idea of this patch is that a harmonically rich waveform for the oscillator (a sawtooth, perhaps) is fed through a Voltage Controlled Filter (VCF) and a Voltage Controlled Amplifier (VCA). The ADSR-type Envelope Generator controlling the VCA would be set to give a long release time.

The essential character of the voice is given by setting the AD Envelope Generator controlling the VCF frequency to give moderately slow attack and decay times. This opens and then closes the filter, and provided the resonance control effect is set suitably you can obtain a 'wah' effect as the higher frequencies come in and then fade away. I'm fairly confident you will have heard sounds very like this one on quite a few keyboard albums in the past.

The first component we need to create is the 'harmonically-rich waveform'. Using the 'modulator/carrier' configuration we saw in the first issue we can produce complex waveforms by playing about with the modulating Operator's frequency and output level. This is done in the following way. Continue on from the VOICE INIT by turning off Operators 3 through 6 (3 and 4) and then press the green OUTPUT LEVEL keypad and select Operator 2 if it is not already selected. Your display should look like:

```
ALG 1 110000 OP2
OUTPUT LEVEL = 0
(e1100 OUT=17 OP1)
```

We are now in a position to alter the modulation. The difficulty here is that it's preferable to alter both the output level and the frequency of the modulating Operator at the same time, since it's the interaction of these two parameters that determines the overall modulation effect. As we have only a one-dimensional data entry control rather than something like a joystick (which is 'two-dimensional' and can therefore handle two parameters at once) we have to keep stepping backwards and forwards between the output level and the various FREQUENCY keypads (stick to FREQUENCY COARSE for the moment), changing the parameter values a little at a time and listening to the results.

If for example, you set Operator 2's FREQUENCY COARSE to 4.00 and its output level to 75, the result sounds like the sort of thing we are looking for. But to get an actual reproduction of the output of a sawtooth waveform takes a little more experimentation, so we will give you a bit of help. If you set up your synth as shown in Figure 2, it will give a 'sawtooth' output wave. You could select algorithm 9, for example (algorithm 5 on DX9 and above), and set the parameters for Operators 1 and 2 (for the higher numbered DXs, turn operators 1 and 2 off and use 3 and 4 instead) as specified in Figure 2. At least then you'll know exactly where you're starting from..

Figure 3 shows the connections made for the sawtooth wave (ignore the pitch 'EG' part which relates to the 'Syndrum' voice to be described later). Note the feedback loop from the output of Operator 2 (Operator 4) going back to its input. The feedback parameter controls how much of the output is fed back to the input and thus acts in much the same way as the output level parameter of a modulating Operator. Of course, the more output you feed back the more complex the resulting output becomes, which means that the fed back signal is also made more complex, which in turn means that the self-modulation is more complex, which again means that the output is more complex, which ... and so it goes on. Of course, that's the joy of feedback, it's self-perpetuating; ask any electric guitarist. We'll consider this subject in more detail in a later article: in the meantime you might well be able to see why such feedback is used to create a noise source on the DX (consider the definition of white noise; equal level at all frequencies).

So far all we have is the equivalent of a

fig. 1
The 'cheap synth' sound
on the cheap synth.

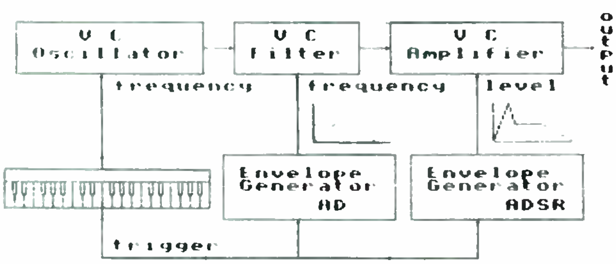
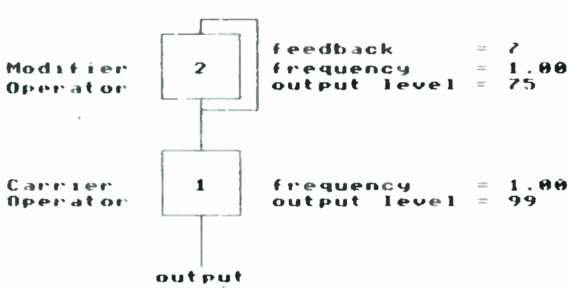
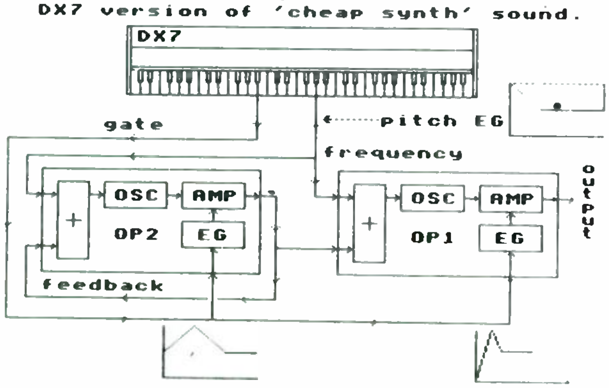


fig. 2
Sawtooth Wave



(Taken from Dave Bristow's 'Getting Started' booklet from Yamaha.)

fig. 3
DX7 version of 'cheap synth' sound.



VCO producing a sawtooth waveform. The 'wah' effect – due to the VCF and its AD EG shown in Figure 1 – can be produced by setting the parameters of the EG in the modulating Operator, to give the required slowish attack/decay curve. It's important to realise that the envelope should not start from and descend to a level of zero as it does in the AD envelope in Figure 1. If it did we would only have the more complex harmonic content of the sawtooth during the 'wah' and we would be left with the simple unmodulated sine wave of the carrier operator for the rest of the time!

To obtain an envelope for the modulating Operator (number 2) as shown in Figure 3, try the following parameters for Operator 2's envelope generator:

Operator 2	1	2	3	4
RATE	40	40	99	99
LEVEL	99	60	60	60

I'll leave you to set up your own amplitude EG ADSR curve, which will mean you deciding on and setting up the parameters for Operator 1's EG, of course.

While the voice we've just produced is not going to win any prizes (understatement of the year!) it shows a very important principle of FM synthesis in action. It's not too difficult to see that with several Operators connected in a more involved algorithm with various modulations (possibly of other modulations!) going on, we can produce extremely complex timbres. However, of far greater importance is the timbral movement we have created in programming the 'wah' in this 'analog synth' voice.

Why?

Because at each point of modulation in our more complex algorithm we could set about forcing such timbral moment. Also worth considering is the fact that at each such point we can apply not just the simple AD envelope, but rather the multi-parameter envelopes available for *each* Operator on the DX series. This is one of the facilities that allows DX synthesizers to imitate acoustic instruments so well, since the change in the relative levels of fundamental and several

other harmonics during the sounding of a note can be synthesized accurately.

The second sound we shall attempt is that irritating syndrum sound indispensible on disco records back in 1980. Again, the simple approach – with just two Operators being used – means that the quality of the imitation is fairly poor. Things will improve in later articles, rest assured.

Once again, the two Operators are used in the Modulator/carrier configuration, as in our first example. The sound we want here borders on the metallic, and this can be achieved by using the same method as that described for the analog synth voice. This time feedback is not required. The values used for Operator 2's (or Operator 4's on the cheaper DXs) frequency coarse and ouptut level are 2.00 and 70 respectively. Operator 1 (Operator 3) is already set up sensibly after the VOICE INIT, of course.

To get the characteristic pitch rise or fall associated with a syndrum the pitch envelope generator is brought into use. The pitch information fed to the Operators is a combination of the keyboard pitch and the pitch EG envelope (this combination takes place logically at the point indicated by the dotted arrow in Figure 3). As you can see, the pitch envelope starts high and settles to a medium value (actually 50): this will give a falling pitch.

When the pitch EG is not in use, all of its ▶

level parameters are set to 50, which corresponds to the keyboard playing at the pitch it thinks it is turned to – if you take my meaning. The pitch EG has the same parameters as the EGs in the Operators.

Since we want to start high we must set level 4 high. Note that the envelope starts *and* finishes at level 4 – the pitch EG diagram in Figure 3 shows the return to level 4, represented by the dotted line across the top of the envelope diagram at its extreme right. We don't actually hear this pitch change because it occurs only after key release, when the carrier's (amplitude) EG curve is at zero anyway.

The fall in pitch must take some time so we need to set up a suitable rate parameter. Since the pitch change occurs from level 4 to level 1, it's rate 1 that needs altering. You can see this easily if you look at the EG diagram on the front panel of your synth. The pitch EG parameter values that I settled on are as follows:

Pitch EG	1	2	3	4
RATE	70	99	99	99
LEVEL	50	50	50	90

The last component we have to set up is the amplitude envelope of he syndrum. This involves choosing suitable parameters for the carrier Operator's EG. The sound should have a fairly percussive start and then fade away over about half a second. Envelope generation on the DX instruments is essentially no different from that on any other synthesizer, so I don't see this side of things causing too many problems.

The percussive start to the envelope is obtained by setting both rate 1 and rate 2 to high (ie. fast) values. Level 1 is left full on at 99, which should always be the case for the main (or only!) carrier Operator, in order to avoid unnecessary loss of level. Rate 3 is then set fairly slow, to give the half-second fade out. Rate 4 is left alone since effects after key release are ignored for simplicity's sake. Note changing in the display when I'm setting up envelopes – instead, I tend to experiment by moving the data slider about until it sounds right. Not very scientific, but it works.

I found the following values effective for Operator 1's envelope:

Operator 1 EG	1	2	3	4
RATE	96	91	49	99
LEVEL	99	92	0	0

This gives you a useable syndrum sound (although it may not be that great over 5 octaves).

Next time we will look at making more complex (and therefore higher quality sounds) by using the factory presets as a starting point: this cuts out a lot of the hard work of programming a sound from the initialized voice with all the parameter entry that entails. ■

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FUTURE SHOCK

TECHNOLOGY

As technology gets cheaper and low-cost instruments start offering facilities that were once the preserve of expensive computer systems, we preview two new upmarket machines – the Series III Fairlight and PPG's Realizer – that ensure the big boys stay ahead of the pack.

By Simon Trask and Dan Goldstein

'RELAX. Take a few deep breaths, think pleasant thoughts. It may take a little while to become expert with the CMI III, but it will be worth it. Now get the CMI III up and running... When you and the machine are both ready, there are a few explanatory notes about using the CMI, and a tutorial, designed to lead you through some basic routines... After that, you will be confident of success.'

That, ladies and gentlemen, is what you read when you get your Series III Fairlight home, extricate the manual from its packaging, and open it up on Page 1. To be fair, there's no other obvious way of introducing an instrument so sophisticated, it makes its predecessor look like a Marconi valve radio.

But the Series II CMI was a hard act to follow. When it was unveiled seven years ago, it was musically, technologically and conceptually streets ahead of almost everything else – and it took everyone else a good while to catch up. But catch up they have, and the Series II is now being challenged on all sides by cheaper sampling keyboards and MIDI-linked computer / sampler / sequencer combinations.

But for the last three years, Fairlight's engineers have been aiming higher, and the result of their endeavors is the 16-voice Series III CMI. It's still a sampling-based instrument, but its *power* is orders of magnitude greater.

The machine is now starting to appear in small numbers in the US, but if you fancy shelling out a round $60,000, you might be in for a bit of a wait – there's a mile-long list of takers at Fairlight USA. For Series II owners, the massive cost is alleviated by the fact that 'old' Fairlights can be traded in for new, after which they're whisked back to Australia; what happens to them after that is anybody's guess.

In appearance, the new Fairlight doesn't look greatly different from its predecessor. The most obvious changes are a new slimline alphanumeric keyboard with inbuilt graphics tablet (much more effective than the earlier lightpen) and a new monitor. The keyboard is unchanged, which means it's six octaves (F-to-F) long and has plastic keys, and is sensitive only to attack velocity. This compares unfavorably with the new Synclavier keyboard, which is weighted and includes

New Series III Fairlight looks little different from its illustrious predecessor; inside, wholesale changes mean new system sounds better, records for longer, and costs nearly twice as much as Series II

Photos: Matthew Vosburgh

polyphonic aftertouch, but you could argue that with MIDI now forming such an integral part of the Fairlight's design concept, you could just as easily use a high-quality MIDI keyboard as controller.

On the storage front, the new Fairlight has both a 1Mb eight-inch floppy disk drive and a 140Mb hard disk drive onboard, plus a 60Mb tape streamer connected via the SCSI interface (more on that later). Transfer of sample data between internal memory and hard disk uses Direct Memory Access, a procedure that allows memory-to-memory transfer without tying up the central processor. Samples can be loaded from hard disk while a sequence is playing, and loading is so fast that a sample can be loaded as it's being played.

The Series III is also a *multi-tasking* system, which means that unlike MT's Editor, it can do several things at once. Thus you could be running a sequence while working on or loading samples, or altering MIDI allocations. It's rumored you'll be able to run two monitors to take advantage of these capabilities – should be great for people with two heads.

Beneath the surface, the CMI software is written on top of the 6809-based, Unix-like OS9 operating system, itself modified by Fairlight. It's possible to exit from the CMI software and indulge in such pastimes as word-processing, playing games, and re-writing Fairlight's software (maybe). Like its predecessor, the Series III is actually based on two 6809 processors; one of these runs the bulk of the system while the other is concerned mainly with sequencing. There are also two 68000 processors, one dedicated to handling MIDI and SMPTE, the other acting as an all-purpose waveform processor complete with its own 512K of RAM. Each of the eight dual-channel audio output cards has its own 6809 processor and 64K of working memory. The modular hardware design concept of the previous Fairlight has been retained, and consists of 27 slot-in cards which handle, in addition to the above, such features as disk drive interfacing, video display and SCSI interfacing.

Impressive as these technical details are in themselves, it's the implications they have on the Series III's performance that deserve most attention. The main areas of improvement lie in sample quality, sample storage, voice organization, MIDI and sequencing.

Let's start with sample quality. The old Fairlight's samples were eight-bit, of course, and the maximum sample rate was around 30K. The Series III allows you to sample in eight-bit or 16-bit modes, and in mono or stereo; all combinations can be mixed in performance. Default sample rate is the CD-standard 44.1kHz, but for stereo sampling this can be increased to 50kHz (actually 48kHz due to channel switching of the ADC), and for mono sampling a staggering 100kHz – though whether anyone really *needs* to sample at 100kHz is debatable. The maximum playback rate is 194kHz, allowing samples recorded at maximum rate to be replayed at up to an octave higher than their original pitch. Series II Fairlight users will be glad to know their current sample libraries *can* be used on the new machine, courtesy of a conversion program; the same applies for Page R sequences.

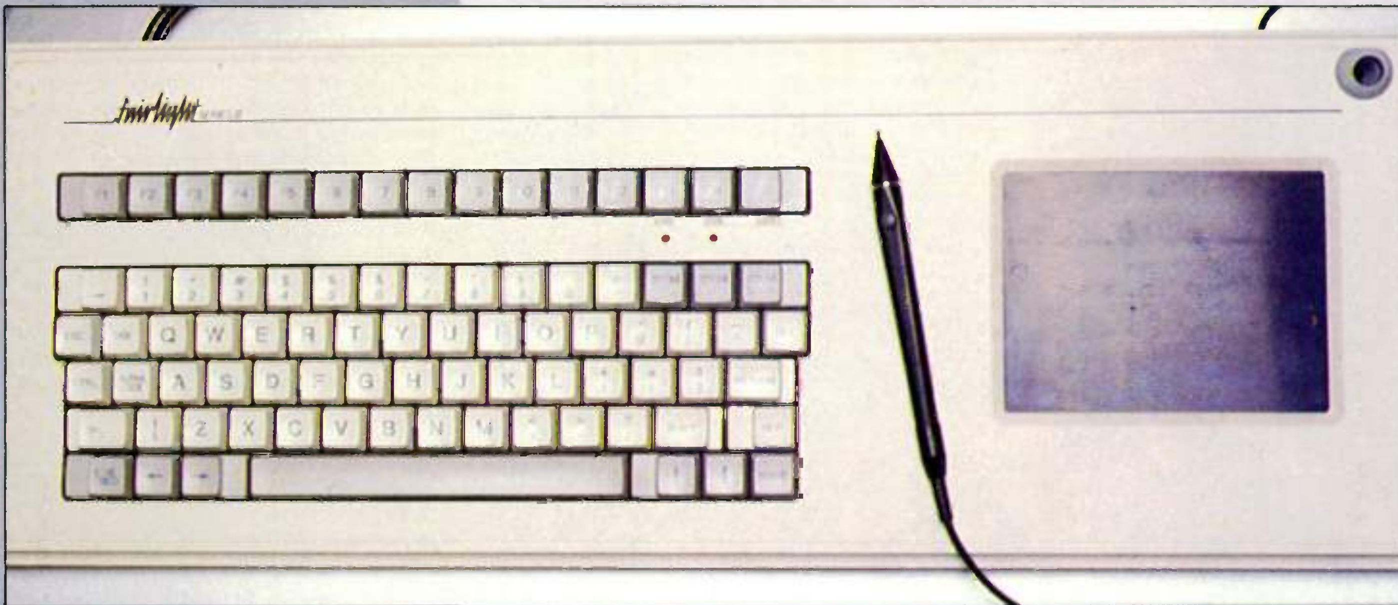
Since the original appearance of the Fairlight, the cost of memory chips and peripheral storage media has fallen drastically. This has enabled the Australians to adopt a more flexible internal architecture, so that whereas the Series II's samples are stored in a modest 16K of waveform RAM on each of its eight channel cards, all the Series III's sample data is stored in a common 14Megabytes of RAM. The waveform processor card handles the transfer of samples to the dual-channel output cards. This allows for greater flexibility in determining the number and duration of samples that are to be held in memory.

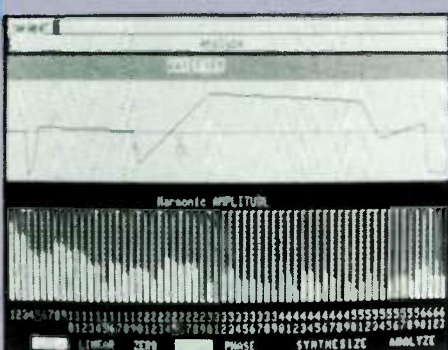
Voice organization has been expanded as a direct result of the new massive storage capacity. At the highest level is the System, which consists of Instruments, which in turn consist of Voices, which in turn consist of up to 128 Subvoices, or individual samples. The Instrument is what appears on the keyboard; the Voices which make up each Instrument are assigned an 'Nphony' value (this term is retained from the Series II, and refers to the number of Fairlight voices, up to the maximum 16). You can have one 16-voice polyphonic Voice or 16 monophonic Voices, or any combination in between. Voice layering depends on how many Voices are allocated to an Instrument – two Voices give you a simple dual voicing.

But the organization is a lot more versatile than that. For each Voice, you can allocate as many as 128 samples across (and beyond) the keyboard. So you can have a different sample on each key, or indeed any combination that takes your fancy, from simple two-way split up. How many of these samples you can play at once depends on the 'Nphony' value chosen for the overall Voice.

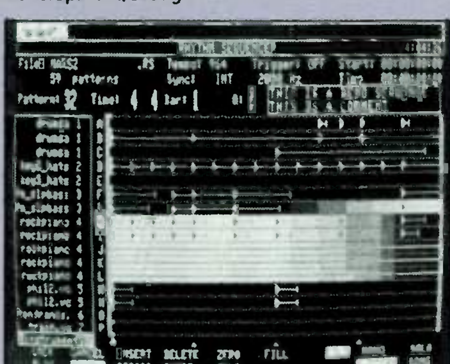
Whereas MIDI was an add-on to the Series II, it's been fully integrated into the new system. And with three MIDI Ins and four ▶

TECHNOLOGY





Fairlight page screens aren't drastically changed, but several pages have now been merged together in the interests of operational neatness; CAPS composer/arranger software is still under development, though



MIDI Outs lurking on the main unit's rear panel, this is unlikely to be a half-hearted implementation. As noted earlier, one Instrument can be played from the CMI keyboard at a time, but several Instruments can be resident in memory as part of a System. However, using the MIDI inputs (for instruments or sequencers), it's possible to play all the Instruments simultaneously, within the Fairlight's 16-voice limit. It's also possible to patch any input channel (MIDI or CMI) to any output channel (MIDI or slave CMIs). The range of possibilities is vast.

The Series III's version of MIDI is also closely tied in with the new machine's highly-specified CAPS sequencer. With four MIDI Outs, the latest Fairlight can accommodate up to 64 separate channels of MIDI data, which could account for 64 of CAPS' 80 tracks. That should provide more than enough scope for most people, and offers more than enough channels for the inclusion of multi-timbral MIDI instruments, which are becoming more common.

Continuing the sequencing theme, the Series II's Page R has been so phenomenally successful it would have been inconceivable for it to be omitted on the Series III. It's now known as the Real-time Sequencer, and is 16-voice polyphonic, with the voices structured as they are for Nphony; in other words, if you have a four-voice piano you can sequence it (in real or step time) in that format, though you can also choose to work on voices monophonically.

In most other respects, though, the Real-time Sequencer is unchanged from its predecessor. This applies also to the song construction page, where songs are built up from Sequencer patterns. There are 26 sections, each of which can chain as many as 255 patterns; these sections are then chained into songs, with each section capable of being repeated up to 127 times, or infinitely.

The centerpiece of the new Fairlight's sequencing power is undeniably CAPS, though. CAPS stands for Composer/Arranger/Performer/Sequencer, but unfortunately the software has yet to be seen in finished format. In fact, information on the system is extremely scarce. Apparently, it'll be an 80-track system featuring on-screen music notation, and will include MCL (Music Composition Language) in its array of facilities. It will be SMPTE-compatible (as the Rhythm Sequencer is now), and files will be interchangeable between the Rhythm Sequencer and CAPS.

Future options will include sample-to-disk (stereo or mono, 8-bit or 16-bit), music printing, a film music processor based on SMPTE, and a facility for connecting a whole array of Series III slave units, should your wallet be able to take the strain.

As well as MIDI interfacing, the Series III has a Click Out (which can be set in software to any frequency known to man), four Clock Outs (ditto), a metronome output, Roland DIN sync and SMPTE In and Out. Also included is an SCSI (Small Computer System Interface) connector. SCSI is an industry standard interface for connecting computers and peripherals, and in the Series III's case will allow for the connection of up to four slave Fairlight units, a second hard disk and optical disk storage. This last option will be of the Write Once Read Many (WORM) type, and will offer storage capacity in the gigabyte league. WORM disks (of all the great Fairlight acronyms, WORM is the greatest) are currently the nearest anyone has got to read/write optical storage, and in the Fairlight's context, will be best suited to master recordings.

The front end of all this increased power has inevitably undergone some changes as well, though there's plenty that'll be familiar (or at least half-familiar) to existing Fairlight users. Essentially, you still work within a page-based system, though windowing techniques have been adopted for the ever-useful Help pages and for disk directory listings, and the pages themselves are made up of windows.

At the moment there are seven pages: Page One, FFT (Fast Fourier Transform), FX, Sample, WE (Waveform Edit), RS (Real-time Sequencer), and Mix, though some of these are also broken down into subpages.

Page One is where you define voice organization and MIDI Input and Output routings. The FX page allows you to define

Oddball PPG Realizer also has deceptive exterior; complete software control means machine is potentially capable of replacing most other hi-tech musical instruments – at a price

parameters for individual Voices and Subvoices, while a subpage allows subvoices to be mapped onto the keyboard. The WE page lets you zero in on any section of a waveform for editing purposes (inverting, reversing, zeroing, rotating and so on). A Fairlight wouldn't be a Fairlight without Fast Fourier Transforms, and the new machine offers increased power in this area, for whereas the Series II could handle 32 harmonics and had a fixed phase, its successor has a mind-boggling 255 harmonics, while control over both amplitude and phase of each harmonic is possible. As it turns out, the FFT page is an amalgamation of several pages from the Series II, including waveform drawing and display along with harmonic 'faders' on screen.

The Mix page, which is the most recent addition to the Series III's architecture, is responsible for cut-and-paste operations, and also for the Merge function (which interpolates between waveforms) carried over from the Series II.

It's not possible, within the confines of this introduction, to do more than scratch the surface of the new Fairlight's capabilities. But one thing is clear. Fairlight have reaffirmed their superpower status with a machine that sets new, higher standards in sample quality and quantity (both excellent) and, just as important, in sequencing power. The Australians have rightly perceived that an instrument in the Series III's category should offer a powerful sequencing section that'll incorporate any number of other instruments via MIDI, giving it access to and control over a sound section far beyond its own capacity.

So the new Fairlight differs from its predecessor in being not only a powerful sampling instrument in its own right, but also the centerpiece of any music production environment.

Wonder how long it's going to take everyone else to catch up this time.

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DISTINCTIONS BETWEEN THE SERIES III FAIRLIGHT AND PPG'S REALIZER can be drawn quite easily. Whereas the new CMI represents the refinement of well-established techniques, the upgrading of a machine already accepted and respected by the industry it serves, the Realizer is a ground-breaker, an innovation made up not of tried and tested building blocks, but of many smaller innovations.

That does nothing to detract from the Series III. For thousands of committed Fairlight users the world over, it will have no competition. And maybe it's a little foolish to assume the two systems are competitors at all, since although their musical goals are similar, their ways of achieving those goals are almost entirely different.

Technologically, the Realizer's biggest leap forward is its independence of electronic hardware. For although it requires a fair chunk of digital memory and considerable processing power to carry out its tasks, all those tasks are carried out in software – including the sound-creation process itself. The Realizer has no oscillators or filters in the conventional sense, and little in the way of recognizable synthesizer hardware. Each sound is produced using eight software-controlled signal processors, and outputted via 16-bit digital-to-analog convertors.

PPG's transition into software has been made for the excellent and rather obvious reason that, if a machine's capabilities are defined by the software that controls it, those capabilities can be altered and extended at any time simply by writing new programs. So the Realizer is the first computer musical instrument that truly deserves to be called open-ended. No hardware upgrades should ever be required once the basic system has been purchased, unless you suddenly decide you want more memory space or processing power.

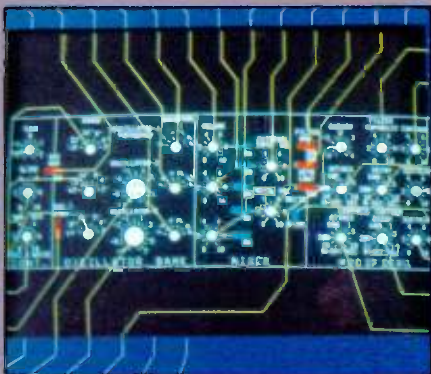
The Realizer's functions can be divided conveniently into three groups. The first of these covers sound-creation, an exclusively digital process which, as we've seen, is dependent entirely on the software programmed into the system. Theoretically, it should be possible to configure the sound-creating section in any way you choose – established or otherwise. Altering details of each configuration is something that can be performed directly from the Realizer's control panel (see later), but initiating new systems requires both a detailed knowledge of the arithmetical principles involved, and the programming technique necessary to stir the PPG's 68000-based signal-processing hardware into action.

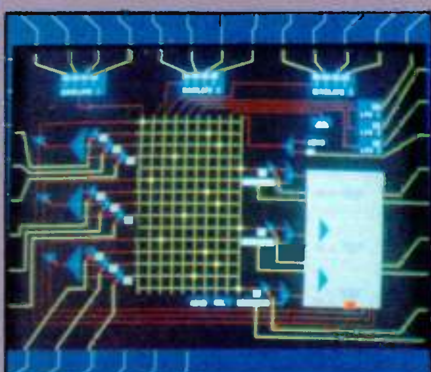
So far, the Germans have written software for two sound-generating configurations, one a carbon copy of the Minimoog's synth section, the other an FM system loosely based on the one used by the Yamaha DX series.

The Realizer also has facilities for digital real-time sound modulation. Its software-controlled effects processors are theoretically capable of modulating any sound source that can be converted into digital form and fed into the system, and again, the section benefits hugely from being based in software. Traditional effects like reverb, repeat echo, phasing, flanging, chorus and so on make only a small part of what the Realizer is capable of doing to a digital signal, though it should be pointed out that PPG's software writers have yet to tackle the modulation side of things in any great depth.

Third and last on the list of Realizer possibilities is digital recording. With the help of its built-in hard disk unit (more later), the Realizer can store 12 minutes of music in mono, six in stereo, and three in four-track format, all at a sampling rate of 40kHz. All recordings can be synchronized with the integrated MIDI Multitrack Sequencer.

More exciting than these sections individually, however, is the way their building ▶





Color screen displays show two alternative sound configurations, a Minimoog replica and an FM-based system; clever graphics are complemented by easy-to-use analog controllers, user-assignable to any parameter

blocks can be combined and routed in any order you wish. A typical configuration employing the capabilities of all three sections could involve two Minimoog voices, a single FM voice, a couple of PPG Wave voices and three sampled sounds, routed through a complex modulation section, and sent out on separate MIDI channels. The vast array of routing permutations conjures up visions of FM voices being routed through analog synth filters, repeat echoes being modulated by different rates of chorus effects, and guitar samples being played through distortion modules set up to simulate (and improve upon) the action of a traditional valve amp.

Externally, the Realizer is made up of two self-contained units. The first of these is a 6U-high rack-mounting module, which houses all the sound-generating electronics and has an 85Mbyte hard disk unit (derived from PPG's new HDU storage system) built in.

The second unit, shown here, is the Realizer board, best described as the system's remote control module. The board looks unconventional, but for a good reason: unlike almost every remote controller currently marketed, the Realizer unit incorporates a wide selection of analog controls that enable the system's software to be manipulated in a manner most musicians, engineers and producers are already familiar with.

The unit is dominated by a 14-inch, 16-color monitor of high resolution, on which all the Realizer's possible processing configurations can be displayed. Surrounding the screen are no fewer than 37 analog controllers (six sliders, 31 rotaries), all user-programmable to perform different functions, depending on the software currently in use. A separate numeric keypad caters for other specific programming requirements, and a graphic pad provides a precise means of controlling system configurations direct from the screen display.

The best thing about the Realizer's control unit is that, as long as the software is sympathetically written, it isn't too far removed from analog front panel designs. For instance, PPG's Minimoog software includes a screen display that is an exact equivalent of the original synth front panel, with each control represented on screen and parameters adjusted using the sliders and rotary pots.

FM configurations look less inviting, but they're still a big improvement on the almost total lack of visual feedback offered by most modern digital instruments, and like everything else on the Realizer, they benefit enormously from being represented in color.

The present state of the Realizer is nothing like as finished as that of the new Fairlight. PPG say most of the hardware is complete, leaving software writing as the main area for further development before the system is ready to go on sale. When it does become available, the Realizer is expected to sell at something in the region of $50,000. Not cheap, but considering the amount of research time that must already have gone into developing the Realizer, and the fact that any updates will be software-based and therefore quite affordable, the system looks conspicuously good value for money.

Unlike the new Fairlight, the Realizer is intended as an all-singing, all-dancing music production system that will effectively take the place of most other hi-tech musical instruments, not to mention quite a few outboard signal processors. It will be MIDI-compatible, however, so you'll be able to control Realizer voices from a dynamic keyboard – with just about any split/layer arrangement you can think of. Not unnaturally, PPG suggest their own PRK as the ideal black-and-white partner.

Other outside-world connections include sockets for footpedals (software-assignable to any parameter), and external audio inputs for anyone interested in using the Realizer to modulate 'ordinary' music signals like an electric guitar or mic input.

No external storage facilities are deemed necessary, since the Realizer's hard disk unit is capable of storing not just digital recording data, but also sound patches, routing configurations, and complete operating systems.

The mere idea of a central, sophisticated music production system capable of doing the job of many smaller units is certainly appealing. Especially when that system is as open-ended as the Realizer. It's difficult to imagine many users writing much of their own software, let alone designing entire sound-creation systems derived from complex arithmetical formulae. But the potential is there, and if PPG can develop software quickly and sympathetically enough, there should be no shortage of takers for each new package.

Yet PPG's real achievement lies not with developing a highly flexible, software-based computer music system. That has been done before, and the principle is now being adopted by a number of mass-market companies for use in much less ambitious machinery – witness Akai's range of systems disks for their S900 sampler, and Yamaha's software-inspired manipulation of delay technology in the SPX90 multi-effects processor.

What's surprising, and at the same time extremely welcome, is the extent to which the Realizer's designers have pursued the concept of user-friendliness. Gone are the unwieldy programming languages, the tedious digital parameter access systems, and the uncommunicative screen displays that have prevented so many computer music systems – at all price levels – from being really straightforward and 'natural' to use. In their place is a set of controls familiar to anybody who's meddled with an old monophonic synth or spent time behind a mixing desk, coupled with the most appealing set of screen graphics this writer has ever seen.

At a single stroke, the Realizer takes the flexibility of old patchbay-style synths, adds the modern technological benefits of digital signal processing and sampling and supreme 16-bit sound quality, and puts the whole lot in a package that's more accessible than most current budget polysynths. Or, as PPG put it so charmingly in their own literature, 'there is no discrepancy between what you touch, see and hear – it is just one harmonious whole.'

Hear, hear. *By Dan Goldstein* ■

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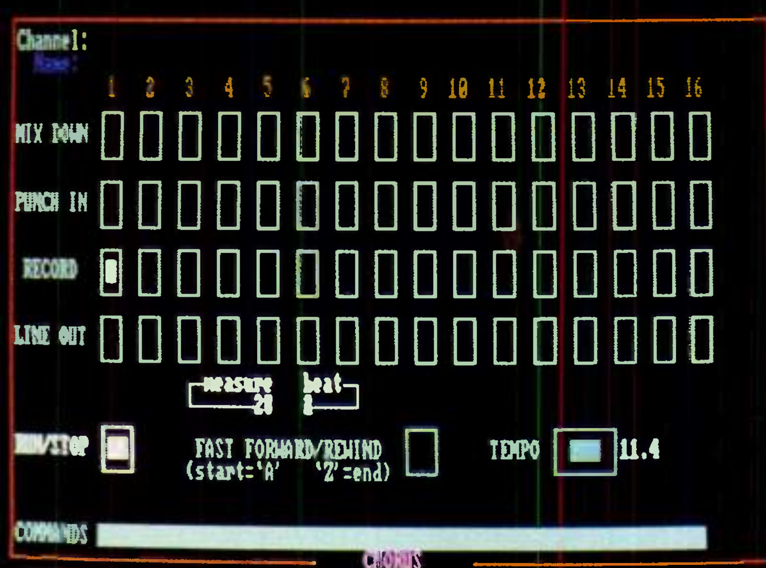

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Warm Music, Cool Thoughts

They may not yet be a household name, but Australian duo Icehouse are making some stirring modern pop music, and they have a fair bit to say on the subject of technology, too. *Interview by Tim Goodyer.*

MENTION THE NAME ICEHOUSE TO A MUSICIAN, and you'll probably get one of three responses: (a) 'Hey Little Girl', (b) 'Oh yeah, another Ferry/Bowie ripoff', or (c) 'How does an Aussie get a name as Welsh as Ivor Davies?'.

Well, for a start it's *Iva* Davies who's the driving force behind the Australian outfit that had a hit back in 1983 with the single 'Hey Little Girl'. And yes, his voice *does* bear more than a passing resemblance to those of Bryan Ferry and David Bowie.

It may seem things have been a little quiet for Iva and Icehouse over the last couple of years, but the truth is that more promotional activity surrounds the export of lager from Australia than music, and if you don't have your finger on the pulse, you can lose track of what an Aussie act is up to *very* easily.

But far from being empty, the intervening three years have seen the release of Icehouse's third album, *Sidewalk*, ballet and film scores, and extensive touring that's included a support slot to none other than Bowie himself.

To bring the story well and truly up to date, Davies and fellow Iceman Bob Kretschmer recently took a couple of days out from an American tour to talk to MUSIC TECHNOLOGY about album number four.

And so it is that I find myself in the foyer of London's chintzy Kensington Hilton Hotel, waiting for the two Australians to put in an appearance. And it's two Australians I get, except that one of them is manager John Woodroffe, Kretschmer arriving slightly later.

'We ran up a $500 bill last night at a

Russian restaurant', Davies explains. 'And that was just the vodka!' The story is corroborated by Woodroffe and the interview begins – a little cautiously...

The album *Measure For Measure* has been climbing up the album charts for a couple of months now. It's the first joint songwriting venture for Davies and Kretschmer, and, more than any previous Icehouse release, it displays a wide variety of styles and influences.

'Bob and I locked ourselves in a studio in Eastonville, which is a suburb of our city, for about three months', says Davies. 'That was kind of falling down stairs into the studio each morning and longing for bed. We kept fairly civilized hours actually, but it was a bit like being on another planet. We had a lot of fun trying different methods of writing on each song. I don't think we've actually worked out how to write yet, because there hasn't been one process which has been the same for more than one song; it's been like starting from scratch every time.

'It seems like an incredible amount of time when you make an album: three months of recording and then whatever in the mixing, but it's probably not, in most people's terms. The two producers involved had both come out of projects that had lasted two years, so it must have been like a long weekend in comparison.'

Said two producers are Rhett Davies and David Lord, the former well-known for his work with Brian Eno and Bryan Ferry, the latter for his production assistance on Peter Gabriel's fourth album. Did having two producers working on the same album cause any problems?

Davies explains: 'They actually worked independently of each other, except for the mixing process. I think it was a bit of an education for them, both working on something like this, because we were spending three weeks with one of them and then moving over to the other for the next three weeks, and they have completely different workstyles. We did five songs with Rhett and five with David, but somewhere along the line they crossed over. I think both producers were intent on extracting the real essence of the style, concentrating on vocal performances and stuff like that.'

We are soon joined by the errant Kretschmer. Contrary to expectations, he is cheerful and coherent. Perhaps Russian vodka isn't all it's cracked up to be...

Both Davies and Kretschmer are primarily guitarists, so it's not surprising there's a lot of guitar in evidence on *Measure For Measure*. But both parties favor the unpredictability of composing on a less familiar instrument.

'It would be nice to be able to sit down with a guitar and a couple of lyrics and write a song. I have done that on the odd occasion but generally it's a case of starting with some kind of sound. Quite often that sound will get ditched along the way, but more often than not, that's the best place to start.'

The vast compositional resources currently on offer to the synth player make the choice of keyboards as a songwriting aid an obvious one for Icehouse.

Davies: 'Generally I start on keyboards because I find it's more visual and also very logical. I sometimes find myself falling into old habits on the guitar which I like to avoid. I like the fact that I make mistakes on a keyboard, and more often than not I keep these unintentional accidents. I guess I use the Fairlight mostly, but I like the Prophet 5 for harmonies. I use the Fairlight as a sort of sketchpad for arranging and editing.

'For this album, we set up our own oblique strategies to get random things to happen – these usually involved huge amounts of alcohol! In the end it deteriorated into being completely random, to the point where we were putting the Fairlight in Record, and then running around for 20 seconds to find something to bang or play.'

Kretschmer: 'Lyrically we got all these little bits together that I'd written down, and Iva would sing a line, and then I sang a line while glancing at this mess of lyrics. You should hear the extended version of that!'

Davies: 'The strange thing about doing things like that is that you think you're dealing with a random process, when you're not. What you're dealing with is your method of selection at that point in time, which is dictated to by a certain frame of mind or whatever. The result is that even though the lyrics were scattered on the table, the selection made sense. It's an interesting way of doing things... one of the experiments that worked.

'There are many ways to write a song and we're not saying that we'd use one method exclusively, but if you start off with a guitar you'll get a completely different song than if you do it with this method.

'I always write onto tape, building up from there. We did all these songs in this way and ended up with 24-track demos, so we'd already spent quite a lot of time recording when we came to London. There had to be a good reason for doing it all again: we had to improve the arrangements and so on, and particularly the sounds. Starting off with the sound that was on the demo, which functioned perfectly well, we looked hard at it to see if we could find anything better.'

It's easy to lose the essence of an idea through overworking it, but Davies is happy

this hasn't been the case with *Measure For Measure*.

Davies: 'That was probably my biggest fear, but I guess in some ways Bob, myself, David and Rhett were kind of a foil for each other. It drove me crazy some of the time, not being able to see the necessity of doing all this. I always sat and watched, though – I had to remind myself what we were doing. On the other hand, the performance of the vocals and guitars was the counterbalance. That's why the album is called *Measure For Measure*, because of this balance: two producers working in totally different styles, and Bob and myself working so differently, too.'

As Australian music envoys, Davies and Kretschmer have perpetuated the national drinking myth with some panache. They've also set about promoting their second national export – the Fairlight CMI – with similar enthusiasm. Davies waxes lyrical on the subject.

'I've been working with a Fairlight for two-and-a-half years now. They're built not very far from where I live, so when they need to test out some software, I have a look at it and then ring them up and tell 'em it doesn't work. They're always de-bugging programs and they're going through this stage with the Series III at the moment, but I haven't actually got my hands on one of those – yet!

'The first time I used a Fairlight was to do the soundtrack to video maker Russell Mulcahy's first feature film, 'Razorback', which is about a huge killer pig.

'There's a simple logic about the Fairlight, and I think that's a real Australian trademark – not the simplicity, but the fact that it's so very logical to use. It was designed by a couple of computer people in league with a keyboard player. He really was there the whole way going "well I don't understand this bit..." or "can you make it do this..?"

'There's still no other practical machine for people who aren't mechanically minded. It's really just like using four or five video games; it's not necessary to be a computer buff to understand it.

'I have a theory about instruments: there's only one real natural instrument, and that's your voice. A piano is just a box with strings in it. I don't use synthesizers any differently than I do those sorts of normal instruments. People often say synths are sterile, but if you pick up a violin, which is considered to be a "human" instrument, you can choose whether or not you make it sound like a machine.'

Kretschmer: 'If you use a certain sort of sequencer, it's going to dictate what the song is going to be like. If you want to use that, you can – it's available. I like guitar bands where you don't have to use all this wonderful machinery.

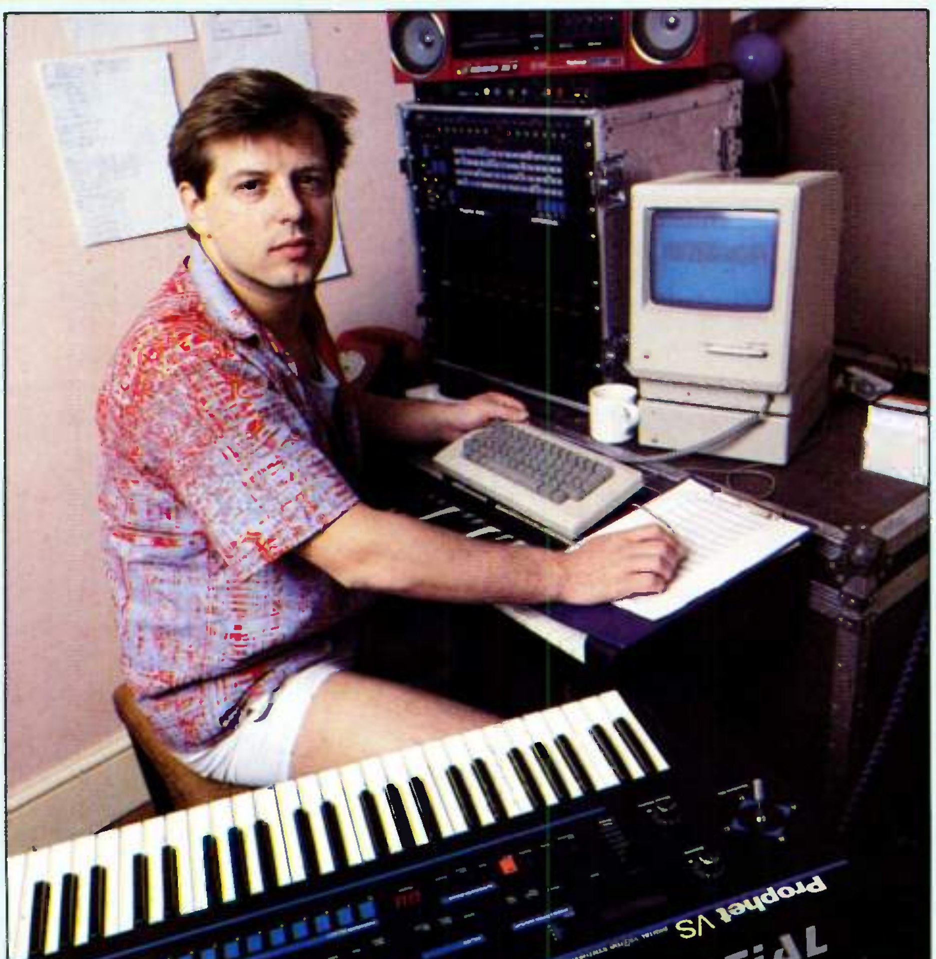
"People say synths are sterile, but if you pick up a violin, which is considered to be a 'human' instrument, you can choose to make it sound like a machine if you want to."

Iva and I both like good songs, and hi-tech has never impressed me as being the most desirable thing. But certainly, using the Fairlight as we did for the ballet and the album, we couldn't have got that amount of material together in the time we had, so they're great for that sort of thing, they're great tools.'

Davies: 'I like machines you can trick. Unless you can throw a spanner in the works and get the thing to hiccup, it doesn't really interest me. Quite a few times, what we ended up doing was setting up a situation where the Fairlight would fool itself. There are simple ways to do that: for example, creating a piece of music with a certain set of instruments and then loading a completely random set of instruments which have nothing to do with it. It's like writing a set of drum rhythms and then loading a set of violins instead of the drum voices – the pitch that the drums have is nothing to do with what is going to sound good harmonically with the new sounds. You don't know what harmony you're going to have. Sometimes you get a cacophony, and sometimes you come up with something amazing.'

Kretschmer: 'It's good and spontaneous because if you make a mistake you can end up saying: "well, that sounds great". But then, Iva's got a great knowledge of music. He's a trained musician, so he can sort out what's going on harmonically as well. You really have to know your music to get full use of all the instruments. In the hands of someone else, it's just not used properly.

'The other interesting thing about the Fairlight is it doesn't matter what you set up



SIMON LLOYD...

...is the name of the session player whose computer programming helps give the new Icehouse album a distinctive hi-tech sound. *Interview by Sharon Gilbert.*

How did you become involved with Icehouse?

Through the bass player, Guy Pratt, who is now working with Bryan Ferry on his new album. I used to play in various bands with Guy when I was playing the saxophone, and he joined up with Icehouse about three years ago, together with the other keyboard player, Andy Qunta, who used to work with Hazel O Connor. They needed a sax player in a hurry and Guy recommended me. I had about three days getting to know the set and then went off on a European tour with them. Generally the band only tour for three or four months of the year, and the rest of the time is spent commuting, as Bob and Iva live in Australia and myself and Andy Q live here in Britain. I do most of my programming work initially, as long as you have something going – even if it's a drone or a thump or a clang. So long as you can work with it, that's enough. With quite a lot of the songs that we wrote this time, we discarded what we'd originally set out with. 'Angel Street' had these incredible sitars and ended up with this kind of Beatles thing. We just removed what we started off with.'

It's curious that Davies quotes the human voice as the only 'natural' instrument, and yet leaves his own vocal style open to criticism of being derivative. It's obviously an argument he's encountered before, as his reply is as swift as it is considered.

'I've had this criticism of sounding like Ferry with 'Hey Little Girl', but I think my primary task is to extract the individuality out of the way I can sing. Having a lot of respect for Rhett and David, I followed their suggestions, and I think the result of that with this record is fairly obvious: I don't seem to sing as derivatively as I have done in the past.

'I guess there are a lot of people who don't sing these days. The people you spoke about are two singers, and perhaps to that extent there are unwritten similarities. I've been compared to Frank Sinatra, but he's a singer so what can I say?

'It comes down to the nuts and bolts of singing, really, because my voice has grown up a lot in five years, if only from the amount of live work we've done. It's interesting, going back over the earlier recordings and hearing how weak the voice was, and how it's improved. When you talk about someone like Bowie, you talk about a very strong voice.

'I think you can apply vocal style to the kind of context it's in as well. The diversity of the album's material means that some of the songs are quite soft, and the requirement for that is not an overriding style. On the other hand, some of our songs are really powerful and the style changes depending on that context – that's always been my guideline. Variety is very important to me, because I find it difficult to listen to a whole album from anybody. It's far easier for me to listen to the radio – I've got to have a bit of this and a bit of that.' ■

M U S I C

here in London along with a few other projects, and the rest of the band have their own little projects as well, but we still get together for recording and touring.

So what exactly is your role in the band? Do you become involved in the writing and arranging side of things?

Well, this latest album was for the most part written in Australia by Bob and Iva. Had Andy and myself been there, we would have participated, but as it was, I didn't become directly involved until recording started here in London. Most of the stuff is written on the old Fairlight Series IIX. On this album I was doing a mixture of things: I was playing baritone, tenor and alto saxophone, trumpet, and doing some programming on the Fairlight because I've got my own library of sounds.

So you've built up your own collection of samples?

Yeah. I have samples for Synclavier, Fairlight and this, that, and the other. Generally I keep everything I've got on Sony PCM-F1 tapes, as well as on idividual disks for each particular sound; that way I can always dump from the F1 tapes onto anything. I've also got a Prophet 2000 module in my home setup which is a good thing as we use the Prophet a lot, especially on tour.

I've also just got a Prophet VS which I'll be using on this tour, and Andy will be using a Roland JX10. I've spent about two weeks programming the VS; it comes with lousy presets, but all in all, it's a wonderful machine.

Which of the machines you used with Icehouse did you get on with the best, and which did you find the easiest to program?

Well, Iva owns a Fairlight and he shipped this over from Australia to use during the recording of "Measure for Measure". I get along well with it. It's certainly more accessible than some machines I've come accross, and much easier to handle whatever you want from it. Apart from that I like the Prophet 2000 a lot, and probably use it the most, especially live. I think it's a great machine to work with. For example, the telephone segment in 'Mr Big' was created using Iva's Fairlight: I took the final mix off the album, sampled two seconds of it, put it in the Prophet 2000, put a release loop on it, got my start and end marks right, and I now can fill an auditorium with exactly the same sounds that I had in that song. Those effects go down very well live. In fact, I've got the majority of stuff I did on "Measure for Measure" dumped onto F1 tape, which can be used in conjunction with the Prophet 2000 on stage, enabling us to recreate studio sounds live. We also do the backing vocals on the Prophet 2000 live. We found it useful to sample some of the backing vocals from the album, loop them, and add them in with ours on stage, because they're always dead on pitch and we can key our singing to them.

But you can run into problems doing this sort of thing live, because if you have a line of vocals you're trying to sing and you have one long sample, and happen to hit the key at the wrong time, everything gets out of sync, and that can be very dangerous.

Are you going to continue working with Icehouse?

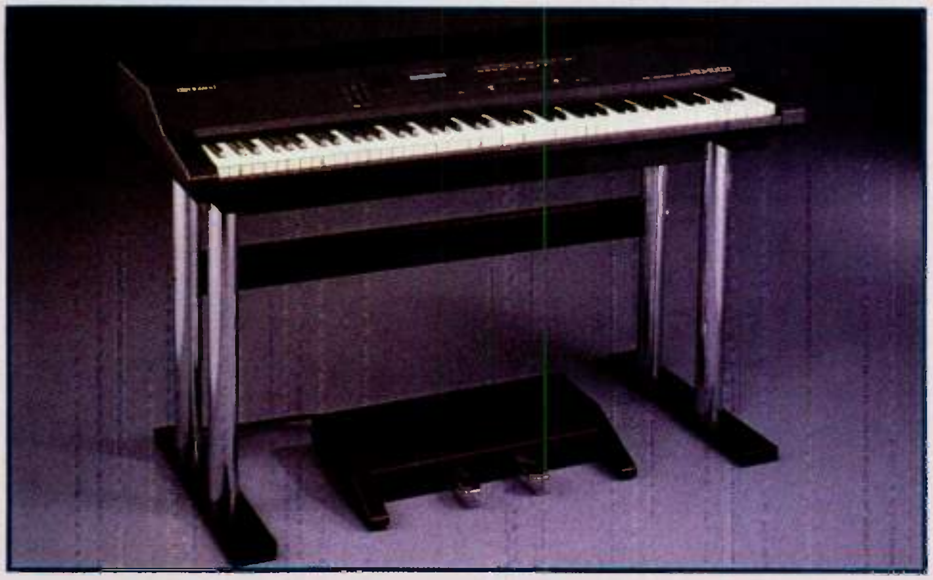
Oh yes, very much so. We're in the middle of a world tour at the moment. We'll be off to New Zealand in a couple of weeks' time and then on to America for a few weeks, then back to Europe, taking in London, and on from there to Australia, probably via Japan.

And are there any other interesting projects in the pipeline?

Well, I'm certainly kept very busy. Aside from the work I do with Icehouse, I get involved in a fair amount of projects. For instance, I've been involved in a couple of film soundtracks, the most recent of which is "Runaway Train" with John Voigt. I also worked with composer Trevor Jones on the soundtrack to "The Dark Crystal", and Bowie's latest movie "Labyrinth" which was very interesting, though very time-consuming. It's a seven-day week, working from sometimes five in the morning doing film work – it's a bit of a shock to the system! But I really enjoyed it, and you soon get used to that kind of routine. ■

R E V I E W

Roland RD1000 Digital Piano



Avoiding all existing digital techniques, Roland's engineers have come up with a new sound-generating process – Structured/Adaptive Synthesis – for the simulation of piano sounds. The first instrument to have SAS is the RD1000.

Review by Dan Goldstein

NO MUSICAL INSTRUMENT has been more heavily imitated than the piano. From difficult beginnings as a grudgingly accepted replacement for the harpsichord, the pianoforte has matured to become the inspiration for a thousand new forms of music, and is probably responsible for introducing more people to the world of music than any other instrument. The result? Pianos are in as huge demand today as they ever have been.

But several factors have conspired against the piano, and it's these factors that have spawned the instrument's imitators.

First, all pianos, whether they're of the grand or upright types, are large, bulky and awkward machines to move around. In today's highly mobile society, that doesn't bode well for the future of a musical instrument.

Second, the piano's inherently complex mechanical structure is becoming more expensive to produce by the minute, in terms of both material and labor costs. So, fewer people can afford new pianos than the manufacturers would ideally wish.

And third, the piano is still capable of producing just the one – admittedly immortal – sound, while today's hi-tech synths and samplers can have their sounds altered at will by the user.

To be successful, a modern alternative to the piano has to score over its predecessor in all the three areas just discussed. Electronic pianos have been more portable, more affordable, and more versatile than their acoustic counterparts for some years now. But until recently, they hadn't succeeded in fulfilling a fourth, crucial criterion: they didn't *sound* too much like a piano.

Specialist electric instruments – like the Clavinet and Fender Rhodes – have become appreciated for producing a sound which is distinctively *not* piano-like, and these, too, have spawned their own legion of imitators. But so far, all attempts at replicating the original piano sound have failed in one way or another – usually several ways.

The SAS System

BUT HERE WE ARE IN THE FALL OF 1986, and the Japanese musical instrument industry – one of the major world centers of piano production – is preparing itself for a revolution in contemporary piano technology, with each corporation putting its money on a different system for recreating the classic piano sound.

Matsushita (Technics to you and I) is sticking to the PCM (pulse code modulation) technology it's been using in home multi-keyboards for some time, with the PX range of electronic pianos employing PCM systems to stunning effect. Yamaha is sticking by FM – the system that's made the DX7 synth such a worldbeater – with its recent PF70 and PF80 pianos; their realism isn't quite as great as that of the Technics range, but they cost a good deal less to buy. Korg, meanwhile, is using digitally stored sampled piano sounds in its new SG1 (the SG stands for Sampling Grand, but the instrument doesn't actually sample), and Ensoniq – the only US company engaged in the great piano race against time – is doing the same with its new Piano.

But it's Roland, a company with a history of going the long way round a problem in order to come up with the best solution, who've embarked on the most rigorous development program in the quest for The Perfect Piano Sound. Roland's engineers, with a wealth of knowledge in the PCM and sampling areas, have rejected both systems for the company's new range of digital pianos, despite the fact that they're used in other Roland instruments. PCM, they claim, is just right for drum voices, while sampling is good for tuned orchestral and rock band sounds like strings, brass, guitars and so on. According to Roland, the piano deserves a new system of its own, so the company's engineers have given it just that – a new digital resynthesis system that goes by the name of Structured/Adaptive Synthesis, or SAS for short.

Initially, precise details about the workings of SAS have been difficult to get hold of outside Japan. Perhaps understandably, Roland's R&D people are being guarded about releasing the fine details of what the process does and how it does it. What we do know is that, in developing SAS, Roland engineers have painstakingly sampled all 88 notes of several different brands of grand piano (Steinway included), and analyzed the resulting waveforms on a mainframe computer. More important, the study then went on to analyze the changes in harmonic structure that occur not only as pianists play up and down the keyboard, but as they play one note with different velocities. In total, the harmonic structure variations at 128 different velocities – all the way from pianissimo to fortissimo – were analyzed *for each key*. Which goes some way to explaining why it's taken Roland so long to come up with SAS...

Having created an advanced digital algorithm of the pianos' harmonic relationships, the design team then set about dedicating the information onto a single custom VLSI chip. They succeeded, and the result is a range of new instruments that utilize this chip. Included in the range are two domestic machines – what Roland calls 'Contemporary Keyboards' – named HP5500 and HP5600, and two pro instruments: the RD1000 piano and its modular counterpart, the MKS20. It's primarily the RD1000 we're looking at here, but seeing as the MKS20 is identical in all important respects, it might be more suitable for players short on space or dollars.

The RD1000

IN THE TRADITION OF PIANOS, the RD1000 is no lightweight. It weighs in at a meaty 95lbs, with the pedal unit (housing soft and damper (sustain) pedals) adding a further 18lbs, and the stand adding 29lbs or 39lbs, depending on which model you go for. So much for portability, though it should be stressed that the RD1000 is still a deal lighter than the average acoustic upright (let alone a grand), and that most of its new digital rivals are no better in this respect.

And in any case, the Roland's massive weight and dimensions (nobody's yet found a way of condensing an 88-note keyboard without sacrificing playability) mean that it should stand up well to any abuse you, your roadcrew, or your fans can throw at it on the road.

That 88-note keyboard is made up of coated wooden keys, and feels a good deal softer and 'easier' on the fingers than a normal acoustic keyboard. That will delight synth and organ players brought up on plastic keyboards, who at last have the chance to use a more responsive set of ivories, without having to go through a period of culture shock. It should also please hardened piano fans, even though it may take some while to adjust to what is undoubtedly a 'looser' system than what they're used to.

Above the keyboard is a sleek control panel, sparsely and attractively decorated in the contemporary idiom. Immediately obvious are a 40-character, backlit liquid crystal display (not as helpful as some, but then it doesn't have too much information to convey), and the Alpha dial, centerpiece of Roland's new – and much vaunted – system of parameter selection and value adjustment.

Parameter selection? Yup. Unlike most of its rivals, the RD1000 offers a number of programmable parameters, which can be used to tailor each sound to your own tastes and requirements. But more on these parameters later, once we've gone into what impression those sounds make on first contact.

The Sounds

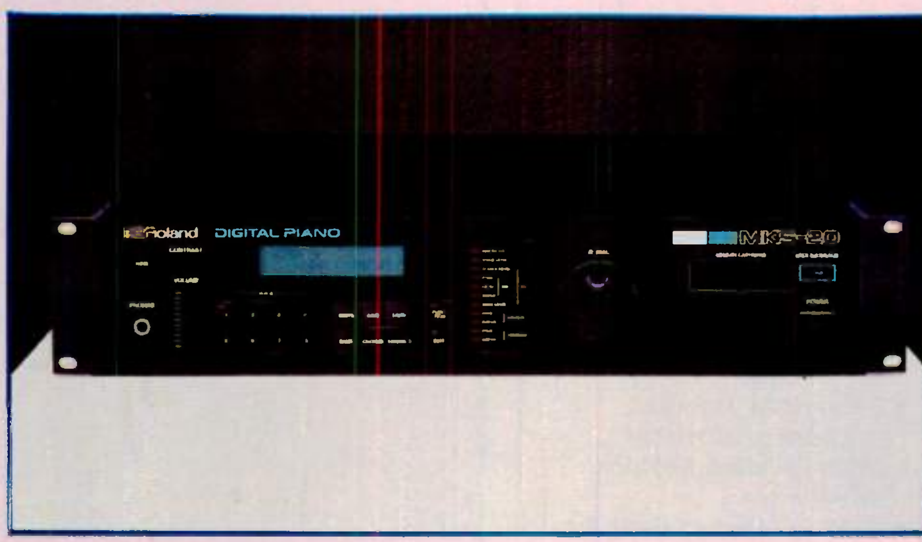
ESSENTIALLY, THE RD1000 CONTAINS EIGHT digital algorithms based on the harmonic characteristics of eight different sounds: three acoustic pianos, a harpsichord, a clavi (read Clavinet), a vibraphone, and two electric pianos (both fairly Rhodes-like). But these sounds – stored in ROM within the machine – provide only the starting point for further variations. In total, you can store seven edited versions of each sound in RAM, as the RD1000 arranges its voices in eight banks of eight. So, we have a total of 64 different sounds, eight of them preset, the other 56 user-programmable. You can dump 64 of your own edited sounds to Roland standard M16C memory cartridge (the socket's on the back panel), though you can't give them names as you can on many programmable synths, which could lead to some identification problems if you're going through a lot of cartridges.

The three acoustic piano voices are subtly different from one another, in much the same manner that you'd expect makes of grand piano to sound different. Working with the three preset sounds, the differences are mainly concerned with frequency curves – though the fact that the variable parameters include a three-band EQ section means the distinctions become blurred once you start getting into edited sounds.

But no matter. All three piano banks display a fine sound balance, especially at the top end, where the hammer strike forms such a significant part of the sound on an acoustic piano. Mid-keyboard, things are still rich, vibrant and above all realistic – though there is a 'zing' to the output which I've heard some players describe as an 'electronic' element marring what is otherwise a splendidly natural-sounding acoustic tone. You can use the EQ controls to eliminate the offending zing, but at the expense of a little brightness. The same effect makes its presence felt at the bass end of the keyboard, too; it's almost as though, in an attempt to stress their new machine's realism, Roland's engineers have overemphasized the characteristic buzz which pianos exhibit over their lowest couple of octaves.

But don't get this problem blown up out of proportion. It certainly didn't bother me when I was playing the RD1000, and my guess is that many players – especially anyone brought up on a Yamaha electric grand, which exhibits a similar but much more pronounced effect – won't even notice it.

The beauty of the RD1000's piano sounds lies in the realism with which their harmonic structure alters subtly as you hit the keys with varying degrees of velocity. It's something you can't put your finger on (pun not intended) until it's pointed out to you, but once you've played an electronic piano whose output changes the way the Roland's does, other instruments – including the RD1000's main competitors – start to sound lifeless by comparison.



Of the other five preset sounds, the harpsichord is bright and (my notes say) flavorsome, and of course not too sensitive to velocity – whoever heard of a harpsichord with a velocity-sensitive keyboard?

The clavi is similarly bright and lacks little in the realism department, though again, there's just the merest hint that an overemphasis in the upper midrange deprives it of a little warmth. Close your eyes, though, and you'd never know the guy in the room wasn't playing a vintage Clavinet D6.

The vibraphone and electric piano sounds represent, for me at least, the RD1000's finest moments. The vibes, especially, are wonderfully metallic without ever sounding brittle, and with the help of the built-in tremolo unit (see later), they leap out of the speakers with all the vitality of Gary Burton in overdrive. And you don't need to be a virtuoso vibes player, just a keyboardist with the minimum of dexterity and a moderate idea of how the vibraphone is played.

As I've said, both the electric pianos are close approximations of a Fender Rhodes, and like the acoustic piano sounds, they feature superb re-creations of the *mechanics* of the instrument being imitated. And, because nodody is going to object to an electric piano sounding a little non-acoustic, the RD1000's 'zing' doesn't intrude here at all. No matter how good an imitation of a Rhodes you've heard a DX7 perform, the RD1000 goes one better.

The Parameters

ONE OF THE PROBLEMS WITH SO MANY ELECTRONIC pianos (both yesterday's and today's) is that they don't present the user with much opportunity to alter their basic sounds. I guess it all comes down to the purist ethic – if you're trying to imitate a piano, why mess the sound up with a load more electronic gadgetry?

But as long as musicians are playing electronic pianos through graphic EQ units, digital delays, and psychoacoustic enhancers, there'll always be a reward for the enterprising manufacturer who sticks a set of user-programmable parameters on his piano.

That's just what Roland has done with the RD1000. The parameters are divided into two groups: those that govern the entire instrument (titled System Functions), and those that are programmable for each of the 64 voices (titled Voice Functions). The former include MIDI channel selection for reception and transmission (with a 1-16 range offered for both), and keyboard touch response; the MKS20 omits the latter, but the RD1000 offers four values for it, labeled A, B, C and D. B is the default value, under which the shift in dynamics and harmonic structure increases in a linear way with the amount of velocity applied to a key. Under setting A, the increase in volume and tone is less ▶

pronounced, while settings C and D introduce a more dramatic alteration.

The voice-programmable section comprises an individual Voice Level control (variable from −13dB to +2dB in 1dB increments, with 0dB being the default value for all voices); the three-band EQ system; and Chorus and Tremolo modules, both with variable rate and depth (15 levels for each).

Initially I was disappointed that Roland hadn't taken the opportunity of fitting a programmable six- or eight-band graphic EQ to the RD1000. In some ways I still am, because there's no more precise – or more instantly visual – way of adjusting a frequency curve. But the Roland's three-band system is a little more versatile than it appears at first. Because although the bass and treble sections are set by simple shelf EQ controls with preset cutoff frequencies (100Hz and 10kHz respectively, with 10.5dB cut/boost variable in 1.5dB steps), the midrange gets a parametric EQ section with a center frequency adjustable between 400Hz and 4kHz, the same cut/boost control as the bass and treble, and a separate 'Q' (or bandwidth) control, variable in eight arbitrary stages, with the highest value giving the narrowest bandwidth.

"The RD1000 contains eight digital algorithms based on the harmonic characteristics of eight sounds...but these provide only the starting point for further variations."

Programmers who've grown used to the idea of sending digital samples through analog synth sections will be disappointed to find no filter or envelope controls on the RD1000, but these would have been prohibitively expensive to develop in software terms. And all in all, the existing range of programmable parameters is capable of inspiring a wide selection of different edited voices, and fully justifies the inclusion of programmable patch memories on the RD1000.

MIDI

IT'S INEVITABLE THAT, SOONER OR LATER (probably sooner), a bunch of musicians are going to want to use the RD1000 as a MIDI controller keyboard, taking advantage of that long, luxurious, velocity-sensitive set of ivories to manipulate external voices from other MIDI machines. And when they do, they'll find the RD1000 well equipped in many ways, but lacking in a couple of MIDI features that could have increased its flexibility as a controller.

On the positive side, the machine can receive patch-change data values 1-64, and transmit the same data values in the range 1-128 (you use the Alpha dial to select a second 'bank' of external program changes). The RD1000 has two further controllers – an External Volume control next-door to the internal one, and an expression pedal that plugs into the pedalboard – which can be assigned, in six different permutations, to control three MIDI functions: Foot Control, Volume and Expression. The machine is capable of sending all three of those codes as they are, but converts them all to volume data when it receives them. The soft and damper pedals, tremolo and chorus settings are also received and transmitted by the RD1000, using their standard MIDI control change codes.

Less welcome than these features is the fact that although it both receives and transmits key velocity information, the RD1000 can do neither for aftertouch. Now, that isn't so bad in the context of the instrument's own internal voices (well, do you want to apply aftertouch to a Steinway?), but it does diminish the piano's appeal as an all-purpose MIDI controller.

A further point against the RD1000 here is its inability to split its keyboard into zones, with each zone assigned to a different MIDI channel. This prevents you from, say, playing an internal piano sound at the bass end, and an external lead synth voice further up.

Conclusions

NOW IS THE PLACE, I GUESS, to mention that I'm not normally all that enamored of the electronic piano as a species. As a synth player who thrives on the chance to manipulate sound in as many ways as I can think of, and who occasionally resorts to an acoustic piano for songwriting chores, I find most electronic pianos bland, predictable and sonically uninspiring.

But the RD1000 is none of these things. It's one of the few instruments I've come across – in over three years of reviewing modern keyboards – that has stopped me from thinking about my job as a writer by making me think about *music* more than anything else. After 30 minutes with the new Roland, my head was full of new musical ideas, and my notebook was empty.

Why the sudden rush of inspiration? Well, I guess a lot of it has to do with the way the RD1000 sounds. Some may accuse SAS of producing a forced sense of naturalness, but my feeling is that, for pitched, percussive sounds of the kind the RD1000 and MKS20 are intended to reproduce, the system is a winner. The sense of dynamic and harmonic realism SAS seems capable of generating is intense, and compared with sampling, it has the advantage of being relatively undemanding of memory: you don't need a mass of expensive chips to put SAS on an instrument, just plenty of patience at the development stage.

And although Roland's engineers are stressing that SAS has been developed specifically for piano applications, I for one eagerly await its adoption in other fields. Anyone for a fully programmable, SAS polyphonic synthesizer?

The RD1000 may be flawed in some peripheral control areas, but as a tool for making music, first and foremost, it's worth every cent of its asking price.

It's the best electronic piano I've tried. End of story. ■

PRICE Roland RD1000: $3,695; MKS20 module: $1,795
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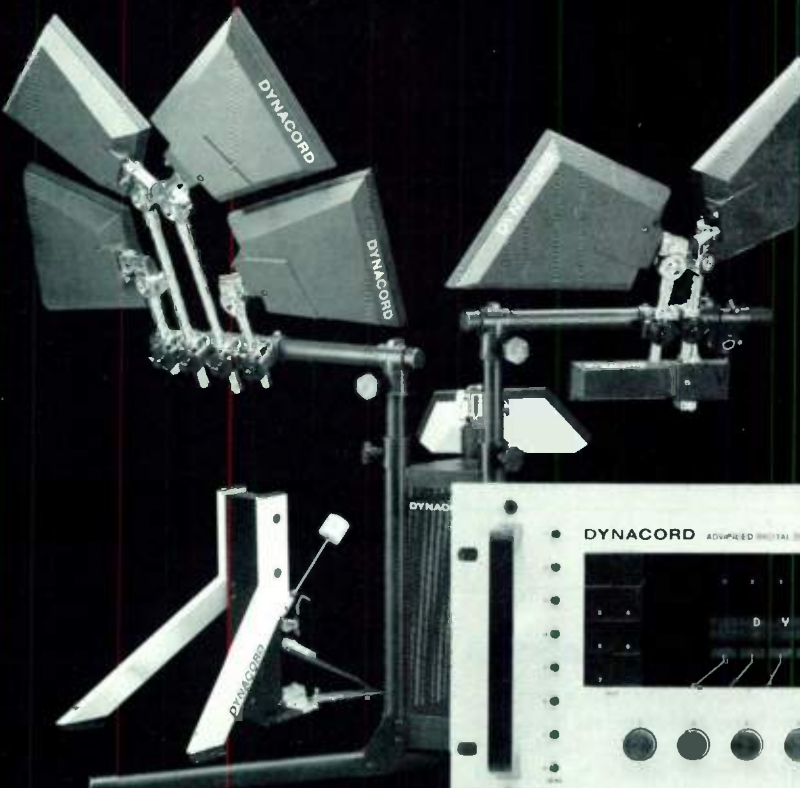
DYNACORD

THE NEW GENERATION



POWER PADS

Tired of that old familiar shape and look of electronic drum pads? Dynacord's new Power Pads will give your electronics some personality with a futuristic, ultra-slender design. A unique spring construction (compliance-controlled suspension) allows the Power Pads and Power Kick to respond just like a natural drum head. The impact surface gives into the body of the drum slightly when struck. Hardware? The Dynacord Drum-Caddy completes the modern percussionists' futuristic new look while putting an end to the "forest of stands."

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RHYTHM STICK

How many times have you wished you weren't hidden by your drumset, relegated to the back of the stage with the other band members between you and the audience? The Rhythm Stick is a remote MIDI controller which enables the drummer/percussionist to play any MIDI instrument from the front of the stage. Any musician can play the Rhythm Stick. The controller features eight trigger selectors to specify the "instruments" and two slap sensors with which the selected sounds can be played. The Rhythm Stick also remembers all programs entered in its user programmable memory. So if you're tired of being hidden by all your hardware at the back of the stage, get yourself a Rhythm Stick.

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Making Sense Of...
THE MIDI IMPLEMENTATION CHART

M I D I

This month we move from Channels, Modes and Note Numbers to look at those sectons of the chart which deal with the interpretation of performance data, velocity, aftertouch and pitch-bend. *By Paul Wiffen.*

	Function	Transmitted	Recognized	Remarks
Velocity	Note ON Note OFF	O 9n v=64 O 9n v=0	O 9n v=1-127 O 8n v=1-127	

	Function	Transmitted	Recognized	Remarks
After Touch	Keys Ch's	O ×	O O	

	Function	Transmitted	Recognized	Remarks
Pitch Bend		O	O 0-12 semi	9-bit resolution

THE VELOCITY SECTION OF THE CHART tells you whether or not the MIDI instrument you're reading about is able send or receive information about how quickly the notes are played and (in some cases) released. The available range is 127 values, with 127 being the fastest. If the instrument can transmit or receive velocity for Note Ons, then this will be shown in the appropriate column as v=1-127. If the instrument is unable to interpret velocity codes, then Note On will be shown as v=64: this means that any note will be sent or received as having a standard 'average' velocity. The 9n which precedes the v values simply refers to the MIDI status byte for Note On.

Now, most instruments don't implement release velocity so there is a shorthand available for Note Off: simply Note On with velociy zero. This is therefore sown as 9n v=0. This means that a string of Note Ons and Offs all have the same status byte (9n), so running status – which allows the status byte to be omitted if it is the same as the last piece of information – can be brought in; this, in turn, means that each note only needs two bytes instead of three, thereby keeping the amount of MIDI data to a minimum, and preventing the buss fom becoming clogged.

Only those keyboards which can transmit and/or receive release velocity actually need to use a separate status byte for Note Off – 8n. The velocity range for release is just the same as that for the attack, so the code shown against either transmitted and/or recognized would be 8n v=1-127. As in our example, many MIDI instruments, such as Roland drum machines and keyboards, are not velocity-sensitive when played directly (the cost to make the player interface – be it a keyboard or drum pad – is just too high), but are still able to respond to velocity data via MIDI. The example shown here gives the sort of velocity implementations you might expect to see in such a case.

The section dealing with pressure or aftertouch is a little simpler, as there is no choice of how you send pressure as there is with Note Offs, for example. All that's required is the O or × which mean 'implemented' or 'not implemented' respectively. However, there are two separate types of aftertouch data that can be sent. The first of these, 'Keys' – or polyphonic pressure as it is often known – allows for each note can to its own aftertouch amount. Normally this is only found on expensive machines like the Yamaha DX1 and Prophet T8, because being able to derive the amounts for MIDI transmissions requires a pressure sensor under each key.

Other machines – like the DX7 – adopt a more economical approach and just have one overall pressure code. This is referred to as monophonic pressure, and is shown on the MIDI implementation chart opposite Ch's (standing for channel). In other words, all notes on that channel share the same pressure amount at any given time.

Like velocity, you'll find some keyboards able to receive more aftertouch information via MIDI than they can generate or transmit themselves. Clearly this is more likely with rack modules, but it applies to many keyboards, too – pressure often costs even more than velocity to implement mechanically, though its interpretation is a matter of few simple software routines.

And so we come to pitch-bend, which is one of the thornier areas of MIDI, not because there is incompatability between the way various manufacturers transmit and receive pitch-bend data, but because most keyboards have a different pitch-bend range, which can often be varied on the front panel.

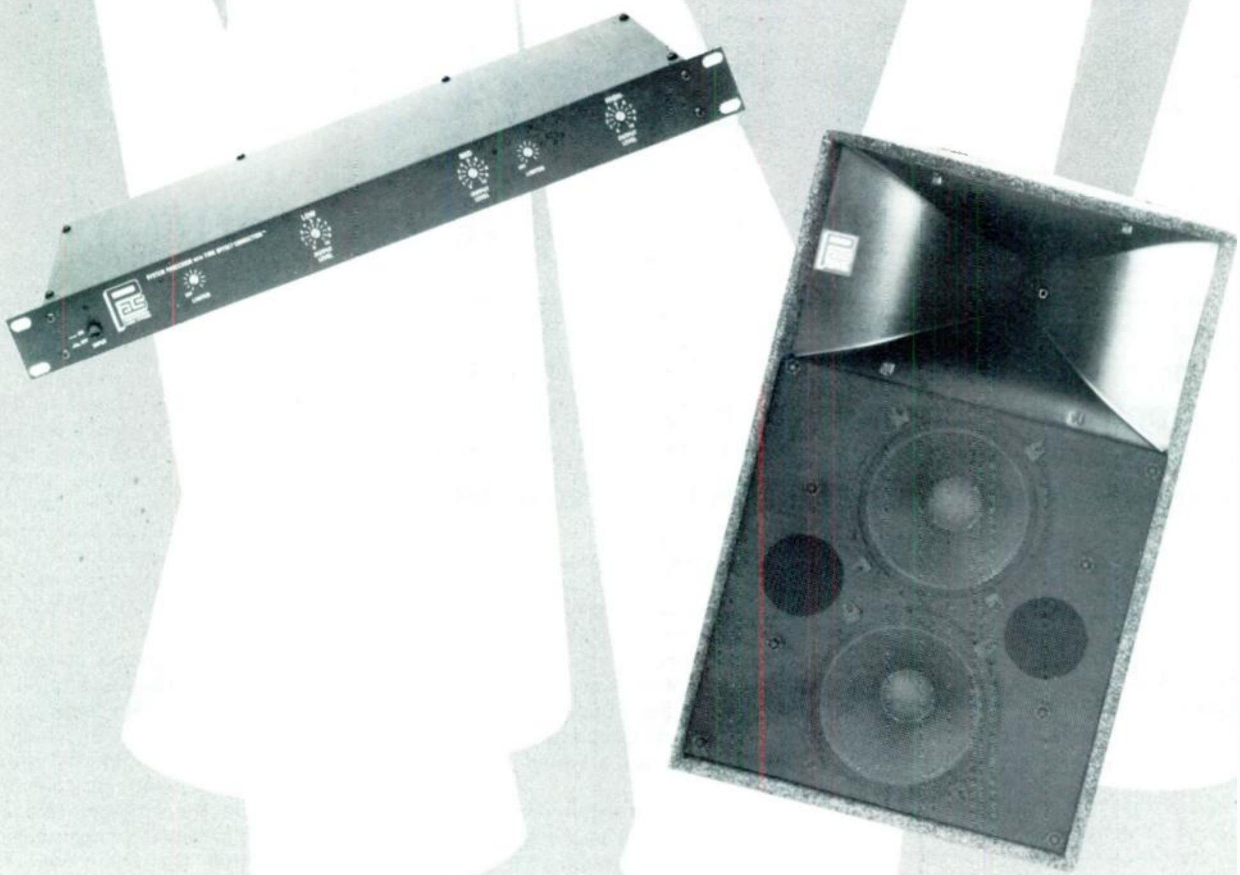
What this means is that the maximum MIDI pitch-bend amount will be interpreted as a different interval on each MIDI instrument, depending on how many semitones the maximum bend interval is set to on that keyboard. It's for this reason that, next to the O which means that pitch-bend data is recognized, the maximum musical range will be shown. In our example, the 0-12 semi means that this instrument can bend up or down one octave, or 12 semi-tones. It's up you go to adjust the pitch-bend ranges of your various MIDI instruments (where possible) so that bends all operate over the same range – unless you want the weird effect that results when different instruments are bent by differing amounts simultaneously.

For reasons of hardware, many sampling machines can only bend over a four semitone range, so you may find you have to bring synths down from their one-octave range in order to match them to a sampler.

Moving back to the chart, the resolution shown in the remarks column is rarely crucial as it merely refers to how finely the bend is analyzed. Don't worry if your instruments have different resolutions. A nine-bit system can communicate perfectly well with a seven-bit resolution. True, one will bend in finer steps than the other, but this is unlikely to show unless the discrepancy is a large one.

Next time, we'll be looking at the various MIDI Control Changes (both continuous controllers and switches) which have developed into conventions. ■

The MRS-2
With Time Offset Correction (TOC™)



The **MRS-2** sets the stage with a **Dedicated System Processor** that is in complete control of the performance. What is **Time Offset Correction? TOC™** corrects for the distance between the low frequency components and the **Constant Coverage** mid/high frequency horn; resulting in ultra low distortion sound reproduction.

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Digidesign SoftSynth & Burner

Software for the Apple Macintosh

Digidesign's Sound Designer program gave Macintosh owners the chance to have a good look at their samples. Their two latest programs, SoftSynth and Burner, present new ways to create samples and burning sample data into EPROMs for use in drum machines. *By Rick Davies.*

SoftSynth

THE SOFTSYNTH PROGRAM IS A CLEVER WAY of getting a sampler to not sound like a sampler. To the end, a sampler will always operate like a sampler, churning out sample data at various speeds with usual 'chipmonk' effects when samples are played at pitches much higher than the original sample, and the 'grunge' of clock noise when samples are played far below.

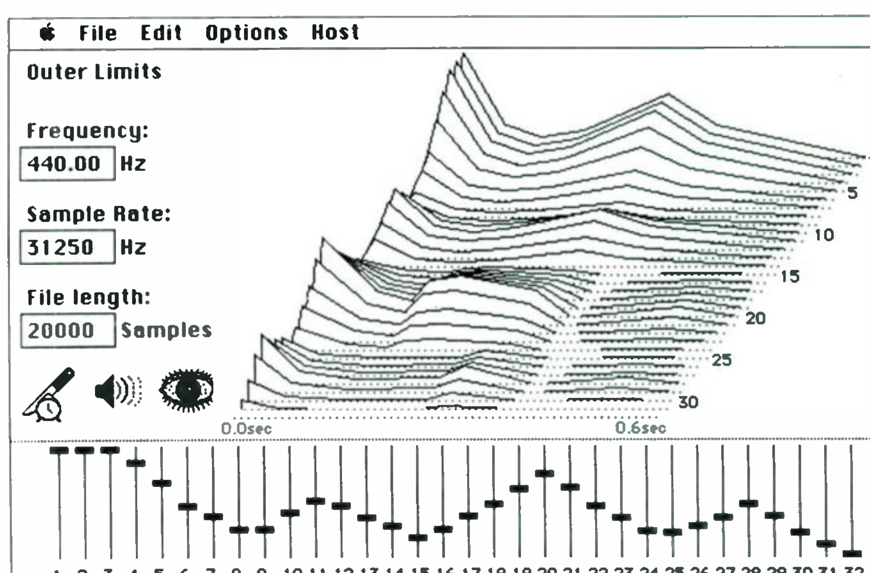
But at least you can do something about the sample source material. It may appear that samplers are limited to manipulating 'real' sounds, but SoftSynth proves that a personal computer-based system can be used to circumvent the sampling process altogether, and produce truly original sounds.

What SoftSynth does is *create* samples on the Macintosh, rather than merely edit samples which come from the 'real world'. These sounds are created by additive synthesis, meaning that it is possible to control the mix of 32 harmonics through the duration of a sample. If this were to be implemented in analog hardware, it would be the equivalent of having 32 oscillators, each with its own complex envelope. Even if digitally-generated oscillators were used, the hardware required for such a synthesizer would still price it far beyond most musicians' reach.

Instead, SoftSynth analyzes the harmonics you tailor with Mac's high-resolution graphics, then calculates a sample which can be downloaded by sampler for playback. Although it takes a while for the sample to be generated, this is a practical compromise. A number of samplers implementing the MIDI Common Sample Dump Standard can receive a SoftSynth sample dump, or you can choose to create a Sound Designer-compatible disk file.

Since the Mac is not required to spend all of its time generating sound as the SoftSynth controls are adjusted, the controls themselves are easy to use and take advantage of the Mac's graphics whenever possible. The main programming screen shows a frequency analysis of the current sound, with all 32 harmonics clearly displayed. Below the frequency analysis are 32 'faders' corresponding to the harmonics. If you want to change the harmonic mix, simply 'grab' the desired fader with the Mac's mouse, then move it. The frequency analysis doesn't respond to such changes instantly, though. To update the display, grab the eye icon, then after a moment, the display features the edited harmonics.

The frequency analysis shows the way each harmonic's amplitude varies in time, but there's more to each harmonic than can be seen from this screen. Grabbing the number of the desired partial from below the faders takes you into Single Partial editing mode. The screen shows a section of the selected harmonic's amplitude envelope, and below that, its pitch envelope. Yes, each



SoftSynth: main programming screen

harmonic can be detuned by a complex envelope – and by an initial amount as well as by the pitch envelope.

The envelopes themselves are certainly more flexible and easier to adjust than those provided on most synthesizers. Each can have as many as 40 linear segments, and is shaped by simply grabbing any segment with the mouse, then stretching it into the desired shape. The usual Macintosh mouse-activated scrolling controls allow you to look closely at selected portions of the envelope.

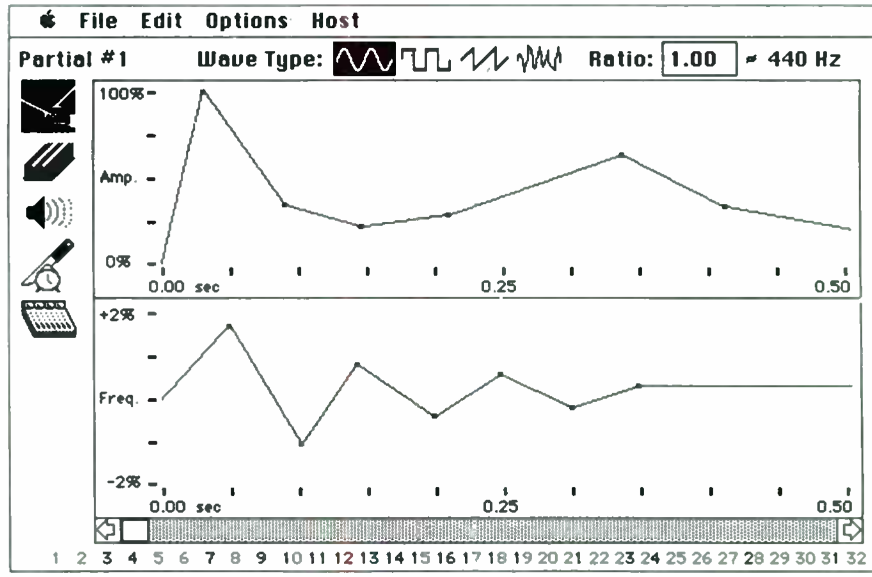
A feature which I feel has been lacking on many instruments for some time now is 'envelope copying', which avoids the drudgery of creating envelopes from scratch every time you build a new sound. Needless to say, this ability is not missing here, which is a good thing when you consider the complexity of SoftSynth's envelopes. Also, since each envelope is easily edited, it's an easy matter to generate 32 variations on one envelope for the harmonics, examine the frequency analysis as you audition the sound, then continue editing as you deem necessary. Incidentally, you can listen to each harmonic individually if you want to.

As if these features weren't enough, each harmonic can be a sine, sawtooth, or square waveform, or any of three band-limited noise waveforms. Sure, this is a deviation from standard additive synthesis which is based on Fourier analysis, and hence, on the combination of sine waves – but so what? The band-limited noise is a nice alternative to the white noise usually found on analog synths, too.

With an understanding of Single Partial mode, a whole new realm of sound is open for exploration, but that's hardly where SoftSynth quits. An unusual icon showing a knife cutting into an alarm clock is the gateway into Time Slice editing mode, in which all 32 harmonics are adjusted simultaneously instead of one at a time.

The Time Slice editing screen shows a master envelope which controls the amplitude of the sample, while numbered rectangles (representing timbres) are displayed along the envelope's time axis, coinciding with break points in each harmonic's amplitude envelope. Since there are 40 potential break points in each harmonic's envelope, and there are 32 harmonics, there could be a lot of timbre break points appearing in Time Slice mode. To avoid confusion, SoftSynth eliminates the least significant timbre break points, and displays only the most significant ones.

These timbres can then be moved around, or copied to other parts of the sample by grabbing the desired timbre with the mouse, then dragging it to another point on the master envelope. This is a 'brute force' method compared to Single Partial editing, but it's extremely powerful.



SoftSynth: single partial programming screen

As it turns out, these two editing modes are actually interactive. That is, you can switch between Single Partial and Time Slice editing modes at will. A big plus.

The range of sounds SoftSynth can create is pretty much limited by the amount of time you make available, and as with any synth, it's useful to have some presets to use as starting points while you learn how the system works. So, SoftSynth comes with a palette of 15 (or so) preset harmonic files, so you can start with a piano or brass sound, then alter it as you please.

One feature which had not yet been implemented, let alone announced, at the time we examined SoftSynth is the 'random' sound generator. Such features are becoming more popular these days. Hybrid Arts' DX Droid and Sequential's Prophet VS both feature similar functions, but there is little resemblance between the results of these three systems. Apparently, SoftSynth will let you set limits on certain parameters, so that you can loosely define the type of sound you want to produce. SoftSynth then generates random harmonics based on these restraints, and it'll be interesting to hear the range of unusual sounds that this feature will no doubt produce.

The final stage is loading the synthesized sound into a sampler. The 'Host' (or 'Sampler') menu lets you set up the Mac for the MIDI interface in use. Following that, the 'File to Synth' function (selected from the 'File' menu) takes care of sending the sample data to the MIDI–equipped sampler in question. The sampler can now play the sound no differently than it would had you instead sampled it from some far more expensive instrument. The big difference is that you made up the sound yourself, and that the SoftSynth program costs only $295.

Summing up, it looks like SoftSynth could be the dream of many synthesists. The controls are there for endless tweaking of sounds, and if inspiration or your patience wanes, you can always resort to generating random samples. Your sampler will never know the difference.

Burner

WHEN LINN INTRODUCED THE LM1 in the early 80s, everyone was impressed by the quality of the eight–bit drum samples and the ease with which patterns could be assembled. Soon enough, a number of similar drum machines appeared and proved that there was a huge market for drum machines if the price was right. Since these drum machines had their sampled drum sounds on EPROMs which were, in general, easy to replace, there was soon a demand for alternate sounds. If the drum machine's manufacturer didn't offer alternate sounds, then an independent company would.

Many moons later, samplers have started to take over the drum machines' role in the recording process. The drum machines may be recorded onto tape to begin with, but often each sound is recorded on a separate channel so that later, it can trigger a sampler and be replaced by another sound. These days, there appears to be little time to have your samples converted into drum chips.

Which is too bad really, since EPROMs don't lose memory when power is switched off, and are also a good deal more roadworthy than the 3½" disk drives currently ▶

featured on many samplers. Digidesign (who started off as DigiDrums, by the way, producing alternate drum chip sets for various drum machines) has recognized the number of drum machine owners who are trying to keep their sounds up to date, and come up with Burner, a hardware/software package for the Apple Macintosh (and Macintosh +) which 'burns' EPROMs for a variety of drum machines, using Sound Designer sound files. It does require owning Sound Designer, or at least obtaining Sound Designer files, but Digidesign appear to have already taken that into consideration. Anyhow, before we get into obtaining sound files, a look at Burner's operation is in order.

The Burner package consists of a program disk and an EPROM burner which can be used in other applications as well. There is only one screen in the Burner program, and burning an EPROM is easy.

First, to make sure that the EPROM you are about to burn has been erased, insert the EPROM into the ZIF (Zero Insertion Force) socket on the burner itself, then select the 'Blank?' utility. If the EPROM fails the test, it must be erased with an ultra–violet EPROM eraser, then tested again. Once that's out of the way, the burning process continues: load in a sample file on the Mac, select the type of EPROM, the drum machine for which it is intended, the number of 'split' in the chip or in the sound, then select the 'Burn' utility and wait for the hourglass display to 'run out'. Each of these functions, incidentally, is carried out using the Mac's mouse. It's then recommended that you verify the EPROM has been successfully burned, and a utility exists for this as well.

The sound files themselves are stored in 16–bit format, so Burner converts the data into whichever format is required (8–bit linear, 8–bit companding, or whatever) by the drum machine in question. Currently, Burner makes EPROMs for the Linn LM1, Linn 9000, Oberheim DMX, DX, DXa/Stretch, Sequential Drumtraks, Simmons SDS1. SDS7 and SDS9, and the cymbal chip for the E–mu Drumulator. It's unfortunate that it doesn't burn the other Drumulator chips, but this is apparently due to the complexity of the 'header' which must be included with the sample data on the EPROM in order for the Drumulator to know where to find each sample.

So what happens if you don't happen to own the Sound Designer program, but want to get sound files for Burner? Well, it appears that Digidesign will make sound files available by modem through their BBS, and emphasize that since most percussion sounds are relatively short, sound files could easily be transfered over networks such as PAN.

Burner retails for $595, including the EPROM burner, and I can certainly see it receiving a warm welcome form many musicians who will be happy to find that the technology which at one time appeared to have left their drum machines behind has come back to give them a lift.

MORE FROM Digidesign Inc, 920 Commercial, Palo Alto, CA 94303
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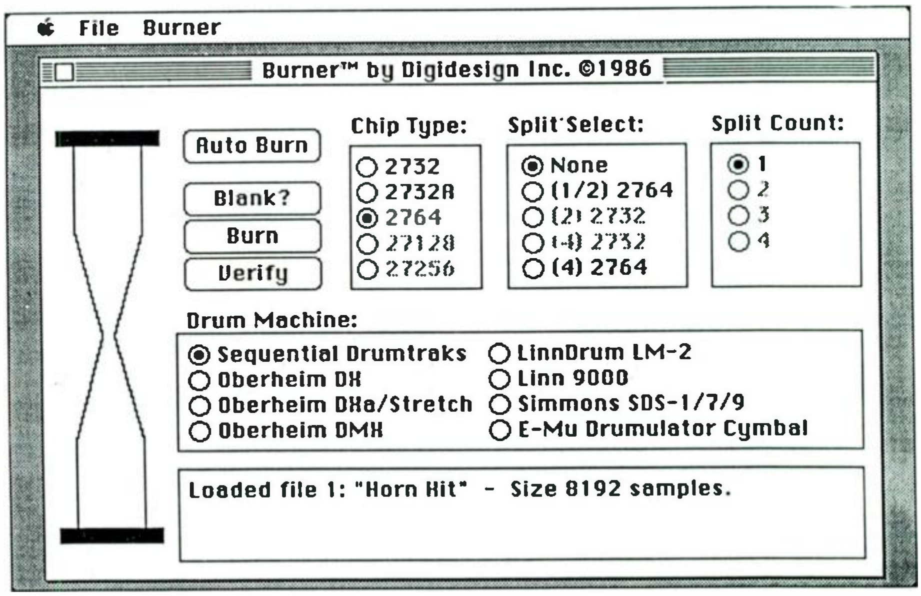


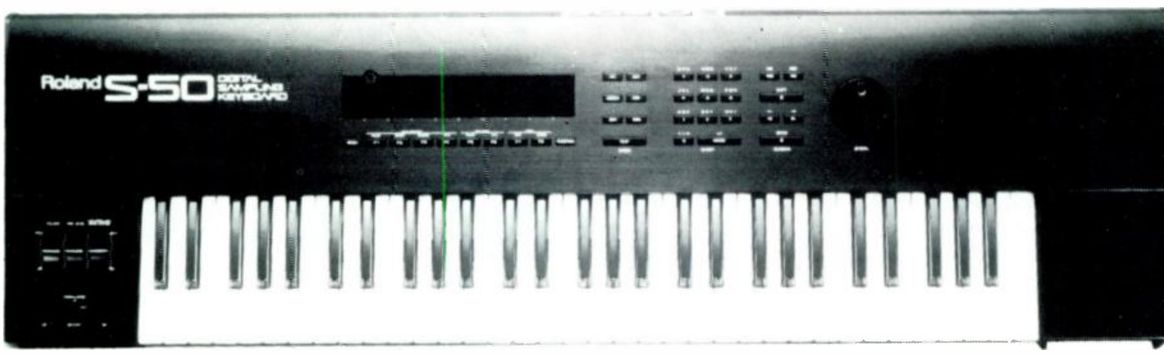
Burner: main programming screen

An Introduction to...

SAMPLING

S A M P L I N G

Part 2: The Mechanics of Sampling



Following on from last month's introduction, we take a closer look at how sampling works and some of the things you need to decide before you actually make a sample. *By Paul Wiffen.*

IT IS POSSIBLE TO BE REASONABLY PROFICIENT at sampling without knowing anything about what actually goes on inside your sampler. But understanding the process can only help in recognizing problems when they occur, and knowing the sort of things you can try to get around those problems.

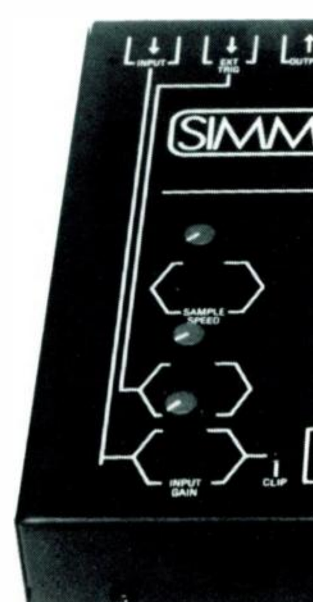
Before we do anything else, there is one point that should be made about sampling. It is nothing more or less than recording. You need to have some of the same skills as the recording engineer – setting the right input level (too low and you get a poor signal-to-noise ratio, too high and distortion sets in), picking the right microphone to use with acoustic instruments, using EQ to improve sounds, and so on. You need to turn your ear to listen to sounds more closely, not only once you've sampled them, but before you begin sampling.

For these reasons, you'll find your sampling should benefit from reading the practical recording features we run from time to time.

Anyhow, let's begin by looking at what happens when you press the Record button on your sampler. The incoming signal (ie. the electrical signal in the cable from mic or instrument) is fed through an analog-to-digital converter, known in the trade as an ADC. This works by analyzing the incoming sound thousands of times a second, and converting that analysis into digital form, so that it can be stored in digital memory (known as RAM – random access memory). This means that when we want to hear our sample back, the numbers are simply read out of the RAM through the opposite of an ADC, a digital-to-analog converter (abbreviated, you'll be surprised to hear, to DAC), which turns the number back into a standard audio signal and then feeds it into the sampler's output. In practise things aren't quite that simple. There are a few additional problems that need sorting out, but we'll leave them 'til later.

First, let's look more closely at the two main factors which govern the quality of our sample – the accuracy of the analysis and how often it happens. The analysis performed by the ADC looks at the energy present in the input signal (ie. the energy present in the sound wave once every so many microseconds) and converts that into a binary number. The fidelity depends on how many microseconds there are between samples (the less the time, the higher the quality) and the resolution of the binary number (the more digits in the number, the closer to the original the result is).

Taking the second of these factors first, we refer to binary numbers as eight-bit, 12-bit, 16-bit and so on. These figures refer to the number of digits in the binary number: when a number is stored in digital memory it uses one bit of memory to store each digit, hence the 'bit' in eight-bit. Binary numbers are just strings of 0s and 1s, which is how they can be represented in digital form as either On or Off, by a process akin to the action of a switch. The binary system prevents you counting beyond 1, so to represent denary (the system we normally use) values greater than 1, you have to string them together 0s and 1s together. 10 in binary means 2, 11 is 3, 100 is 4, and so on. Each time you add another zero, you double the denary number, so by the time we get to an eight-digit number (10110101, say), the maximum value we can have is 255



(11111111). If we include 0 (or 00000000 in eight-bit binary), then we have 256 values possible. So an eight-bit sample breaks down the electrical representation of the sound wave into 128 different levels.

If we move on to 12-bit (110110000110, for example) resolution, the maximum number (111111111111) is 4095 in denary (4096 if you count zero as a value). This means that 12-bit resolution is 16 times more accurate than eight-bit (256 x 16 = 4096), not the 1½ times that you might assume by looking at the ratio between eight and 12.

It's primarily 12-bit sampling systems that are being released at the moment (the Prophet 2000, Akai S900, Korg DSS1, Roland S10 and S50 are all 12-bit systems). More expensive systems like the Series III Fairlight, Synclavier and PPG are already using 16-bit systems, but for the time being this sort of technology can't be bought for less than $10,000 – though take a look at this issue's news story on the Hybrid Arts/Nilford Laboratories ADAP system, which uses the Atari ST computer to make 16-bit sampling technology more accessible. Working with 16-bit systems, by the way, is 16 times again more accurate than 12-bit, with a mind-boggling 65,536 different levels of analysis possible. (For reference, Compact Disc uses 16-bit analysis.)

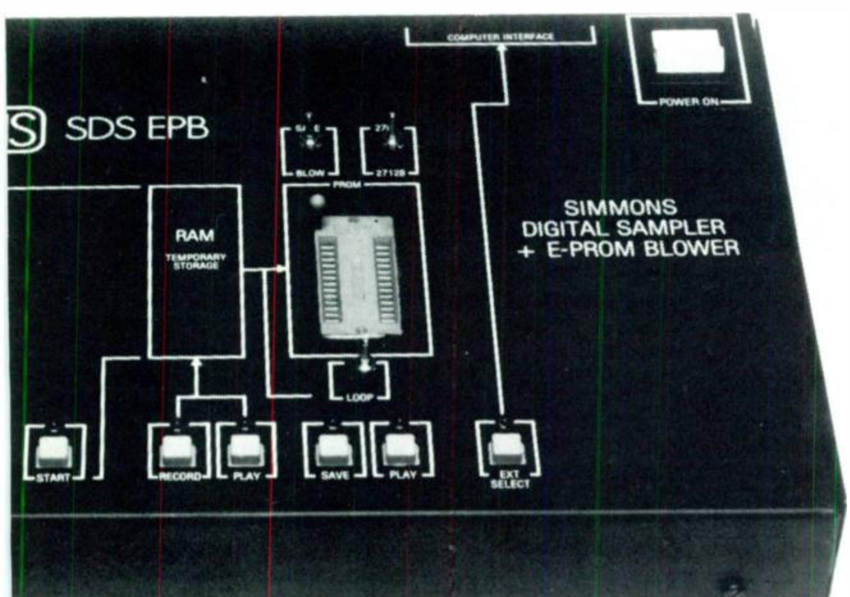
Now, in addition to the straight analysis method in which the range of the waveform is divided into 256 (eight-bit), 4096 (12-bit) or even 65,536 (16-bit) equal levels, there are also several proprietry methods which some designers use to squeeze more fidelity out of the same analysis. These work by compressing the data before storing it, and then expanding it before replaying. In sampling, this technique is known as companding (a term originally used in the recording studio, where audio signals are put through this compression/expansion process to achieve better noise levels on tape).

We use the terms 'linear' (for the straightforward analysis) and 'companded' to differentiate between two systems which use the same bit analysis. And in practise, the E-mu eight-bit companded system is more comparable to 12-bit linear systems on than the eight-bit linear format of the Mirage.

The other crucial factor is how often this analysis is made – in other words, how fast the sampling is done. This is known as the sample rate. Typical samples these days range from 8kHz up to 50kHz. This 'kHz' refers to the measure Kilohertz, which stands for 1000 cycles a second. So 50kHz means 50,000 times a second, or one sample every 20 microseconds (a microsecond is 1/1,000,000 of a second). This sort of rate gives you fairly high quality; Compact Disc, for example, uses 44.1kHz (44,100 times a second). The current 'standard' rate seems to be 31.25kHz, which is available on the Prophet, Korg and Roland samplers. This gives fairly reasonable samples, provided the original signal doesn't contain too much top end.

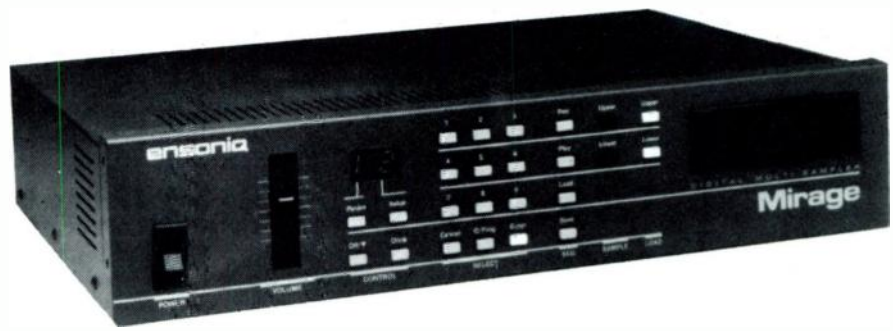
Things like cymbals, hi-hats and other bright sounds really need a higher rate, like the 42kHz the Prophet offers. The Akai S900 also allows you to sample at this rate, but you wouldn't know it to look at the kHz figures the machine's display gives you. This is because the S900 doesn't show the sample rate you're using, but the resulting audio bandwidth, which, just to make things complicated, is also measured in kHz. This refers to the highest audio frequencies which can be sampled, and at 42kHz sample rate, the S900 truthfully shows this to be 16kHz. This is by no means the highest frequency the ear of a 13-year-old can pick up (this is sometimes in excess of 20kHz), but is probably very close to the limit of the hearing range of anyone who has been working in the rock 'n' roll business for more than a couple of years!

Other machines are often quoted as having an audio bandwidth of half their sample rate, but this is misleading. The theoretical maximum bandwidth (according to a gentleman named Nyquist, after whom the theorem is named) *is* half the sample rate. In order to register the presence of a frequency, at least two samples must be made in the course of one waveform or oscillation, ie. at twice the speed (or rate, or ►



frequency) of that oscillation. The reason for this is that, to register the presence of an oscillation, you need to capture at least one point in its positive phase and one point in the negative part of its cycle. But unless you happen to catch both the maximum negative and the maximum positive value, you won't have anything like a true register of its amplitude (volume) or harmonic content. So for most purposes, you need to allow more than two samples per oscillation, ie. a bandwidth of rather less than half the sample rate.

There is another problem – and solution – which also makes Akai's thruthfulness refreshing. Frequencies above or just below the theoretical limit of half the sample rate (which are normally inaudible because they're too high), while not faithfully recorded, do get picked up partially and then get misinterpreted as lower frequencies. The closest analogy is that of wagon wheels or aircraft propellors in the movies. Because they are going faster than the speed of the camera (which is at 25 frames per second, say), they appear to be going backwards or more slowly (ie. at a lower frequency) when in fact they may be moving two, four or even 20 times faster. Aliasing, the name we give to this when it happens to sound, is the audible equivalent of the same effect: frequencies much higher than the audio bandwidth are registered as frequencies within that audible range.



The second difference between your synth filter and the anti-aliasing version is that there's no messing around with 12dB- or 24dB-per-octave roll-offs. The typical anti-aliasing filter is quoted as being in excess of 100dB-per-octave, which simply removes all frequencies about the cutoff point – a very sharp cutoff instead of the smoother ones you get from synth filters.

Even with all this heavyweight filtering going, on you may still find some aliasing showing up, especially when you transpose your sample down. You can usually use the synth filter that most samplers offer to cut out the problem, but you may find you begin to lose the brightness of your sample. Still, using keyboard-tracking with the filter, you can usually get reasonably acceptable results, and there are also a few pre-sampling tricks you can use if the problem is really bothering you (see this month's 'Toward More Creative Sampling' feature for more details).

There are two other types of distortion you may run across when sampling, and these are less easy to cure as they may well be inherent in the design of your sampling system. This is particularly true of quantization noise, which is the unfortunate by-product of the steps that the digital recording process uses. Because the signal is turned into a series of discrete levels instead of the smooth curve it was originally in its analog form, unwanted frequencies (usually higher up the scale) are generated in an unpredictable fashion, which gives a noise component to the signal. The smaller the steps (ie. the higher the bit analysis), the less noticeable and the higher pitched the noise effect is. This is why 16-bit and 12-bit systems are comparatively free of quantization noise while eight-bit formats are renowned for it. Still, many people love the crunchiness that eight-bit systems can give to drum sounds, so it has its uses.

Again, subtle filtering can help to reduce the effect, but if it starts to really annoy you, the best (long-term) answer is to look at upgrading to a higher bit system.

The other main drawback of sampling can be easily gotten around on some machines, while being unavoidable on others. This problem is clock noise, and it's an audible result of the sample rate, or the speed at which you make samples. If you make samples at rates which are in the audio range (less than 20kHz, for argument's sake), then the actual workings of the playback rate will probably end up sounding like a frequency in the sample. Things get better when you move up out of the audio range with your sample rate, but by-products can still occasionally be heard.

The answer here is usually to up the sample rate. On some machines this is easily done, while on others it is just not possible. If you have a sampler that falls into the latter category and you don't like the sound of clock noise, it's time to look around again for a more flexible sampler. One way of keeping clock noise (and even aliasing) to a minimum while squeezing maximum audio bandwidth and fidelity out of your system, is to use sample rates which are multiples of the fundamental frequency you're sampling.

On samplers that offer a limited flexibility of sampling rate, you'll be limited to sampling only certain notes. With others that offer a wide range, you can set the rate to match the pitch of your samplem, so long as you know which multiples to use. Some manufacturers, like Ensoniq in their 'Advanced Sampling Guide' manual, actually give recommended rates for sampling specific pitches, and the old Fairlight manual is another example of this. If more manufactureres made such details available, then higher-fidelity samplers like the Akai S900 (if only it told you sample rate as bandwidth...) could be made even better. A future article will go into the fine details of this process.

One final area of possible confusion that needs clearing up is the use of the letter 'K' after numbers. Not only is it an abbreviation for the sample rate (42K for example) or its associated audio bandwidth (16K) instead of the correct kHz or Kilohertz, it's also used to refer to the amount of memory a sampler has, or which a sample occupies. One reason all samplers don't just sample at a rate of 50kHz and be done with is that this uses up memory at an alarming rate. So unless you have the vast memory of a Fairlight, Synclavier, PPG HDU or CD available, you have to compromise by using slower sample rates.

The K in 256K of memory refers to the number of 'words' the sampler can store in thousands (or near enough; 64K actually means 65,536 but don't let that confuse you – 1K of memory equals 1024 'words'). A word is the length of binary code used for each individual sample, so eight digits form an eight-bit word, 12 form a 12-bit one, and so on. So, the correct way to refer to the Mirage sample memory is 128K of eight-bit words, while the Akai S900 has 512K of 12-bit words.

Memory is used up according to the sample rate being used, so the Prophet 2000 sampling at 31.25kHz would use up approximately 32K of memory in one second (using 12-bit words). The resulting audio bandwidth will probably be around 13kHz.

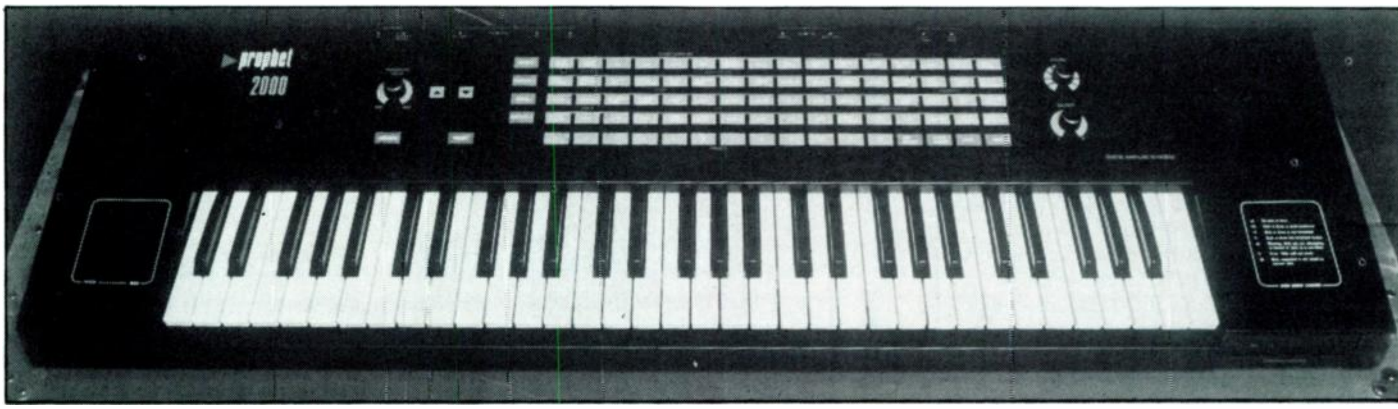
To help you decide what he means when the salesman is throwing K figures at you, bear in mind sample memory is usually expressed in multiples of 64 (128K, 256K, 512K and so on), while sample *rates* range from 8 to 10kHz on very cheap units, and from 10kHz all the way up to 50kHz in the $1,000-plus price range. 100kHz sample rates tend to be only available when you pass $50,000, though who knows what may turn up next year...? And finally, audio bandwidth is *always* less than half the sampling rate, and often considerably less. Don't let that salesman tell you otherwise. ■

Getting the Most From...

MONO MODE

M I D I

Part 2: Prophet 2000/2002



Having looked at an application of MIDI Mode 4 on the Casio CZ synths in our first issue, we now move into the field of sampling and look at how Mono Mode can be used to sequence groups of different samples. *Text by Paul Wiffen.*

ONE OF THE REASONS that more expensive systems like the Fairlight, Emulator, Synclavier and PPG have proved so popular in recent years is that they're all able to sequence various different samples simultaneously. Until this year, such multitimbral sample sequencing was only possible if you had more than $5000 to spend. But with the new wave of cheaper 12-bit samplers, it was inevitable that at least one or two manufacturers would make their machines able to replay several samples simultaneously, even if they had no onboard sequencing capability.

The way they found to do this was via MIDI Mode 4, where each voice in the machine is set to look at a different MIDI channel, according to the 1.0 MIDI specification. Sequential, the first company to use Mode 4 on a sampler, have actually gone beyond the essentially monophonic assignment which Mono Mode implies. Instead of tying a particular voice to a channel, they hit upon the idea of assigning each sample to its own MIDI channel.

There are several advantages to this. First, in a general sense, it is actually the sounds you want to treat as separate entities on different tracks of your sequencer. As all the voice channels in your sampler are (theoretically, at least) the same, it doesn't matter which D-to-A converter is used to play the sample track. If this is the case, then a second advantage with a machine like the Prophet 2000 with its fast dynamic allocation – is that it can use a voice channel which is (temporarily) not sounding one sample to play another. This means it can go beyond the limitations of having one voice channel per sample (with monophonic playback) that machines like the Fairlight and Synclavier have imposed until their most recent updates.

Because the Prophet 2000 (and 2002) can be accessed polyphonically in Mode 4, it can receive up to the full polyphonic capability of the machine on each channel, ie. for each sample. This *doesn't* mean it can play eight notes with all 16 possible samples simultaneously, though. If six voices are being triggered on one MIDI channel, then only two are left for the next sample to play.

However, as soon as a note is finished playing with one sound, it is free to play another sample. So the Prophet can be playing an eight-note piano sound one second, and the next, have all eight voices instantly available to play brass or even drum sounds. In fact, it would be possible to cycle through all of the 16 possible samples one after the other, playing eight notes with one sound, then immediately switching to the next and so on. This is known as pseudo-128 voice operation – in other words it is possible to give the illusion of there being 128 voice channels (16 x 8), using just eight.

So how do we go about achieving all this? First of all, we need to get 16 samples into the Prophet. If you have the basic model 2000 or 2002, then you have 256K of sample memory. Split between 16 samples, that's 16K each. At the middle sampling rate (31kHz), that means you have on average half a second per sample. While this is fine for most percussive sounds, you may find it restricting for things like piano, strings and brass.

One solution, of course, is to get the memory expansion up to 512K, which gives you more like a second's worth of average sample time when split between 16 samples. But the whole point of using Mode 4 is to be able to sequence different types of sound. You'll probably have some very short samples as well as long ones, and if you look carefully at the end points of some of the shorter sounds (like bass drums, closed hi-hats, rimshots) you should find they last a good deal less than half a second. So, by using the Recover Memory function, you may well be able to free up a fair chunk of memory for your longer samples.

Diagram 1: Recording with just sequencer and 2000

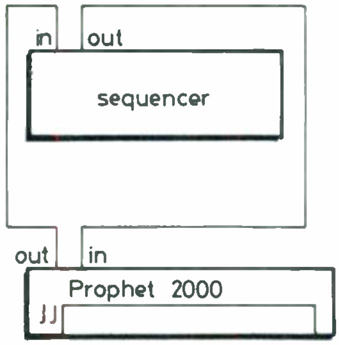


Diagram 2a: Recording with another MIDI keyboard, 2000 or 2002 and sequencer with MIDI Out/Thru Mix

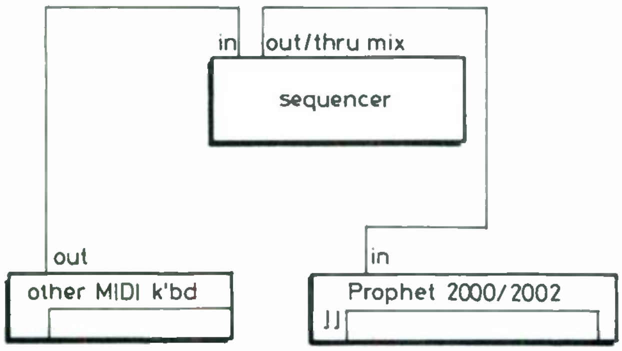
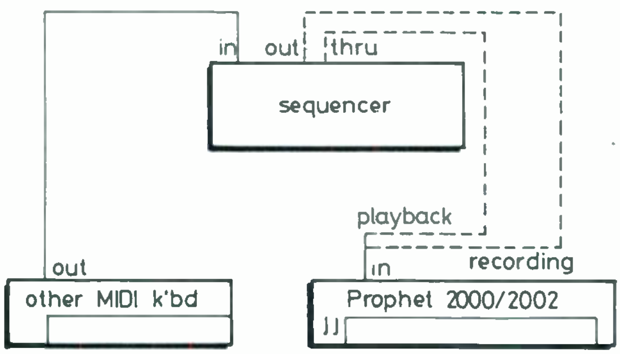


Diagram 2b: Recording with another MIDI keyboard, 2000 or 2002 and sequencer without **MIDI Out/Thru Mix**



With instruments which cover more than the three-octave range you want to sequence, you'll have to sequence different MIDI channels. You might, for example, put the bass end of a piano on sample number 5 and the top end on 6. On the whole, though, it's usually best to to make a 'one-size-fits-all' sample which covers the range you want to sequence. If you have trouble doing this, take another look at our 'Toward More Creative Sampling' piece in the last issue.

Try to get a good, contrasting range of sounds. Start with some basic drum sounds – kick, snare and toms, perhaps a closed hi-hat – unless you plan to sequence the Prophet alongside a drum machine. Then you might want a bass sound of some sort; either a real bass guitar (slapped samples are still fashionable this year - see Madonna's 'Papa Don't Preach') or a synth bass sound. Then maybe a guitar sample for rhythm purposes - either a muted string pluck á la Nile Rodgers, or a powerchord, depending on whether you're sequencing dance music or heavy metal. Now the choice becomes broader: you might like to use traditional sounds like piano, strings and/or brass, or more esoteric sounds like sampled wine glasses and bottles. The choice is yours.

Once you have all these samples in the memory of the machine at once (use the Load One Sound parameter if you're compiling them from other disks), then what do you have to do to map the sounds ready for use in Mono Mode? The answer is quite simply, nothing! This is the beauty of the Prophet's Mode 4 implementation: you don't need to go through a lot of messing around with maps and keyboard assignments. When you switch to Mode 4 on the front panel (by going to Mode in the second line of parameters – Digital 1 – and turning the parameter knob 'til '4' or '4.' appears in the display), each sample is automatically assigned to its own MIDI channel. For instance, the sample in Sound number 5 is automatically looking at MIDI channel 5, and the same is true for all the other samples and their respective MIDI channels. In other words, once you've got the samples into the 2000 (either from disk or sampling) and switched Mode 4 on, you're ready to begin sequencing.

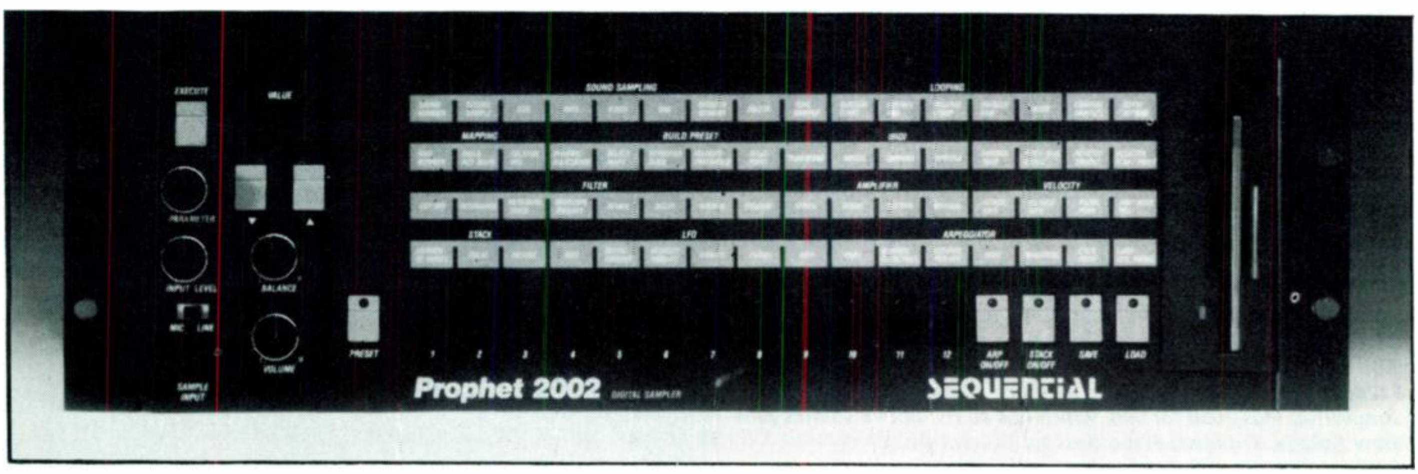
This is most easily achieved using another MIDI keyboard which can send on all 16 MIDI channels (and *not* an unmodified DX7) or using a sequencer that can assign MIDI channels to different tracks (if a DX7 is all that's available); you simply record each track using a different MIDI channel, and that's all there is to it.

The original sampled pitch of each sound will always be assessed on C3 (MIDI note number 60) so you may need to retune if you didn't sample a C, but otherwise you should be able to fly through sequencing the parts.

You'll need to be careful you don't exceed more than eight notes sounding at any point in your piece, because any extra will either not get played or 'steal' from voices already sounding. But you'll be amazed how full-sounding a piece of music you can create, providing you don't insist on having a five-note string chord behind everything.

If you don't own another MIDI keyboard (and if you're a 2002 owner, you may be in trouble there), you'll either have to sequence the 2000 in step-time, or follow a Mode-switching procedure as outlined below.

Mode switching is necessary because the Prophet 2000 doesn't transmit what is played over the keyboard when in Mode 4. So to record a sequence, you need to switch to either Mode 1 or Mode 3 (Poly Mode with Omni On or Off respectively) to be able to record each track into your sequencer. You'll also need to check you're ▶



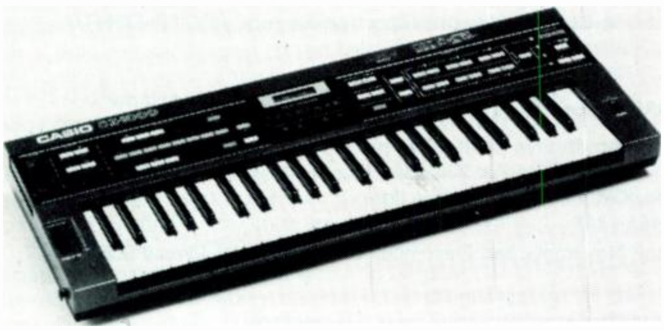
PATCHWORK

PROGRAMMING

The page where MT's writers let go of the reins and leave it to readers to contribute their own synth patches. Here's the first selecton from the sounds we've received from readers so far....

Casio CZ101/1000
Module Two
R Leigh, Oakland CA

Christened 'the perfect CZ sound' by its creator (the things people will do for a free sub), 'Module Two' is the result of manipulating two 1+1 synth string sounds, merging them together into one patch, and fiddling with the Detune and Vibrato parameters until a strong Carillon-type of sound emerges. As usual, the patch lends itself to experimentation, and the programmer suggests you 'detune by two octaves for a powerful synth strings effect'. All in all, a distinctive, versatile sound.



TONE NAME	CARTRIDGE NO.	TONE NO.
MODULE TWO		

PARAMETER

LINE SELECT	MODULATION RING	MODULATION NOISE	DETUNE +/−	DETUNE OCTAVE	DETUNE NOTE	DETUNE FINE	VIBRATO WAVE	VIBRATO DELAY	VIBRATO RATE	VIBRATO DEPTH	OCTAVE +/−	OCTAVE RANGE
1+2'	OFF	OFF	+	2	00	09	1	20	51	18		0
(1,2,1+2',1+1')	(ON/OFF)		(+/−)	(0−3)	(0−11)	(0−60)	(1−4)	(0−99)	(0−99)	(0−99)	(+/−)	(0−1)

1

DCO 1

WAVE FORM FIRST	WAVE FORM SECOND
5	2
(1−8)	(0−8)

ENV (PITCH)

STEP	1	2	3	4	5	6	7	8	
RATE	00								(0−99)
LEVEL	00								(0−99)
SUS/END	END								

DCW 1

KEY FOLLOW	
2	(0−9)

ENV (WAVE)

STEP	1	2	3	4	5	6	7	8	
RATE	99	40							(0−99)
LEVEL	99	00							(0−99)
SUS/END	SUS	END							

DCA 1

KEY FOLLOW	
2	(0−9)

ENV (AMP)

STEP	1	2	3	4	5	6	7	8	
RATE	72	38							(0−99)
LEVEL	92	00							(0−99)
SUS/END	SUS	END							

2

DCO 2

WAVE FORM FIRST	WAVE FORM SECOND
2	1
(1−8)	(0−8)

ENV (PITCH)

STEP	1	2	3	4	5	6	7	8	
RATE	99	99							(0−99)
LEVEL	66	00							(0−99)
SUS/END		END							

DCW 2

KEY FOLLOW	
0	(0−9)

ENV (WAVE)

STEP	1	2	3	4	5	6	7	8	
RATE	99	00							(0−99)
LEVEL	99	00							(0−99)
SUS/END	SUS	END							

DCA 2

KEY FOLLOW	
0	(0−9)

ENV (AMP)

STEP	1	2	3	4	5	6	7	8	
RATE	92	95	92	99	32				(0−99)
LEVEL	99	87	99	88	00				(0−99)
SUS/END				SUS	END				

Oberheim Matrix 6
Phaedra
Gianni Godino, Atlanta

Nice to see the excellent Matrix range getting a look in. 'Phaedra' is 'dedicated to all Tangerine Dream fans everywhere', according to its creator, but before the rest of you reach for the hair gel, it's not so cosmic as to be confined entirely to that genre. The patch is well suited to atmospheric chordal work, and a little subtle variation can be introduced by reducing the amount of frequency modulation slightly (parameters 22 and 23). A good playing technique will make the most of the velocity sensitivity that's programmed. ■

	0	1	2	3	4	5	6	7	8	9
00 DCO1	Freq	Fr/Lf1	Sync	Pw	PW/Lf2	Wave	Wsel	Levers	Keybd	Click
	12	0	0	31	+63	63	PULS	BOTH	KEYB	OFF
10 DCO2	Freq	Fr/Lf1	Detune	Pw	PW/Lf2	Wave	Wsel	Levers	Keybd	Click
	12	0	+10	31	+63	31	PULS	BOTH	KEYB	OFF
20 VCF/VCA	Mix	Freq	Fr/En1	Fr/Prs	Res	Levers	Keybd	E-VCA	VCA/Vel	VCA/En2
	31	40	+50	+50	0	OFF	KEYB	44	+15	+63
30 FM/TRCK	FM	FM/En3	FM/Prs	TrackIn	Track1	Track2	Track3	Track4	Track5	
	0	0	0	ENV3	0	15	31	47	63	
40 RMP/PRT	R1 Spd	Trigger	R2 Spd	Trigger	Port	Spd/Vel	Mode	Legato	Keymode	
	0	STRIG	0	STRIG	0	0	LIN	OFF	REASON	

50 ENV1	Delay	Attack	Decay	Sustain	Release	Amp	Amp/Vel	Trigger	Mode	Lftrig
	10	30	5	63	44	63	+63	STRIG	NORM	NORMAL
60 ENV 2	Delay	Attack	Decay	Sustain	Release	Amp	Amp/Vel	Trigger	Mode	Lftrig
	0	38	26	44	27	40	+63	STRIG	NORM	NORMAL
70 ENV 3	Delay	Attack	Decay	Sustain	Release	Amp	Amp/Vel	Trigger	Mode	Lftrig
	0	0	20	0	20	40	+63	STRIG	NORM	NORMAL
80 LFO 1	Speed	Sp/Prs	Wave	Retrig	Amp	Amp/Rp2	Trigger	Lag	Sample	
	63	0	RAND	0	0	+63	OFF	ON	KEYB	
90 LFO 2	Speed	Sp/Prs	Wave	Retrig	Amp	Amp/Rp2	Trigger	Lag	Sample	
	30	0	TRI	0	55	0	OFF	OFF	KEYB	

Matrix Modulation

	Source	Amount	Destination
0	KEYB	-9	VCA 1
1	ENV2	+22	VCA 2
2	VEL	-51	E2ATK
3	VEL	-20	E2REL
4	PED2	+33	E1REL
5	PED2	+35	E2REL

ROM Review

Each month we audition a collection of sounds for a popular synthesizer or sample. This month our attention turns to **Synthetic Productions'** programs for the entire current range of Casio CZ synths.

Supplied on EZ CZ cartridges, there are two sets of 64 programs (divided into four banks) available at the moment. The first set starts with piano sounds (as close as I've heard on the CZ), but soon gets onto some really realistic 'analog' sounds: Heavy Sync-sweep and Tomita #1 being two particularly good examples. This type of sound is clearly their forté and while we knew the Phase Distortion was perhaps the best digital method for achieving analog imitations, these sounds took even us by surprise.

Bank No.2 starts with a great flute and moves through some excellent bells, strings and woodwinds ending with a very authentic Simmons sound. The third bank has more brass sounds, solo strings and ends with a very spooky ambient timbre called Ghost.

Back to trad synth sounds in bank no.4 with some great synth basses, of which Mini Analog Bass is a clear winner.

Cartridge number two opens with a breathy Fairlight Flute and follows with several other flute clones. Then it covers plucked sounds like Acoustic guitar and a very creditable Harp and finishes with a couple of Hammond sounds.

The second bank deals with keyboard sounds such as Clavinette and Harpsiorgan. Bank three has more analog goodies like Pulse Funk bass as well as ethnic percussion. My personal favorite, Resonant Whip which rivals a MiniMoog for bottom end and warmth, is just one of the goodies on bank 4.

All in all, the Synthetic Productions sounds really do get the best out of the Phase Distortion process and are highly recommended for all CZ owners.

MORE FROM *Synthetic Productions, 545 West End Avenue, New York, NY 10024.* ■

Korg DW8000
Moments in Love
Simon Cranford, Durham NC

No, 'Moments in Love' is not a new romantic novel title, but an atmospheric flute-cum-vocal sound for Korg's DW8000. Simon says it's 'great for those Art of Noise impressions' (hence the title), and it makes good use of Korg's ingenious built-in digital delay. ■

DW8000 Parameters

Number	Name	Value
11	DCO1 Oct	16
12	Wave	2
13	Level	31
14	Autobend	3
15	Mode	1
16	Time	3
17	Intensity	4
21	DCO2 Oct	4
22	Wave	2
23	Level	31
24	Interval	1
25	Detune	0
26	Noise	23
31	VCF Cutoff	26
32	Resonance	0
33	Kbd Track	2
34	Polarity	1
35	EG Int	7
41	VCF EG Attack	7
42	Decay	7
43	Break P	17
44	Slope	23
45	Sustain	26
46	Release	12
47	Vel Sens	3
51	VCA EG Attack	9
52	Decay	11
53	Break P	0
54	Slope	31
55	Sustain	0
56	Release	4
57	Vel Sens	2
61	MG Wave	0
62	Freq	17
63	Delay	3
64	Osc	0
65	VCF	0
66	Osc Bend	0
67	VCF Bend	1
71	DDL Time	7
72	Factor	15
73	Feedback	6
74	Mod Freq	0
75	Mod Int	0
76	Fx Amount	8
77	Portamento	0
81	Aftertouch Osc MG	0
82	VCF	0
83	VCA	0

THE SERPENT AND THE PEARL

MUSIC

Harold Budd is more than just a leading light in contemporary American music. His work has a natural beauty which seduces the listener with tranquil progressions, delicate warmth, and skilful arrangements that use instruments as diverse as piano, marimba, and Synclavier II. *Interview by Dan Goldstein. Photography by Matthew Vosburgh.*

I FIRST HEARD HAROLD BUDD'S MUSIC when a classical guitarist friend lent me his (then) only recorded work, *The Pavilion of Dreams*. 'It's pretty classical, you won't like it', said the friend skeptically, but he was wrong. If I had to pick one disc to take on a desert island, *The Pavilion of Dreams* would stand a good chance of being chosen.

At the time (eight years ago, at a guess), the record's gentle harp, piano and marimba patterns provided an almost perfect backdrop for the breathy alto saxophone and sustained vocal harmonies that took the place of conventional 'solo' instruments. It was a fascinating combination, made more intriguing by the list of playing credits on the sleeve – Budd himself, Brian Eno, Michael Nyman, Gavin Bryars, and several of the other luminaries associated with the EG/ Obscure series on which *Pavilion* appeared.

The album ended up being something of a cult hit, and continues to sell in reasonable quantities a full decade after its initial release. But Harold Budd has never been too interested in the commerce of modern music, and resisted the temptation to make *Pavilion* the launch-pad for a 'recording career' in the conventional sense.

Even now, records seem to appear with Budd's name on them almost by accident. Two albums with Brian Eno (*The Plateaux of Mirror* in 1980 and *The Pearl* in '84, both on EG) and two solo (*The Serpent in Quicksilver* in 1981 and *Abandoned Cities* three years later) do not a steady output make.

Live performances by Harold Budd are equally rare, and press interviews have been very few and far between.

Yet despite (because of?) his low profile, Budd has continued to write compelling instrumental music, switching comfortably from treated acoustic piano to modern keyboard instruments of every level of sophistication – a Casio CT202, an Oberheim Matrix 12, and a Synclavier, to name but a few.

And although his compositions display obvious recurring themes – long, sustained chords; delicate reverb treatments; bright, percussive timbres – Budd is just as at home penning a two-minute cameo ('The Chill Air' from *Plateaux*) as he is improvising his way through an extended, side-long epic such as the title-track from *Abandoned Cities*.

Getting hold of an interview with Harold Budd turned out to be surprisingly easy. The LA-based composer visited London for a month earlier this year to embark on his latest recorded project, an album with the Cocteau Twins – Robin Guthrie, Liz Fraser and Simon Raymonde.

A day before he was due to return to the States, Budd sat in a penthouse apartment above the Townhouse Studios in West London, and began talking. His manner was as friendly as any musician I've ever interviewed, and the overwhelming impression gained after a meeting with him is that, more than most of his contemporaries, Harold Budd has a musical mind that's open to almost any idea, anytime, anywhere.

The Pavilion of Dreams was the first record you made. Before that, what was your background?

Well, I began composing for 'The Pavilion of Dreams' in 1972. Prior to that time I'd been teaching Music Composition at California Institute of the Arts near LA, and before I actually became a college teacher – which I didn't until 1976 – I was an avant-garde composer of the sort of music that probably wouldn't be associated with me these days. I was very much influenced by John Cage and Morton Feldman, for example, and I wrote music of a very indeterminate nature. I wrote very quiet, spacy theater pieces, some of the music was mere verbal directions about an activity which might take place.

That was rather my rôle as an artist up until 1970 or so, when I really minimalized myself out of a career. There was nothing more to do and I became disenchanted with the conceit of avant-garde music.

For two years, I knew what I didn't want to do as an artist, but I didn't know what I wanted to do directionally. I slipped back into discovering something that no-one else was doing, or was likely to do in the very near future. I divorced myself from modern music in a sense, and began to develop a language which I thought was honest to God me, and totally outside of competition with my fellow composers. It was freedom from all that.

I must admit that, looking back, I'm very fond of those pieces. I really think of myself as a composer starting from about 1972. Prior to that, I was the Harold Budd that I don't recognize today.

Were you looking for an outlet to release your music at that time?

I wasn't looking for an outlet because I was teaching in college, and there was always an outlet there. There were ensembles, and I tended to write music for the ensembles that were available, because I wanted to hear what it sounded like. And my music did get played a lot in modern music venues across the country, so I didn't have any problems in getting performances.

But in terms of a record, it never occurred to me. It didn't seem that it was an alternative that had the remotest possibility of happening.

I think secretly I rather had a hunch that it was just a matter of time, and that something would occur. But I didn't work to a plan and I didn't work toward a recording career, shall we say.

How did you eventually get to _The Pavilion of Dreams_, then?

As I recall things, I had quit my teaching position at college, with absolutely no idea in the world what I was going to do, and brimming with confidence that something was going to happen.

I got a call from Brian Eno in London. He said he'd heard 'Madrigals of the Rose Angel' (one of the works that made up 'The Pavilion of Dreams') and asked me if that was the kind of music I wrote generally. I said it was, and he asked me if I'd be interested in coming to London and recording these pieces. So that's how that started.

How do you feel about your two subsequent collaborations with Eno?

The two albums with Eno are very similar. The language, the timbres and sounds are very much the same. It's curious about 'The Plateaux of Mirror'. It came so quickly and so easily that it was kind of a phantom, you thought you could do 30 more of those with the same 'joie de vivre', but it's not that way.

There was an idea that 'The Pearl' would be similar even from the outset. The surprising thing to me is that it turned out as well as it did, because I'm extremely fond of the album. In many respects I like it more than I do 'The Plateaux of Mirror', even though that came so easily and so gracefully and without trial.

'The Pearl' is more of a unity than 'Plateaux'. All the pieces seem to belong together more.

Eno's material tends to be melancholic, whereas yours sounds a bit more optimistic. The mix of the two was quite peculiar...

Yeah. I think that's one of the points that makes it interesting because it's not a specific mood. There's one there but it's very complicated. Sometimes it's a fusion of conflicting waves of emotion. I like that kind of artful confusion. I think pieces turn out not so interesting when they are so emotionally specific, when you can pinpoint a happy piece, then a sad piece. That's too simple.

That can happen very easily when people put lyrics on things.

Yeah, as soon as you have lyrics you're talking about something specific – unless it's an ironic sort of thing, surrealistic. The work that Eno and I have done together is, in a sense, not far removed from classical surrealism. It has an ambiguity of meaning which I think is an important aesthetic consideration for works that can't be placed inside of a fad or time.

"My work with Eno has no specific mood – that's what makes it interesting...when you can pinpoint a happy piece and then a sad piece, things get too simple."

How did your newest collaboration – with the Cocteau Twins – come about?

Well, I have to admit that I didn't know of the Cocteau Twins until one of the band members, Simon Raymonde, was using a piece of mine and Eno's as a cover on an album. The piece was 'Not Yet Remembered', and it was going to be the Twins themselves who were going to do it, or that's the story I heard. The publisher called me up and informed me about this, and said 'they're really a great group and I think you'd like hearing them', so I called up a record distributor friend of mine in Los Angeles and asked if he carried the Twins. He sent me a compilation cassette, so I don't know what the album itself was, or the names of the pieces. But in any case, I was really taken with it.

Then in November they came to Los Angeles. We met and got along famously for a very brief time, and we started swapping ideas on a collaboration of some kind. They asked if I'd be free to do it, and I said 'yeah, absolutely, anytime, just give me a shout and I'll be there'. And here I am in London!

Have you spent most of your time recording?

Yes, basically. Robin has a studio here. It's a 16-track studio and we do the basic tracking there daily, and then he'll take it to a 24-track studio for transfering and mixing.

The music has been very improvisational. It's working the way most people work anyway when left to their own devices. In this instance, everyone takes the blame and the credit equally, so it's kind of composition by committee. It's pretty apparent early on in a piece that something is happening and is ringing our bell. It's a constant creative process in the studio, because nothing is written out. Very, very little is planned beforehand: it's just a matter of laying some tracks down and seeing what works.

Some very strange pieces have come up, I can tell you. That's really the thing about collaboration ▶

MUSIC

that fascinates me so much – the notion that you come up with music that neither one of the collaborators would have dreamed up if they were left alone. It's something that's absolutely unique to the collaboration. Consequently it's not 'pure' in the sense that it's not Cocteau Twins, and it's not me: it's an odd combination peculiar to the mixture.

Do you prefer that style of working to composing on your own?

It really depends upon the nature of the music. There is some music where I absolutely must make all the decisions myself, because I have a sense about its structure, where it's going, and what I actually want it to sound like. I'm the only one who can make those basic decisions, and an input from a second party is kind of counter-productive.

But when you know it's going to be a collaboration from square one, you go in with a totally different attitude. And of course there are two ingredients. Number one is the people with whom you're collaborating; you must like them personally because it's terribly aggravating work. Number two is that you have to like what each other person's art is, with respect to their previous work, so that you just have a hunch that it's going to work out. That's the kind of trust that's impossible until you actually put it on the line.

At the moment I take it that the music is purely instrumental. Will it have vocals on it?

Yes it will. I think Elizabeth Fraser's put vocals on one song so far. Apparently, her method of working – which I totally agree to – is to take the instrumental tracks home and then compose the words and melodies. She works up something that's comfortable to her and seems to work, and that's her way of working.

You're perfectly willing to accept that she will add another dimension to things that you may not agree with...

Yeah, sure. Because in a sense she's rather forced to accept what I'm doing as well. It all goes back to that trust I was talking about.

Who seems to be coming up with the initial melodies?

We have eight or nine songs – well, pieces – tracked. The interesting thing is that the ones which I come up with tend to be 'pieces', whereas the ones that Simon comes up with on the piano are definitely 'songs' – they're structured that way. Mine frequently don't even change key once they begin, or even change chord for that matter (laughs). Robin's pieces tend to be somewhere in between the two, so I can't say, really.

For the first three or four days I laid down a whole bunch of stuff we had to work on, and then I took a break and Simon began some, then Robin began some and now we're back to the other way again. But it's just going to be raw tracking – I will leave an awful lot to them. We have to see how it works out. It's album length now, and it'll appear as an album.

What instruments have you been using to make these strange noises?

Well, it's somewhat restricted. I have a philosophy that one is obliged to use what's there. You don't need an awful lot of stuff, but you use fully everything that's there. What is there now, insofar as my own input is concerned, is a Yamaha electric grand piano and a Mirage sampling keyboard. Robin of course is the guitarist, and Simon the bass player, although he does occasionally play the piano, rather well too.

"The studio I work at in Los Angeles has both a Fairlight CMI and a Synclavier II – they're a million times more than I need, though I know they do a million things."

And there's an awful lot of outboard equipment there which works very well. It's what gives them the Cocteau Twins sound – it's the stuff they use. Primarily it's the Yamaha SPX90, which is absolutely incredible... an amazing instrument.

Do you find, though, that machines like that sometimes offer you too many options?

That's a very very good point. The answer generally is yes. In this particular instance, since the style that's been developed really only uses a small fragment of the potential of the thing, it's perfectly clear that some sounds work and that all the others don't.

But yes, I feel that if you can do anything, then you're liable to end up with everything... or indeed nothing. It's kind of self-cancelling.

But where we've been recording, there is a limited amount of equipment and an even more limited amount of time. There really isn't room for very much finesse. You have to have a pretty good hunch that you're correct very early on, or else you have to abandon the idea and get on with one that's going to use the time more efficiently.

The Mirage and piano seem an odd combination: from something which is very close to a grand piano where you just lift up the lid and play, to something which you have to spend hours fiddling with to get a sound. How have you found working with something like that?

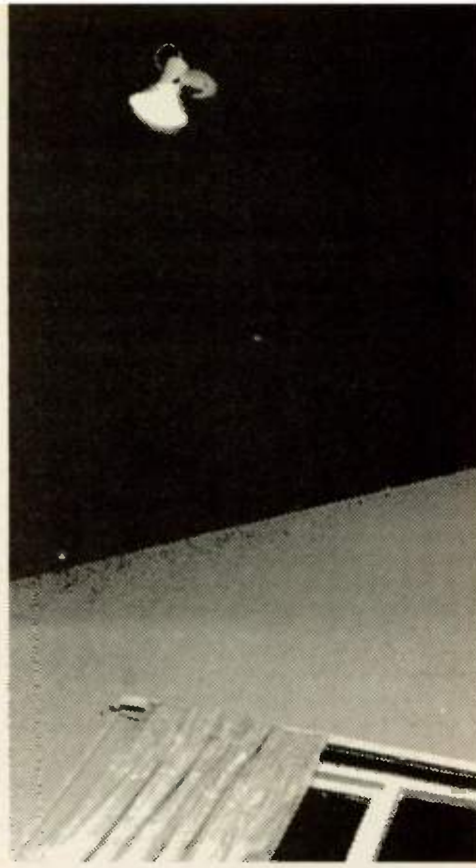
Well, I've worked with something similar before, or at least I've touched on it before. I'd say that 90% of the sounds are crap, and once you find that out, there are certain disks that you avoid, really. With the piano, I tend to work hearing a treatment with it anyway, so I improvise pretty roughly around what the sound is going to be on the tape.

So your composing is very influenced by the way things sound?

Yes. In many respects, the sort of treatment you hear on the piano influences exactly the note-to-note process: the length of time between musical gestures, and the kind of taking advantage of the ringing timbres which I'm very fond of.

Have you done any sampling of your own?

No, I haven't, partly because I don't own a sampler myself. The studio that I work at in Los Angeles – Meta Music – has both a Fairlight CMI and Synclavier II. That's a million times more than I need, though I know it does a million things ... it would go out to lunch if you knew how to program it, I suppose. But really, its capabilities are rather boring to me. What I do like are certain sounds which work for certain pieces. They're rather easy to come by, and the fact that they're on these mega machines is really totally irrelevant.



You don't find it rewarding to explore a machine to find out what it's capable of?

To some extent I do, yes, but not to the extent that the engineer would have to know it, because he would have to know what its capabilities were for situations that are totally unpredictable, depending on the client, for example. For my own purposes, I think it's useful to know as much about it as I can, but I don't see the point in knowing any machine inside-out at the moment.

In what areas has the Mirage been useful?

We've used strings, organ, and chorus sounds... generally things that sound nice when they are sustained for a while, using the instrument as a wash in the background. They don't form the primary sound source, because the guitar, being heavily reverbed, adds a great deal more to it. The guitar does all the glitter stuff much better than the Mirage.

What do you have lined up when you return to the States? Will there be another solo album?

Yes. It'll be a solo work on EG. Half of it is already done, and the other half I still have to do. I have a lot of work to do on it because I haven't got the foggiest idea what I'm going to do.

The first side is a full 20-minute piece. It was done in LA, using the Fairlight and Synclavier,

to London. I did two 30-second ads for an ad agency who were working for Korean Airlines. Apparently, someone had said they could do with something like Harold Budd's music, and someone else had said 'Well, why not get Harold Budd?'. They said I could do anything as long as it was 30 seconds long. The only limitation was that the music had to sound like me!

I saw the visuals, but I didn't have to work to a SMPTE track. I did anyway just to make sure that it would time out correctly, but I didn't have to. The commercial was very open anyway. It was fairly apt for my music.

Last winter I met the head of music production at Universal Studios. I didn't solicit the appointment, he called up and invited me over. He said he'd dearly love to use an original score of mine for a movie, and I said I'd dearly love to do it. But he said that, on the other hand, I must realize that they made 10 feature films a year, none of which are appropriate for the type of music I do, but if something came up would I do it, and would I do it in Europe, in London? I said yes to both questions.

What music do you listen to besides your own?

All kinds of things. I remember that in my high school days I was a great fan of bebop. If it didn't have Charlie Parker on it, I didn't want to know. What a snob I was!

Now I don't listen to all that much. I really rely on friends to turn me on to something that's interesting, or something I haven't heard for a long time. I remember that a student of mine came up to me at one time with the Monteverdi 1610 Vespers and said 'You absolutely must hear this', and it was just devastating. I still think it is, but I couldn't sit and listen to the whole thing.

So I generally play music off records when someone is over who I know is genuinely going to like it, and won't insist on everyone being quiet while it's being played.

You don't seem to play live too often. What's your view of live performance of the sort of music you compose?

Well, I'm not the world's greatest performer, and to make my music sound like I really think it should needs a lot of requirements – things you'd ordinarily find in a professional recording studio. I find soundchecks cumbersome and boring, and since I really don't have much of a clue about what I'd end up playing anyway, what if I have it all set up wrong for a particular piece? I do perform occasionally and I do enjoy it, but I don't see myself in that rôle.

I like the democracy of recorded music: once it's out there it's not yours any longer, people have the option of bouncing the needle, going here or there, and listening to as much or as little as they please. That's why I don't own a CD, incidentally, because once it's on you're a prisoner to it. I can pick up a favorite record and flip it over and say 'Oh yeah, this is the side I like' ►

M U S I C

and a general array of Oberheim keyboards including the Matrix 12. I use the Matrix in a very unsophisticated fashion, which is to say that although it's multitrack, everything is a live performance. I'm not even using the Synclavier to make the notes proper: it's me, what you hear is the way I'm hitting it.

The piece's working title is 'Gypsy Violin', because that's the name of one of the factory presets I used on the Synclavier. It's a gypsy violin sound and it's excellent.

I originally did the piece for an art gallery installation in Los Angeles last November. It was called 'Blue Room with Flowers and Gong' – which describes exactly what it is. It's an environment through which you walk, or just take a quick look and walk away and think 'this guy's crazy'. It was originally the length of one side of a C60 cassette, but that version went on a bit – the sustained notes and so on...

The sustaining note and long, sustaining chord are very much trademarks of yours.

That's true. I really like to find as much life as possible in the smallest amount of material. A very simple scale, a relationship of note against note, especially a sustained note; I milk everything for all it's worth.

For example, I first found this latest piece when I was messing around, and I had it kind of scrolled away in the back of my mind. The first time I sat down and played it, I had a Matrix right above me and I was sitting at the keyboard. I used a bit of reverb – not too sophisticated – and I sat and played for 90 minutes. I was enthralled with the piece and listened to everything, so I didn't have to do any multitracking to find out if it was really going to work or not; it just 'happened'.

I'm generally confident that I can pull this stuff off at the first sitting and get the best version. Invariably I hear two or three versions, but the first is always honest and innocent with all the mistakes, it's more human somehow.

I take it your confidence in improvisation comes from a reasonable training and a reasonable musical ear...

Well, it's nice of you to say that, but in fact I have utterly no training as a performer. I got through college without knowing how to play the piano at all. I was a traditional note composer, and I couldn't play anything on a piano.

Music theory was my strength. It was a different world. I didn't start learning to play the piano until I had to out of self-defence. For the keyboard part of 'Madrigals of the Rose Angel', for instance, I would take out a note card and write the chord down without rhythm for the ensemble, and say 'well it's just got to fit in with the harp somehow', and it was always terrible. It dawned on me that I should be responsible for showing the ensemble how the chord sounded and how it fitted in.

I can now play what I do rather well, but I can play nothing else at all. If you're writing a string quartet, why should you have to play it on the piano? One of my composition teachers demanded real proficiency on the keyboard and a good ear, which I don't have at all; I don't have perfect pitch, or even good relative pitch, for that matter.

Have you done any film soundtracks or theater pieces?

The only thing remotely like that that I've done is something that just occurred prior to my coming

"We've used the Mirage for strings, organ and chorus...things that sound nice when they are sustained for a while, as a wash in the background because the guitar does all the glitter stuff better."

without knowing the name of the tune.

You feel vinyl is still the most immediate means of communication?

I think it's ingrained in our generation. It's going to be difficult to get rid of it. Technology does an awful lot of things, but what it doesn't do is really influence our behaviour – I think we influence the technology.

I know a lot of people insist on CDs only, and many of them insist I'm wrong, but I think they'll take off for a short time and then be replaced by something easier to handle and less expensive.

Given that your musical ideas are very much based around studio work, would it be fair to say that you compose most of your music in the studio?

Yes. I use manuscript paper the way a lot of people use a cassette. I jot ideas down, and I use the manuscript paper as a kind of memory trigger. I often take reams of paper into the studio but I don't read off it – I just use it to jog my memory into an idea.

The engineer at Meta Music is Michael Hoenig, who used to be in Tangerine Dream. He's a good friend of mine so money really isn't a problem – he just tells me to come over when he's free.

I hate being in a studio with a strange engineer who spends most of the time trying to convince you that you're crazy, and doesn't understand what you're doing anyway. I've worked with engineers who want to put the drum track down first, you know, and I can't work that way. I like working with someone sympathetic who knows what the scene is, a good friend. I rely on someone like that a great deal.

Do you have a recording facility of your own at home?

No. Nothing at all. Just a small Baldwin spinet piano gathering dust, and a little Casio 202 which I plug into my son's guitar chorus and piddle around with. I can get as much as I need down from that.

What's the first item on the agenda when you get back to the States?

Curiously enough, a small college in the Midwest commissioned me last year for a piece that had to be 8-10 minutes long, and had to be for any combination of musicians that they have in the ensemble there. They sent a huge list of what they had, and said they'd pay me a certain amount of money and that I could do anything I wanted, but that they had the right to the first performance.

"I got through the college without knowing how to play the piano at all. I was a traditional note composer...If you're writing a string quartet, why should you have to play it on the piano?"

It turned out that I wrote a piece 22 minutes long and it's for a huge choir of 20 people, with percussion and keyboards, and it's the first traditional note piece that I've composed in eight years. It was a lot of fun doing it. So I have to go to the Midwest for three or four days, and be at the first performance.

Looking further into the future, do you have any unfulfilled aspirations, in any field of music, that you'd like to pursue?

No. Maybe it's a personality defect, but I generally discover things as they come along. I don't have any long-range goals about things as a basic personal philosophy, except for the obvious mundane ones. In terms of 'Is there a piece out there that I've always wanted to do?', the answer is no, I don't really operate along those lines.

I'm inspired often by a situation which arises, when there's a certain task in front of me, and if I agree to do it, it's generally because I'm going to like what's happening, without having an idea what the end result is going to be. No career decisions are made.

With that sort of open attitude, you could go in any direction...

Conceivably that's true. I'm the first person to admit that I'm willing to change my mind at the drop of a hat. But I think I would be a very poor commercial composer, for instance. I'll never do anything that I don't absolutely want to do. There may be some commercial success resulting from this latest collaboration. But I don't think I would do anything else if the focus was simply on the amount of units sold. ■

Perspectives on Chicago

Hot from the 1986 Summer NAMM Expo, we bring you details of the most exciting innovations on show.

TWICE A YEAR, THE ENTIRE MUSICAL INSTRUMENT INDUSTRY, manufacturers and retailers alike, converges on an unsuspecting town to show and see the latest musical products. Members of the public are not admitted, but don't worry....Music Technology was there, and what now follows is our exclusive report.

Looking through the pages of this issue, though, we've already managed to cover several of the most important new machines with full reviews or previews. With this in mind, we've opted to give you more of the general flavor and experience that is a NAMM show, by handing over the coverage to several of the Music Technology people who were wandering around the show.

PAUL WIFFEN: The Musician's Perspective

WHILE CHICAGO THIS YEAR WAS NOT BURSTING with new technology to the same extent as Anaheim in January (a treasure trove of innovation), there was still a good number of exciting developments to see, plus more complete versions of those machines originally announced at the beginning of the year) and as usual, four days was not enough for one person to see everything (which is why there were all of us there – Ed).

Several of the most exciting machines unveiled at NAMM are covered more fully elsewhere this issue, but I'll just mention them briefly. The best news for many people will be the fact that **E-mu Systems** launched their first sampler for under $3000, the Emax (previewed this issue). **Alesis** followed hot on the heels of their MIDIverb with the MIDIfex (reviewed this issue), which does for reverb-based special effects what its predecessor did for conventional reverb.

Roland's DEP5 goes one stage further by making several signal-processing functions (including most of the traditional DDL-type treatments) available *simultaneously*. And then we have **Digidesign**, who continue their bid to make the Macintosh all things to all men with additive synthesis and EPROM-burning capabilities being added to the Sound Designer range of software.

E-mu weren't the only company making

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upmarket technology avialable for under $3000. **Kurzweil's** new Model 150 makes those traditional Kurzweil sounds (Grand Piano, Strings, Percussion, and so on) available in a MIDI-controllable, rack-mount version. In all, it has 22 instrument voices, variations of which can be stored in 60 preset locations. The 150 is able to respond to polyphonic aftertouch as well as MIDI velocity data.

Two 16-bit machines I saw impressed me greatly, the **PPG** Hard Disk Unit and the **Hybrid Arts** ADAP (a firmware package for the Atari ST). Both of these use sampling to record audio signals, though the difference in their prices is reflected in the length of time they can sample for; the ADAP system allows 10 seconds' recording in stereo (20 in mono) while the PPG HDU allows 12 minutes of recording shared between four tracks, plus various digital signal-processing functions. But the biggest surprise of all was that the HDU can perform both time and pitch correction: in other words, you can change playback speed without altering the pitch and vice versa.

Of all the stands I visited, **Yamaha's** was perhaps the most innovative, demonstration-wise. The presentations for the new Yamaha Electronic Drums were stunning, showing the new MIDI percussion controllers hooked up to the established TX816 FM modules. This allowed two drummers, John Robinson and Alex Acuña, to play entire pieces (melodic lines as well as drum sounds) just using their drum kits. Simmons has been doing this with its SDS9 kit for some while, but having *two* drummers playing interweaving arpeggios really was a sight (and a sound) to remember.

Talking of good demonstrations, **Korg's** product specialist Jack Hotop was doing some excellent presentations of the DSS1, using some of the many impressive samples he's already made with the new instrument. The sampler now looks very close to release, and features some excellent analog-type manipulation options, plus two built-in DDLs. And if Jack's sounds are anything to go by, it should be accompanied by an excellent library.

Another striking demonstration was given on the **Palmtree** stand of the company's new Airdrums. These are two tubes (you hold one in each hand) which, when moved, are capable of producing a MIDI drum trigger over the full dynamic range. Each stick can trigger six different drum sounds depending on the direction you move it in, which means an entire percussion setup (conventional drums and/or other percussion) can be controlled by one person. The big bonus, of course, comes from the visual aspect of the performance, and these innovative percussion controllers should attract a lot of attention for live shows and videos.

Another percussion controller (slightly more conventional, but not much) that created a lot of interest was **Dynacord's** guitar-like Rhythm Stick. In conjunction with the ADD-One brain, the Rhythm Stick makes a formidable combination, and again gives the drummer an increase in mobility and visual interest, as well as giving guitarists and keyboard players the chance to trigger percussion from somewhere other than behind a set of drums. See next month's MT for an exclusive review...

Moving on to guitar controllers, the show was inundated with them. There were new systems on demonstration from **Ibanez, Charvel, K-muse** (the Photon guitar), **Roland, Zeta,** and **Shadow**. While many of these demos were impressive, we journalists had trouble actually getting our hands on any of them ourselves, so we'll have to reserve comment until they're provided for review. On thing is certain though: this much-neglected area of the market is now really opening up, and guitarists will be spoilt for choice by the end of the year.

Brand new synthesizers were few and far between at NAMM, unless you count models which had been previously announced like the **Kawai** K3 and the **Ensoniq** ESQ1 (both of which should be in the stores by the time you read this, along with the Ensoniq Digital Piano), but Italian manufacturer **Elka** had two new models (each available in modular form as well). These covered the analog to digital range with a conventional dual-oscillator/VCF/dual ADSR model called the EK22, and an FM synthesizer (so close in parameters to a DX, it wouldn't surprise me if the keyboard accepts Yamaha System Exclusive information) called the EK44. Both sounded good, but again, we'll reserve further judgement until we get our hands on them for review.

PAUL WHITE: The Recording Engineer's Angle

TO BEGIN WITH, AKAI in particular impressed me with their new instruments. On the recording side, there was the advanced SMPTE-compatible 12-channel mixer/14-track recorder (the recorder part of which is also available in a rack mount), and an interesting new six-channel/four-track standard cassette recording system which boasts a multi-function autolocate, sync track, dbx Type I noise reduction, 4.75cm/sec or 9.5cm/sec speed and a six-channel mixing console with two-band parametric sweep EQ.

Also of interest were the new ML14 autolocater, the MPX820 programmable and MIDI-compatible eight-channel mixing console, and the GX912 master mixdown cassette. The MPX820 stores all front panel functions including levels, sends, returns, aux inputs, pans and three-band EQ. Impressive!

Not content with their attempt to dominate the home recording market, Akai have extended their range of samplers (which already includes the highly successful rack-mounting S612 and S900) with a new sampling keyboard. The X7000 is a six-voice MIDI multisampler which uses the hardware of six S612s and also features a built-in disk drive. Variable sampling rate is 4kHz-40kHz, and the sampling time is 8sec-0.8sec, with full between X7000 and S612 samples.

Akai also showed their rack-mounting ME20A MIDI sequencer/arpeggiator, ME15F MIDI dynamics controller, ME25S MIDI-programmable note separator, ME30P Programmable MIDI patchbay, ME10D digital delay, and even entered the compact effects market with the new EX65D digital delay, EX70C compressor/gate, EX75P parametric, EX80N noise reduction, and EX85E enhancer. Enough said.

From **Alesis** comes the MIDIfex, which is reviewed elsewhere this issue, along with a neat MIDI control unit which can address 16 MIDI channels and 99 patch changes.

ADA demonstrated their new range of effects, which include programmable harmonization, digital delay, non-programmable multi-effects, MIDI programmable stereo equalizer, and for all those guitarists out there, a MIDI-programmable tube preamp with three gain stages, 128 programs, and programmable effects loop with delay and chorus.

More signal processors, this time on the **Digitech** stand, included the MIDI Master 5000 pitch-to-MIDI convertor, formerly marketed under the IVL label, which allows you to interface almost any acoustic instrument to MIDI equipment, and the MIDI Master 7000 guitar-to-MIDI interface, also an IVL development.

Wandering next to the **Fostex** stand, the new **E** series of professional tape machines leapt out. The E16 is an improved version of the successful B16 ½", 16-track machine, finished in black with microprocessor-controlled transports and record/play logic, so the user can run computer-derived edit decision lists. The E16 also has a frequency response that extends up to 20kHz.

Also in the E series – and much the same size as the E16 – are a ¼" format eight-track machine and two stereo machines (running on ¼" and ½" tape respectively) featuring a third center track for SMPTE control.

Fostex also introduced the FAME (Fostex Automated Media Editing) System. This is a software program designed for Apple 11c and 11e personal computers (though I understand versions for the Macintosh and IBM-compatibles will soon be available). With FAME, a personal computer and the Fostex 4030 synchronizer, audio assembly editing is possible, as the system stores sound effects, cues and so on.

Rack-mounting mixers are now becoming increasingly popular and **Hill Audio** exhibited 16-2-1, 12-4-2-1, and 16-4-2-1 models alongside their more conventional types.

From **Korg** this year at NAMM comes (surprise, surprise) a new digital reverb, but more interestingly, triple and dual digital delays. The SDD3300 has three delays in one rack-mountable box, with all parameters being programmable, and the three units linkable in any way desired, enabling three-phase chorus/flanging, independent left and right stereo chorus with delay and so on. Up to 64 programs may be stored in the memory, and each unit can be used for sampling (with a maximum time of one second), with the 'frozen' playable sound from a MIDI keyboard. The SDD1200 dual delay is not programmable, but similar in many other respects.

Also new from Korg are the DSS1 sampling synthesizer and the DDD1 sampling drum machine. The DSS1 gives 5.5 seconds of sampling time at the maximum sampling rate of 48kHz, or up to 16 seconds at 16kHz.

T E C H N O L O G Y

Additionally, you can alter the waveform, or synthesize waveforms from scratch by specifying the levels of 128 harmonics. Up to 32 programs can reside in the internal memory or stored via the onboard 3.5" disk.

The DDD1 MIDI drum machine features 18 digital drum and percussion sounds. The loudness of each drum depends on how hard you strike the relevant key on the machine's front panel, and the pitch and decay of each sound may be individually set. An optional sampling board allows you to sample your own sounds, and although this contains no means of storing those sounds, the machine as a whole has four slots for factory ROM cards containing alternate drum sets, and a RAM card facility is available for storage of pattern and song data. The DDD1 is scheduled to retail at under $1,000.

New products from Roland include a competitor for the Yamaha SPX90 in the form of the DEP5 digital effects processor. As Paul Wiffen has mentioned, the DEP5 scores by allowing its different processing sections to be used simultaneously, and treatments available programmable reverb, delay, equalizer and chorus. The DEP5 can store up to 99 control settings, which can then be accessed by footswitch or MIDI, as well as by front-panel switches.

On the keyboard front, the two new Roland samplers anounced at Winter NAMM in Anaheim were shown in Chicago in virtually finished form. The S50 has a velocity- and pressure-sensitive 61-key keyboard, 32 envelope generators and 16 LFOs. It features a built-in disk drive and has a 512K memory to store up to 16 different sampled sounds. The sampling rate can be set at 31.25kHz (17.5 seconds), 15.625kHz or 7.5kHz.

The S10 is a budget version eight-voice machine, with just over four seconds of sampling at 30kHz. It also has a disk drive that utilizes the 2.8" Quick Disk standard.

Roland were also showing three new drum machines. The CR1000 has a sound quality similar to the TR505, but is preset only with 24 kinds of basic rhythm patterns. In contrast, the neat little DR220A and DR220E machines are programmable, housing acoustic and electronic voices respectively. They both feature 11 different digital sound sources, 64 rhythm patterns (32 preset and 32 programmable), and can store up to eight songs, each up to 128 measures in length. They don't have MIDI, but both have trigger in and out jacks.

Roland were also building on their early lead in the guitar synth stakes with the GM70 GR-MIDI convertor, which allows any of the Roland guitar controllers to be connected directly to any MIDI device (ie. without using the GR700 floor unit). There's also good news for other guitar owners in the form of the GK1 Synthesizer Driver, which converts a normal guitar into the Roland format and involves a pickup and controller package that's easily installed, with no holes needing to be carved in the body.

New devices from the Boss range include the Dr Pad electronic percussion unit, a pedal compression sustainer, and on the Micro Rack front, there was a new Digital Chorus Ensemble.

Yamaha's new FM tone generator, the FB01 is based on the tone generators found in the CX5M. Retailing at under $500, it contains 240 pre-programmed FM voices, with an additional 96 user-programmable voice memories, though you can't make use of these without the help of external software. Each of the FB01's eight voices can be assigned to different MIDI channels, making it Yamaha's first 'multi-timbral' electronic instrument.

CHRIS MEYER: A Tech's Viewpoint

I only had one day to see the show (mainly because of the ten hours of MIDI manufacturers' meetings the day before), but here are some of the impressions I formed in that short space of time.

The first instrument I saw at the show was the **Casio CZ1**. At last, velocity and pressure hooked up to a powerful and easy-to-program PD (Phase Distortion) synthesizer, and a professional version of this great method of synthesis. The next obvious step for PD voicing is a rack-mount module, and perhaps even more modulation routing capabilities to increase the flexibilty of what is an already powerful system.

Next came the Wersi MK1 Stage Performer, a 20-voice multi-timbral synth with velocity and aftertouch. Up to four voices may be layered at a time, and these zones split up and sent out (and received) over MIDI. It seems to be basically a wavetable synth (à la Kawai, Korg, Sequential, and so on) with onboard additive synthesis provided by way of drawbars – Wersi's organ heritage showing through here. FM and AM synthesis are also said to be possible, and 'sampling technology' has apparently been used in the creation of some of the wavetables. What does it sound like? Big and fat – like a wavetable synth stacked with an analog one. I even heard a variation on the PPG/Fairlight/VS vocal sound I've never heard before (being an acredited junkie of such sounds). The wealth of parameters and the availability of a rack-mount version looks like a very good buy.

Another device which is impressive is the **Bullet** 2000. A brilliant idea, this – an IBM PC-XT compatible computer in a sturdy rack-mount case, with MIDI jacks, power switch, floppy drives, clock in/out, metronome out, and computer keyboard jacks all on the front panel, protected by two macho front handles. This means all you need to get is a suitable MIDI interface (the Roland MPU401, for example), a software sequencer, and perhaps a good patch librarian or MIDI zones program. A rack-mounted high-capacity sequencer with visual editing is just the ticket to take compositions out of rehearsals and into the studio – or out on the road.

At last someone seems to have come up with what deserves to become the standard MIDI interface for the Mac – the **Southworth** JamBox/4. In one rack-mount box is a synchronizer that can take SMPTE or Metronome Click In (yes, one beat-per-minute) and Feed Out Jam Sync SMPTE, MIDI clocking, MTC (MIDI Time Code – the apparent new name for 'MSMPTE'), DIN sync, Metronome Out, and 24, 48, 96, 192 or 384 beats-per-minute Clock Out. Combine this with four independent MIDI inputs and outputs, MIDI routing and merging, Atari ST and Mac interfaces, and the capability to communicate with the Mac at four times MIDI speed (having no delay in sending all four MIDI inputs simultaneously to the Mac) and you have a box which can cope with almost anything you care to throw at it. Four individual MIDI inputs will enable the Mac to become the central computer for zoning, merging, routing, and the such, along with SMPTE synchronization and anything else you would use a Mac for over MIDI.

Additive synthesis finally seems to be making an impact on musical instruments. Additive systems are expensive to implement in real time (needing a separate oscillator or equivalent per harmonic per voice) and difficult to use – ask anyone who owns or has used a Synergy or Crumar GDS. Sampling technology allows the difficult stuff to be done offline, and saved as a waveshape or entire sample to play back later – witness such instruments as the Fairlight, PPG plus Waveterm, and (the only inexpensive one of the lot) the OSCar.

But finally, some inexpensive and readily available instruments are using additive synthesis. Wavetable synths such as the Kawai K3, Prophet VS and Wersi MK1 allow waves to be built using additive techniques. Some samplers (such as the Korg DSS1 and Roland S50) promise to have some form of additive synthesis onboard (although these, too, may be limited to just single waveforms), while others (like the Akai S900) offer it as an software update/alternative. And let's not forget one of the most unexpected hits of the show – **Digidesign's** SoftSynth for the Macintosh (see exclusive review elsewhere in this issue).

Digidesign were also one of three companies at NAMM showing EPROM burners, which allow you to get your own samples into drum machines. The devices in question are the **Simmons** SDS EPB sampler/burner (more or less dedicated to Simmons products), the **Oberheim** Prommer (sampling sounds directly, or via the MIDI Sample Dump Standard), and the Digidesign Burner (using Sound Designer files on the Mac, and burning PROMs for the widest variety of machines of the three). On the other hand, for those who are in the market for a drum machine as well, there's the new DDD1 sampling drum machine from **Korg**.

After the blitz of new guitar-to-MIDI convertors announced at Winter NAMM 1986, this industry seems to be settling down once again. The two main complaints on guitar controllers – bad tracking and inflexible MIDI implementations – are starting to be addressed in earnest. The most expensive and flexible unit of the lot, the SynthAxe, has had its tracking overhauled to the point where they were allowing anybody to come up and have a go on it at the show (I've played an earlier version, and it was nowhere near that responsive). **Ibanez** and **Roland** seem to like using special guitars to ensure good tracking, and I've heard good things about the Ibanez on this subject. The K-muse **Photon** needs the same gauge string across all six strings for the best results. Roland has also resorted to using a separate processor per string on their bass controller (bass *is* harder to track than a guitar – notice the lack of bass controllers on the market). And **Charvel** sells their own custom bridge with the pickup built in.

But the unit that captured my fancy was the new Roland GK1. As Paul White has said, the GK1 is a pickup and small electronic piece that supposedly mounts on any guitar *or* bass, and connects to a rack-mount unit that allows such MIDI gymnastics as Mode 4/Mono Mode with up to four MIDI channels transmitted per string – all programmable. This could be the one for me. ■

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Sound Effects

They stood the world of digital reverb on its head with the MIDIverb, and now Alesis hope to do the same with the MIDIfex, a new budget-price generator of reverb-based special effects. Despite being a preset machine, it sounds more individual than most. *By Paul White.*

NOT LONG AGO, IN A REVIEW OF THE YAMAHA SPX90 multi-effects processor, I commented on the ease with which equipment designers can now use one set of electronic hardware to perform a variety of different tasks, simply by writing new software.

The Alesis MIDIfex, shown for the first time at June's NAMM show in Chicago, is a prime example of established hardware being put to a different job with the help of new, specially written software. As I understand it, the hardware in the MIDIfex is very similar to that in the same company's MIDIverb, the device that stood the digital reverb market on its head only a matter of months ago.

Certainly, the two machines look very similar, as the MIDIfex is packaged in the same way as its stablemate and features a similar list of available preset patches on its top panel – though the preset names don't always tell the whole story. What's more, Alesis will soon be marketing (at very low cost) a rack adaptor that will allow you to mount two of these units side by side in 1U of standard 19" rack space.

Like the MIDIverb, the MIDIfex offers 63 preset delay and reverb effects which may be selected from the front panel or via MIDI on any one of 16 MIDI channels. No user-programmability is available, so you're stuck with the patches programmed by Alesis – though as this is a major factor behind the unit's low purchase price, there shouldn't be too many complaints.

The front-panel display normally reads the number of the preset selected, but when the MIDI Channel button is depressed, it shows the MIDI channel number that the MIDIfex is currently set to receive. In this mode, the MIDI channel may be changed using the Up/Down keys normally used for selecting presets.

In specification terms, twelve-bit quantization is used (as on the MIDIverb) and the background noise is low enough not to have to worry about, provided you drive the input hard enough. Similarly, the bandwidth is limited to 10kHz (again the same as the MIDIverb), but this doesn't seem to cause too many problems with the type of effects being generated.

Both inputs and outputs are stereo and on phono connectors. There is no input level control, but there is a two-LED level meter, and the mix of direct and processed sound can be set using a Balance pot on the rear panel. A single Defeat button mutes the processed sound, and the same may be done by selecting a patch number above 63.

Where the MIDIfex differs from its forebear is in the *nature* of the treatments it offers, which, although being reverb-based, are quite a bit less conventional – and more complex – than those on the MIDIverb. In fact, quite a number of the new machine's effects couldn't otherwise be created without *two* outboard effects units. There are no chorus, flanging or vibrato effects, though, as it appears the hardware is unable to support them.

Presets 1 to 21 are called Echo, and have varying degrees of three parameters: Length, Filtering and Ambience. The presets are all delay effects that have no feedback, but some of them are more complex than they first appear.

The filtering is easy to understand: you get high-pass, low-pass, band-pass or no filtering at all. In other words, you get a sound with reduced bass, reduced treble, a middly sound or a natural sound. Delay too is fairly self-explanatory, maximum delay being about half a second.

It's in the Ambience column that the fun starts, because it's here that a simple delay can become anything but. There are three different Ambience settings, the first being Ambi, in which extra processing gives the delayed sound a sense of depth rather than just making it a blind copy of the input. On vocals this sounds warmer than a simple delay, and it also forms a useful treatment for instruments.

The second setting is described as Thick, which turns out to be particularly interesting. Again, the repeat is not a single straight copy of the input, but sounds to be a burst of closely spaced reflections, rather like a gated reverb – this lends a 'bowing' effect to the echo. An obvious application of this is the treatment of synthesized strings, but it also sounds good on voice, electric guitar, and other synth settings, where the tightly-spaced reflections tend to disturb the rigidly coherent phase relationships that predominate simple synthesized sounds, giving a more natural character to the sound.

The third and final Ambience variation is Wide, which appears to be a double delay which has its two delays very closely spaced, one on each side of the stereo field. As its name implies, this gives the effect of a single delay or echo that has an enhanced sense of width. If a sound requires an echo treatment coupled with a stereo identity, it'll benefit from this setting.

The second row of treatments kicks off with 15 multi-tapped delays – nine two-tap and six three-tap. What this means is that each sound you input is repeated twice or three times. Again though, all is not as straightforward as it seems, as there are various panning tricks thrown in and the enhancements and filter settings used in the first row of treatments make another appearance.

What this adds up to is a variety of textural effects with a dramatic sense of stereo spread. Suitable candidates for treatment are electric guitar, piano, synth and possibly vocals, on which Short and Extra Short presets are useful in creating an ADT effect. Drums, though, tend to sound too messy when processed this way, unless you're after a specific messy treatment.

The rest of this section is taken up with regenerative delay effects, ie. treatments that have some feedback to give a repeating series of echoes. Again, filtering is employed and three of the presets use the above-mentioned ambience treatments to create the illusion of depth and width. These settings can usefully be employed in any situation where you might want a conventional repeat echo, and you do at least have the freedom to select the one that has the best filtering and ambience for your application. For example, a vocal track might benefit from the low-pass filtered version with ambience to give a warmer sound than you could obtain using a straightforward DDL.

The third section of effects starts out with a refreshingly uncomplicated set of four slap-back effects, which on the face of it appear to be single, full-bandwidth delays with varying spacings. These are all short-delay treatments handy for beefing up handclaps or snare drums, and maybe for giving vocals a hard ADT sound.

Following on from here is a series of reverb and spatial enhancement programs. Gated reverb puts in an appearance, as does reverse reverb, and these are in addition to a medium warm reverb and a long reverb with high-pass filtering. These are all fine-sounding reverb treatments of the same calibre as their MIDIverb counterparts, ie. the sort of high-quality effects which, up until six months ago, you couldn't obtain unless you had over $1000 to spend.

More unusual is a preset called Medium Bloom, which has a long build-up to the reverb, rather like the reverse setting, and then a gentle decay like a conventional reverb. Sounds great on vocals and wind instrument sounds.

There's also a reverb program with built-in panning, which is handy for adding movement and life to a static sound, and a setting called Reverse Regen, which is just like a conventional reverse reverb setting except that it repeats until it gradually dies away. Previously, the latter treatment was only possible with a reverse reverb patch plugged through a separate digital delay. It's an ethereal treatment, and it's sure to find a lot of different uses.

Following the reverb programs are three special multi-tap patches which combine elements from the multi-tap presets with reverb. The most involved is Multi-tapped Reverse Pan which, as its name implies, comprises several taps spread across the stereo field, accompanied by that characteristic reverse reverb effect. This gives a wonderful (and slightly unusual) moving texture, and is a great aid to making a simple sound source appear far more complex than it really is.

To finish, there are half-a-dozen spacial enhancement processes, two called Thickener and four called Stereogen. These all add ambience and coloration, and generate an impression of stereo spread without adding any perceptible delay – particularly effective on electric guitar and 'plucked' synth sounds. Because they add coloration, these effects can be used to disguise well-used preset synth or drum-machine sounds, as well as to widen less common sounds from acoustic or electronic instruments. They could even form the basis of some interesting treatments for backing vocals.

All in all, the MIDIfex is a difficult machine to evaluate. No other single device currently on offer represents competition for it, and there are treatments available within it that no other signal processor – regardless of cost – can achieve.

That said, a first encounter with the MIDIfex isn't necessarily as impressive as the equivalent meeting with a MIDIverb. Skim briefly through its presets with a drum machine or an electric guitar, and you could easily dismiss many of them as unsuitable. Check out all the treatments with an electric piano, and you could come to the opposite conclusion. That's how *distinctive* the new Alesis' treatments sound.

The bottom line, though, is that the MIDIfex gives inexperienced (and relatively hard-up) musicians and engineers access to complex production effects that might otherwise need a lot of expensive outboard gear and expertise to emulate. And the fact that these effects sound completely different depending on what you feed through them means there's less chance of them becoming clichéd.

At a time when so many musicians and producers are using the same effects, the same samples, and the same preset synth voices, it's nice to be confronted with a machine that gives you the chance to be individual. How ironic, then, that it should also be a machine that offers no programmability at all. ■

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ADAP SOUNDRACK

Just when 12-bit sampling systems were becoming the norm, Hybrid Arts – in conjunction with developers Nilford Laboratories – come up with an economic 16-bit system based on Atari ST computers. *By Rick Davies.*

IT SEEMS LIKE ONLY YESTERDAY THAT 12-BIT SAMPLERS had managed to settle comfortably under the $3,000 mark. The keyboard and rack-mount samplers offered by a handful of manufacturers differed in the ways most electronic instruments do – control layout, quality of factory-supplied sounds, interfacing capabilites – but for the most part, there was enough room in the market for all of these models, and it seemed that soon all manufacturers would produce 12-bit samplers at similar, or better, prices.

Perhaps that's why Hybrid Arts' ADAP Soundrack came as a surprise at this year's summer NAMM show in Chicago: no one was expecting 16-bit stereo sampling to fall into the same price range as these 12-bit instruments so soon.

The ADAP (Analog/Digital Audio Processor) is a hardware/software package that turns the Atari 520ST or 1040ST computers into powerful sampling systems – for just $1,995. In some aspects, Hybrid Arts' approach may be considered a bit risky. After all, many musicians are still hesitant to incorporate personal computers in their music system, perhaps due to the staggering number of systems to choose from, or due to the questionable roadworthiness of personal computers.

On the other hand, just by looking at the number of sample editing systems based around personal computers, it seems inevitable that samplers will need some extra processing help if any heavy work is to be done. So why not keep the hardware cost down as much as possible?

The ADAP consists of a simple 1U rack-mount chassis which houses A/D and D/A hardware and matching left and right ¼" inputs and outputs, a ribbon cable which connects to the ST's cartridge port, and the software that makes the whole thing work. The ST houses the system's MIDI jacks, and, depending on which ST model is used, a disk drive for storage of samples. A color or black-and-white monitor serve equally well

for displaying function menus, prompts, and other pertinent information.

Once you get used to working with a computer instead of a dedicated musical instrument, you will realize that, yes, you can play a computer. If you're not convinced by this idea at first, perhaps the lure of stereo 16-bit sound quality will eventually persuade you to give such a system the benefit of the doubt.

What the ADAP does is far more than simply sample and playback any audio source you can find. It also provides an assortment of cut and paste-style editing tools which can rearrange passages of music or dialogs into any shape you could imagine; provides two channels of digital audio so that each channel can be edited independent of the other; and provides flexible conversion abilities which allow samples to be downloaded to digital audio equipment using the standard 48kHz sample rate, or to MIDI-equipped samplers implementing the MIDI Common Sample Dump Standard format. Not exactly run-of-the-mill features.

Since the ADAP's sample quality is as good as just about anything else on the market, its flexibility lends itself to editing short musical passages with accompanying dialog, editing and filing of samples which could be dumped to other samplers as required, as well as the standard application as a musical instrument.

Interfacing

The main screen of the ADAP program is the 'Command' screen, from which functions are selected using the ST's mouse. This screen also displays a portion of the sample currently under examination, and lets you hone in on the desired section with 'Zoom-In' and 'Zoom-Out' functions. Other functions on this menu are the various 'cut and paste' functions, as well as sample reverse and looping functions. Many programs compare their editing functions with those of a wordprocessor, but the ADAP seems to truly deliver in this area.

This menu is also where you select either the left or right digital audio channel for editing. This is sure to lead to some truly off-the-wall sounds. Imagine a stereo sample having one channel looping while the other plays backward! An extreme application? Certainly. Fun? Absolutely.

An oscilloscope function allows you to monitor the audio inputs in real-time with eight-bit resolution (the screen graphics require too much of the ST processor's time to allow better resolution).

Connections

The back panel of the ADAP rack leaves little to the imagination when it comes to connections: there are two inputs, two outputs, an input level control, and a power connector and switch. It's unfortunate that this simplicity is not necessarily to the ADAP's advantage. The demand for individual outputs on sampling instruments has been made clear over the past years, and the stereo outputs on the ADAP may well be viewed as insufficient for many of the applications engineers are sure to conjure up.

The ribbon connector between the rack and the ST is the 16-bit parallel pipeline that enables the ST to control and access the rack's conversion hardware. Beyond outside appearances, this system bears no resemblance to other computer-based sample-editing systems, because the rack cannot do anything without the ST, which co-ordinates everything from the response to incoming MIDI data to the routing of the audio sources to the audio outputs.

Comparisons

At some point, it is necessary to consider the pros and cons of a system such as this one, and specifically, how it compares with dedicated sampling instruments.

First of all, there is the question of sound quality. The ADAP can sample at 44kHz, 22kHz, 11kHz or 48kHz in stereo or monophonic. The sample rate affects the amount of available sample time and the number of notes that can be played simultaneously. For example, at 44kHz sample rate, a 1040 ST-based ADAP system can sample 20 seconds of mono audio (10 seconds stereo), and later play back with six-note polyphony. At the 22kHz sample rate, however, the sample length doubles, and 14-voice polyphony is possible.

This is a unique trade-off to my knowledge, and will surely be a popular one at that. One of the ways that the ADAP is able to pull this off is by not having VCFs on each voice, as is typical of most instruments with a fixed number of voices. This means that there are no filter sweeps available, but if you don't require that feature, then the ADAP's digital EQ should help tailor the tone of your samples to some degree.

The ADAP will be able to hold 64 multi-samples (sample time allowing), playable over MIDI by various controllers. (Unfortunately, the prototype I looked at did not yet have the ADAP's entire MIDI specification implemented.) Again, the dynamic voice allocation helps the samples to get out in an orderly fashion, but is alas probably responsible, in part at least, for the two-output limit.

If separate processing of each sample is the major concern, then the ADAP's internal digital effects 'rack' should help compensate for the lack of individual outputs. In the 'rack' portion of the program, the screen displays five spaces in which you can place effects such as delay or reverb. These programmable effects are merely algorithms which may be applied to samples upon playback which affect the sample data, rather than the final analog audio signal. Better yet, samples can be stored on disk in processed form. The ADAP is to ship with reverb and delay effects, leaving three additional 'rack spaces' for other effects which may be purchased on separate disks. It'll certainly be interesting to see how these internal effects compare with conventional effects when the ADAP is in production.

This brings us to the matter of sample storage. Files are saved to disk using graphics programs, and each 3½" disk can store one full 1040ST memory. The addition of a hard disk to the system allows storage of 30 files, and would also speed up the loading time considerably.

If desired, these files could be transfered over the Atari's serial interface with little fuss, allowing other digital audio systems to use the data.

Wendell Brown, President of Nilford Laboratories, indicated that they intend to use a sample file header in the same format as the one used by Digidesign's Sound Designer files, so files could be swapped between these two systems.

Similarly, should an ADAP system be joined by a sampler which uses the MIDI Common Sample Dump Standard Format, samples could be transfered back and forth as desired.

Conclusions

All in all, the ADAP strikes me as a system with great potential. The sound quality, sample editing features, and attention to detail look promising, and considering the price, it's excellent value – even with the additional cost of the ST computer which, needless to say, would obviously come in useful for other things as well. I can see that this also opens up other possiblities, since the system's performance is dependent on software, and software can be updated.

Hopefully, the ADAP's multiple personality will not confuse many people. It may not replace dedicated sampling instruments in all situations, but it certainly is working in new territory without losing its footing in more common areas. It will be interesting to see with which other areas of digital signal processing the ADAP will deal. In time, of course. ■

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S A M P L I N G

Toward More Creative Sampling Part II:

ALIASING...

And How to Avoid It

If you've spent any length of time sampling, you'll be familiar with aliasing and the detrimental effect it can have on samples. There are plenty of ways you can remove it, though, along with methods which avoid the effect in the first place. *By Paul Wiffen.*

BEFORE WE CAN BEGIN TO DO ANYTHING about aliasing, we need to be able to differentiate it from the other two main problems which can plague the intrepid sampler: clock noise and quantization noise. If you don't know the difference between these three effects, you might do well to read through the 'Mechanics of Sampling' article elsewhere in this issue.

Aliasing can sound like anything from keys being scraped across a metal surface, to bell-like or ring-mod type sounds. For these reasons, there are times when it can be an interesting effect (like the distorted roughness that quantization can add to drum sounds), but usually, it's nothing but a nuisance.

The first and most obvious thing you can do with aliasing is to use a filter to get rid of it. You will, of course, lose the top end of your sample, so the only way to use this method realistically is if you have filter tracking available. This means that you can have the filter closed lower down the keyboard (where the problem is usually most obvious), but fully open by the time you get to the original sampled pitch and above. Be warned, though, that it can be difficult to get the balance right between the Cutoff Frequency and Keyboard Tracking, especially if the parameter system on your sampler doesn't allow you to get access to both at once and you have to keep moving backwards and forwards between them.

But there will always be times, especially on eight-bit machines, when you won't be able to get rid of aliasing completely without making your sample sound much duller. If this is the case, try using a psychoacoustic enhancer such as the Aphex Aural Exciter to put back some of the sparkle. This is better than trying to use mixing console or separate parametric EQ, as these can only work on what top end is left after filtering, and any there is will probably still have the residue of the aliasing left in it. An enhancer, on the other hand, uses the lower frequencies to deduce what higher harmonics might be in the signal, so it shouldn't bring the aliasing noises back to prominence within the sample.

Of course, all these measures are really compromises designed to cure the problem rather than prevent it. What we really need to look at, as soon as aliasing makes its presence known, are ways of resampling the sound. Without wishing to state the obvious, the first thing you can try is raising the sample rate – so long as the sampler you have offers you a choice of these. Never try and squeeze an extra little bit of saved memory and end up with a sample of dubious fidelity; it always restricts the uses you can put that sample to later.

If you have a Mirage with an Input Sampling Filter, you have a great deal more control. For the $144.95 that the filter costs, it really does allow you make some appreciable improvements in the quality of your samples. Refer to the list of available cutoff frequencies that comes with the filter, and slowly close the filter down to the point where the aliasing ceases to annoy you. The Input Sampling Filter also allows you to increase the sampling rate to 50kHz, which you'll also find a great help.

While 12-bit samplers may give you greater fidelity in the first instance, they don't allow you access to the cutoff frequency of the anti-aliasing filter, so we need to find some more ways of correcting aliasing. One tried and tested (if rather long-winded) method is to record your sounds on tape, play them back at half-speed and sample them, then transpose them down an octave on the keyboard so that they're being played an octave higher than sampled – which then gives you the original pitch. You should find that the aliasing effects disappear completely when you do this, because you're effectively doubling the sampling rate by playing the sample into the machine at half-speed.

You don't get anything for nothing in this world, though. Because you're recording at half-speed and then an octave higher, you end up doubling the amount of memory you use to record your sample. And if you have the type of sampler that only plays an octave above the original sampled pitch, this method makes the original 'taped' pitch of your sound the highest note you can play. If the sound you're sampling is a snare drum, this limitation won't worry you too much, but it can be a problem for tuned instruments. In the latter case, you'd do best to tape the highest note you want to play and then sample that pitch at half-speed. That way, you should still get the best part of a two-octave range out of your sample.

An alternate technique – and one that doesn't impose pitch limitations – is to use EQ on your sample input, and experiment with different settings to try and find the frequencies that are causing the aliasing. Once you've discovered the frequency band that's causing the trouble, you can resample with that band attenuated, which should lead to a drop in the aliasing effect.

When using the above method, I've occasionally come across a band of frequencies which 'mask' the aliasing to some extent. You'll probably have trouble finding such effects if you deliberately set about looking for them (I came across them by sheer chance), it's worth keeping half an ear open for any lucky breaks.

Hopefully, this piece will have given you a few ideas to help you keep unwanted aliasing at bay. However, before you set about eradicating all aliasing effects, remember that a lot of people seem to like the sound of aliasing, and several careers have been built upon its characteristic sound. ■

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M U S I C

Concert Review
Peter Hammill
The Roxy, Hollywood.

A PETER HAMMILL GIG THESE DAYS IS A RARE EVENT (indeed this was only the second he has played this year, the other one being in London), not just because of how indifferently he plays, but because it is rare these days to see one person on a stage with a minimum of equipment hold an audience spellbound with the sheer power and intensity of his performance. Using just a Yamaha CP80 and a contact-miced acoustic guitar (both unfortunate compromises necessary to touring musicians, I'm afraid), he held the stage normally occupied by an entire band, and his presence filled the entire venue.

The material included material from his new album *Skin*, his previous solo albums, right back to the very earliest Van Der Graff. Songs both old and new were greeted with the same rapturous applause from the audience who were clearly familiar with all of Hammill's 20-year career. Songs like 'Time Heals' (from the immaculate *Over* solo effort) and Van Der Graff's 'Still Life' blended perfectly with newer material and even the two non-Hammill penned numbers (both by the mysterious Chris Judge Smith, as Hammill referred to him) were integrated into the set seamlessly. Only two songs 'If I Could' and 'Ophelia' suffered at all, and that was only because there were not several Peter Hammills available to do the falsetto backing vocals of the album versions (this man needs a Korg DVP1). But even so they came off well considering there was only one performer.

Hammill is one of the few performers left who can play a poorly amplified acoustic guitar (with some tuning eccentricities) or a compromise electric piano like the Yamaha CPs (which are starting to sound very dated in the face of new products by Ensoniq, Roland and Technics) and it really doesn't matter. His performance overcomes the technical limitations of the gear. There is a lesson for all of us technology buffs, that we should never lose sight of the importance of the music when we are surrounded by all the technology – a trap that Peter Hammill is never likely to fall into. ■

Eddie Jobson
Theme of Secrets
Private 1501

WHAT CAN IT BE THAT HAS FINALLY enticed keyboard virtuoso Eddie Jobson to abandon his beloved Yamaha C580 and MiniMoog which have served him so well whilst all around him have been changing their setup every six months?

Well, the credit (or blame) rests with New England Digital on whose Synclavier system the whole of Eddie Jobson's most recent album *theme of Secrets* was performed.

Having said that, it may be somewhat misleading as *Theme of Secrets* is a special project for the Private music label started by ex-Tangerine Dreamer Peter Baumann. And frankly the music owes more to Baumann's past career than it does to that of jobson. Whilst his last solo project *Zinc* in 1984 saw Jobson much closer to his earlier contributions to Curved Air, Roxy Music, U.K., and Jethro Tull, it was apparently a short sequencer part on that album which led Peter Baumann to approach Jobson to do an album for Private.

Clearly, the name of the label gives an indication of the sort of music you might expect to hear on the album: introspective, atmospheric and contemplative, and that is exactly what this album is. The opening track 'Inner Secrets' features reflective piano (a credit to Synclavier) over ambient-type backing, and the next track 'Spheres of Influence' is even more abstract, sitting somewhere between Floyd's 'On the Run' and Musique Concrete. From that point on things get more structured and melodic. Some recognizable elements of Jobson's previous work begin to show up, like unexpected key modulations and complex time signatures but on the whole the emphasis is atmosphere and effect rather than technique and complexity. 'Ice Festival' comes closest to his previous output, continuing his fascination with all things arctic (see 'Alaska' with U.K. and 'The Sahara of Snow' with Bruford).

The title track with its sequenced bass line and breathy vocal samples is closer to Jarre than anything else. 'Lakemist' is a personal favorite combining ambience and dramatics nicely.

The quality of the recording is excellent but then this is only what you would expect from the Synclavier recorded directly onto the Mitsubishi X80 digital multi-track.

Peter Baumann's production gives the whole album a very European Electro-Music flavor and the engineering, by a host of people including Jobson and Baumann is very clean and crisp as befits all the digital technology involved.

This album may not satisfy fans of Jobson's previous 'techno-flash' style (at which he does excel) but with listening the initial impression of superficial resemblances is replaced by an appreciation of the intimacy and grandeur of the compositions. ■

Independent Release: The Telling

EP on In Praesenti

THE TELLING'S MUSIC MIGHT FALL into the category of 'gothic rock', but the music on this, their first EP, indicates influences somewhat less-gloomy than such a label suggests. The band's two members, Eris and Don Swanson, use a variety of keyboards, drum machines and processed guitars, complimented by engineer Richard Burmer's contributions. The EP starts with 'Ghost of Tokyo' revealing the band in a dance mood, with whispered vocals and Emulator II samples adding a few eastern touches to the tune. Instrumental 'Cocoon' ends side one with a quiet mood, then 'Sand Mass' opens side two with a low rumbling drone unlike anything I've heard in some time, let alone on vinyl, accompanied by Eris' keyboard and vocal drones. 'Heaven Never Ends' demonstrates The Telling's ability to blend aggressive rhythms and dark melodic textures with a delicate touch, while Balinese chants are brought in, courtesy of the Emulator II, during the song's processed drum interlude. An impressive debut. For info contact: In Praesenti, PO Box 15216, North Hollywood, CA 91615.

M U S I C

Reader's Tape: Group: Surrender

3 track demo

EVER SINCE THE FIRST ISSUE OF MT hit the streets people have been sending us their demo tapes, or handing us them at fairs and other events. Of those we haver received so far, the three song offering from Surrender are among the best we have heard.

Surrender are a 3-piece from the San Fernando valley area. They are Julie Larson (vocals, guitar, percussion), Dave Hill (guitars, bass, keyboards, backing vocals) and Andrew White (drums, percussion). They have been working together now in various formats since 1984.

Of the three, 'New Shoes' is the most danceable with a slight new wave feel, jagged vocals over a punchy rhythm track. My personal favorite, 'Falling Over You' is a ballad with a wonderfully fluid guitar solo. 'Everything Takes Time' relies heavily on Simmons and other percussion to drive the song along.

The songs were recorded on a mobile facility known as 'the Truck' (✆ (818)-507-TRUK), engineered by owner Bruce Black. The quality is very professional and there are several interesting effects used in the recording process.

If you have a demo tape you would like us to listen to, send it to *Music Technology, 7361 Topanga Canyon Boulevard, Canoga Park, CA 91303.*

I·N B·R·I·E·F

E-mu Emax Sampling Keyboard

The new 'baby Emulator' nearly stole the show at the recent NAMM Expo in Chicago. We take a closer look at one of the prototypes at E-mu Systems' headquarters. *By Paul Wiffen.*

AN 'IN BRIEF' PREVIEW THAT'S TWO PAGES LONG? When *will* Paul Wiffen learn to keep his mouth shut, his paper quantities down, and his word processor in check? Probably never, and certainly not when there are instruments as interesting as E-mu Systems' new Emax sampler around to preview. One trip to E-mu's Scotts Valley headquarters and he's away, grabbing hold of one of just three prototype Emaxes currently in existence, and putting it through its paces.

Prototypes being what they are, though, Wiffen finds a fair bit of Emax's software still in the process of being written, so a full-blown review will have to wait a month or two. In the meantime, here's a rundown of what the 'baby Emulator' can do to get your mouth watering, because what there is of Emax now is already mighty impressive...

Emax retains the same sampling format as its big brother, the Emulator II, but E-mu's engineers have succeeded in reducing the fundamental circuitry so that it can now be fitted on a single silicon chip, known as the E-chip – hence the dramatic reduction in cost. What's more, memory chips have now come down sufficiently in price to allow the Emax to hold the same amount of sample memory as the original Emulator II.

This means that at the same sample rate as the Emulator II – around 28kHz – Emax samples for the same amount of time: over 17 seconds. But E-mu have added variable sampling rates from 15kHz to 40kHz, the range given in the preliminary specifications. And on the prototype unit I tried, the display gave the maximum sample rate as 42kHz, which makes Emax at least the equal of the Prophet 2000 and Akai S900 in this respect.

And the quality shows, too. Recording direct from a CD player, sound fidelity was extremely fine, reaching well into those high frequencies above 13kHz which many samplers reproduce poorly or not at all. Of course, you pay for this extra facility by having less sample time available. Maximum sample length at this rate was just over eight seconds, but fortunately, the display told me exactly how much time was available *in seconds*, not K of memory or hexadecimal.

It's the display (similar to that of a DX7) which makes sampling with Emax so easy. On entering Sample mode, the bottom line of the display acts as a VU meter in much the same way as the EII's, and a threshold can be set visually using a slider. In fact, all the other instructions in Sample mode (Arm Sampling, Force Sampling and Stop Sampling) operate identically to those on the EII.

This user-friendliness isn't restricted to sampling, though. It also extends to the use of all the other modules, and to the provision of one all-important button (missing on most samplers): a dedicated 'load all' switch, which means you only need to perform one action to load a new disk. Great during live performance.

The other modules of Emax (besides Sample) are Master Control, Digital Processing, Preset Management, Preset Definition, Analog Processing, and the Sequencer. Each of these has its menu printed on the front panel (no loose card to lose with Emax), so you can look up which option you want to use and key in the appropriate number on the keypad. Alternatively, you can use the data slider to scroll through the options and then press Enter to select the one being displayed.

Master Control governs essential functions like master tuning and disk formatting/copying. But it also holds a few more unusual features like Select Velocity Curve and Bird Run. The first is an excellent feature which allows you to personalize the touch response of the keyboard to suit your own playing style. The second is a little bit sillier, causing an Emu (yes, the bird) to trot across the display. I'm not sure exactly *why* it does this, nor am I sure why it's there at all. But I am sure E-mu could have made better use of the memory and processing power Bird Run takes up.

Digital Processing covers all the things you may want to do to a sample once you've made it. First there's the essential Truncate, which allows you to trim any unwanted material from the beginning or end of the sample. Being able to type in individual sample numbers is a great benefit here, as sliders can be a shade fiddly for fine adjustment.

Next come the two loops possible on Emax: the normal loop and the Release loop. Each can be specified by its start point and length, but before you can begin to adjust these loops you need to go to the Mode parameter to turn the loops on, which can be a pain. Once you've done this and set start points and lengths, Emax gives you the option of using Autoloop, which works by finding the nearest zero crossings. This goes a good way toward getting rid of awkward clicks or glitching in a loop.

If this is not enough, though, there's another feature – new on an E-mu keyboard – called Crossfade Looping. This process actually smooths out the data around the loop point to create a 'seamless' join. It works fine *so long* as you've got as close as you can before, using the manual and Autoloop functions.

Other, more 'experimental' Digital Processing options include Tapering, Splicing, Combining and Digital Effects, but these were yet to be implemented on the Emax we tried.

Preset Management is a utilities module, allowing different presets to be Created, Copied, Saved, Named or Erased – essential if unexciting. More interesting stuff goes on in the Preset Definition and Analog Processing sections. In the first of these, you can set up all the crossfades and switches you want. There are three options here: Positional (to smooth out the differences between multi-samples as you go up the keyboard), Velocity (to allow different force keystrokes to trigger different samples recorded as 'hard' and 'soft' performances), or Realtime (which allows you to use the mod wheel to change between samples).

It's only when you set up one of these crossfades that you notice what is one of the strongest points of Emax. Normally, when you make such assignments on keyboards, the machine's polyphony is halved: if you're using two voice channels to play different sounds from the same note, then you can only sound half the number of notes. Not on Emax. Each voice is provided with the equivalent of two oscillators, so it can play back two samples simultaneously.

The usefulness of this isn't restricted to crossfade assignments. In Dual Voice mode (another area of Preset Definition), two different samples can be layered to provide, say, Brass and Strings or Piano and Nose-Flute together, again without losing polyphony. This feature is also responsible for the chorusing provided in the Analog Processing section.

Preset Definition *also* covers MIDI options, pitchbend range, arpeggiator setting-up and real-time control assignments. The last of these refers to the controllers: left wheel (sprung), right wheel, pressure (Emax features a pressure-sensing keyboard as well as velocity control), pedals, footswitches, and MIDI controllers, all of which can be assigned to a whole bunch of destinations: pitch, filter frequency, level, LFO, attack rate and sustain, to name but a few.

The two wheels, incidentally, are small and can get tricky to use, though their action is fine.

The Analog Processing section contains all the things keyboard players raised on Mini-Moogs, Prophet 5s and Juno 60s can't live without: VCAs and VCFs (with their own independent five-stage envelopes), filter tracking and resonance, LFOs and velocity amounts to level, pitch, filter, attack and panning... Panning? Well, Emax allows you to pan each voice to a stereo position, and then move it either through velocity or LFO control. It makes for great effects or realistic mixing, whichever you prefer.

On top of everything else, Emax has a sequencer. Not an exhaustive one, it's true, but then it's not designed to be. The idea is that you can dump sequences recorded on more powerful systems (either PC-based packages or stand-alone units) into the Emax, and then save them onto the same disk you're keeping your sounds on. This means that, during a live performance, you can have everything loaded from one disk.

And despite its basic specification, the sequencer is capable of multitrack recording and is particularly strong on the sychronization side, featuring not only MIDI clock but also MIDI song pointers, which allow you to record and playback from anywhere in the track. Emax can also receive 88 notes via MIDI, over all 16 MIDI channels, simultaneously if needed.

On the back of Emax, in addition to the 11 audio outputs (Mix, Stereo, R and L, and eight individual outputs), the MIDI connections and the sample and footswitch inputs, sits an RS422 computer port. This will enable visual editing software on PCs like the Macintosh to communicate with Emax much faster than MIDI can. And as it turns out, Digidesign has already told us that they hope to have their Sound Designer Mac package available for Emax at the time of the keyboard's release in the fall. ■

PRICES Emax keyboard $2595; Emax rack-mount module $2395

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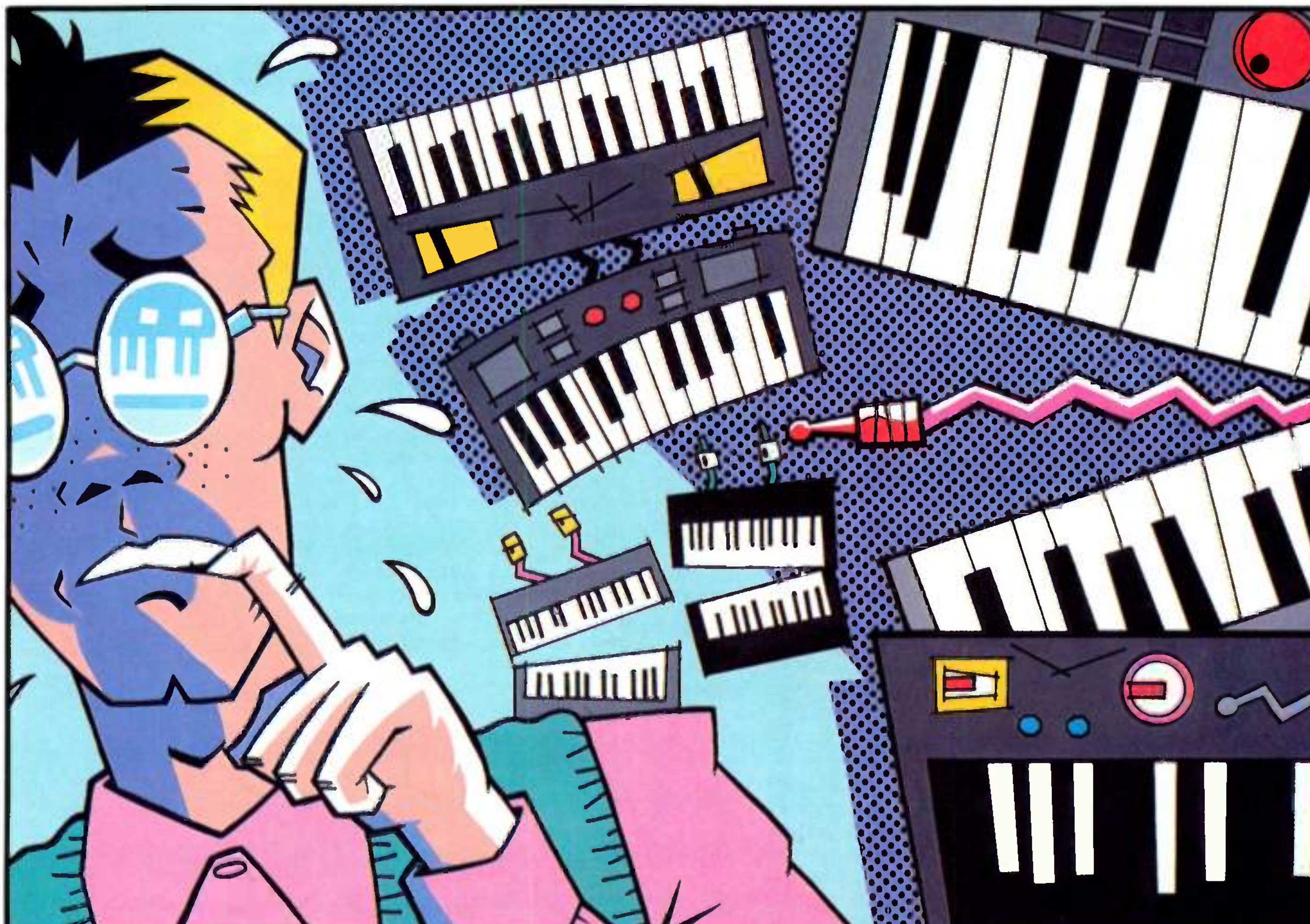

Illustration: Clive Goodyer.

Continuing from the inaugural issue's 'First Take' feature, we conclude our look at how to avoid early recording sessions turning out to be an expensive disappointment.

By Paul White and Dan Goldstein.

IT'S UNDERSTANDABLE THAT SO MANY MUSICIANS DREAM their first recording session will yield results of impeccable quality. Sadly, it's also true that far too many bands over-estimate the *quantity* of material their finances – and the time they buy – will allow them to produce.

Inevitably, trying to record too many songs has one of three consequences. Either you discard some of the pieces halfway through the session when it becomes clear there just isn't enough time to get them all done properly; or you cut corners and end up finishing all the songs to a standard that does none of them justice; or you just run out of time and finish nothing, which usually results in everybody emptying their pockets, foregoing alcohol for three months, and saving up for some more studio time so that all the material can be sorted out.

The most efficient way to work is to put the backing tracks down for all the songs, then all the vocals, and finally any additional bits of decoration (solos and so on) that need overdubbing. As it can easily take a couple of hours to set everything up and get it sounding right, you should aim only to get a maximum of three songs done in a day, unless you have everything *very* well planned and are confident that you can do more. Don't forget to add time for the mixing stage which, for reasons we'll go into later, S T U D I O

can be a good deal more time-consuming than many people realize.

So now you've made all your preparations, decided that professional realism is preferable to over-confident elaboration, and you're about to enter the world of professional recording. Everything from here on should be plain sailing, so long as you adopt the same efficient attitude.

Treat the whole thing like a military operation. Ask the engineer where he'd like you to set up, and once everything is working, get tuned up and check that the sound meets with your approval. If there are a lot of you, don't all play at once or decide that this would be a good time for an impromptu practise. Once the engineer has got his mics in place, he'll probably ask you to run through a song so that he can get the levels right – you'll get a chance to practice then. After this, don't alter any volume settings without telling the engineer, or you may put in the performance of a lifetime only to find that the meter needles are wrapped round their end-stops and everything is distorted to heck.

Other clever tricks to be avoided include the following: tapping your foot on one of the legs of the mic stand; starting to talk at the end of a take before the final cymbal crash has died away completely; and moving mics without the engineer's permission. If this sounds like Colditz rather than an environment likely to produce a creative musical experience, that's partly our intention. Recording can be as demanding as an escape from Alcatraz, but if you are reasonably thoughtful and the engineer is on your side, it can be a lot of fun, too. And in any case, most of the band can relax once all the backing tracks are safely out of the way, and the spotlight turns to the vocalist.

For singers who've never really heard themselves because of loud backing and non-existent foldback, the first foray into a recording studio can be a painful experience of self-enlightenment. Being clamped between a pair of headphones through which you can hear yourself clearly above everyone else frightens a lot of people, so don't worry. If you're responsible for the major part of the vocal chores (backing singers have it comparatively easy), you might have trouble coming to terms with the fact that maybe you're not quite another Daryl Hall after all, but take heart – you should be able to ask the engineer to feed a bit of echo or reverb into the cans, which'll help to lull you into a false sense of security.

Your problems don't end there, though. There are some further rules governing how you approach singing during recording, and one of the least flexible is that you should avoid moving around too much in front of the microphone. It may be easy to get carried away with the music, but swaying from one side to another won't get you anywhere; you'll probably end up having to do another take because the vocals varied so drastically in level first time around. If you prove troublesome in this respect, a considerate engineer will resort to sticking your nose to a drum stick attached to the mic stand, which is none too comfortable.

"For singers who've never really heard themselves before, the first foray into a recording studio can be a painful experience of self-enlightenment."

When all the tracks are down on tape and everyone has dropped in chunks to replace passages with mistakes in them, it's time to get on with the mix. For the uninitiated, the mix is that time when everyone wants to be the loudest except the singer, who suddenly doesn't want to be heard at all.

Respect the engineer's advice, at least to start with, and remember that a good mix doesn't mean all the instrumental parts being equally as loud as each other: if anything, quite the reverse is true. Once the engineer has set up a rough mix, you can ask for subtle changes of pan position, EQ and level, but while this is going on, make sure you're listening to the song *as a whole*, not just your part. And don't be tempted to try to bury a duff bit of playing or singing under yet another overdub – it'll only clutter things up.

Your engineer may feel you're responsible enough to take care of the reins for a short time while he goes and gets a Chinese takeout, and if you're reasonably *au fait* with mixing techniques (Portastudio experience will prove invaluable here), this should be great fun. Beware of getting bogged-down with the number of available facilities, though. It's a trap that's all too easy to fall into, and once you're in it, you'll be struggling to get out again.

The danger looms even larger when it comes to using outboard effects. These are wonderful things (more wonderful now than ever, in fact), but used to excess, they can destroy a recording. Use a little reverb on the drums and vocals, by all means, but don't swamp them. More vicious effects like flanging should be used in small doses if they are not to become wearing, and the same applies to heavy repeat echoes.

Be careful, also, to avoid clichés. Particularly dreadful (and popular) ones include repeat echoes on the last words of lines and at the ends of songs, flanged drum patterns, and panning from side to side during guitar solos. Try to find effects to suit the song: if it doesn't need any, stick to just a tiny amount of reverb to take the dryness off things.

But this problem doesn't end with simply applying each common effect in the right quantity in the right place. There's also the dilemma of being faced with a battery of weird and wonderful new goodies, trying to get weird and wonderful results out of them by fiddling aimlessly, and finishing up with three hours less to complete the mixing and no new effects to show for it. In much the same way as unfamiliar in-house synthesizers should never be used with anything other than extreme caution, so outboard units should be treated with similar care. As technology advances, so these units become capable of creating ever wider ranges of effects, with a corresponding increase in the number of possibilities they present the end-user. Even low-cost machines such as the Yamaha SPX90 and Roland DEP5 offer a vast range of variable parameters you'd have trouble grasping fully in a week, let alone a couple of hours. The moral? If you decide to start tampering with any studio gear that represents unfamiliar ground, keep an eye on the clock. Stick to tried and tested sound-treating principles and don't attempt anything too fancy.

When the mix is complete, run off a cassette copy and check it on a home hi-fi system or car stereo. You can guarantee that whoever else listens to your recording after the event *won't* be doing so over studio monitors, and what sounds great in the studio may sound less impressive outside. If you think your recording may is likely to receive some airplay, take a small radio-cassette machine into the studio with you and see how it sounds.

If you're hiring the master multitrack tape rather than buying it, ask the studio to keep it intact for a few days so that you can remix or replace any bits that are unsatisfactory. You might think you'll be able to spot all these at the mixing stage, but music has a habit of sounding very different the day after, when your hearing has returned to normal.

In fact, the problem of retaining objectivity during mixdown is what leads many people to record and mix on entirely separate occasions, days or weeks apart. It doesn't make much sense to mix more than one song at any one time, which means that during mixdown, it's possible you'll listen to the same four-minute piece of music 20 times in succession. And if you've spent the last four hours recording that piece of music, that's the *last* thing you should be putting yourself through. ■

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ADVERTISERS' INDEX

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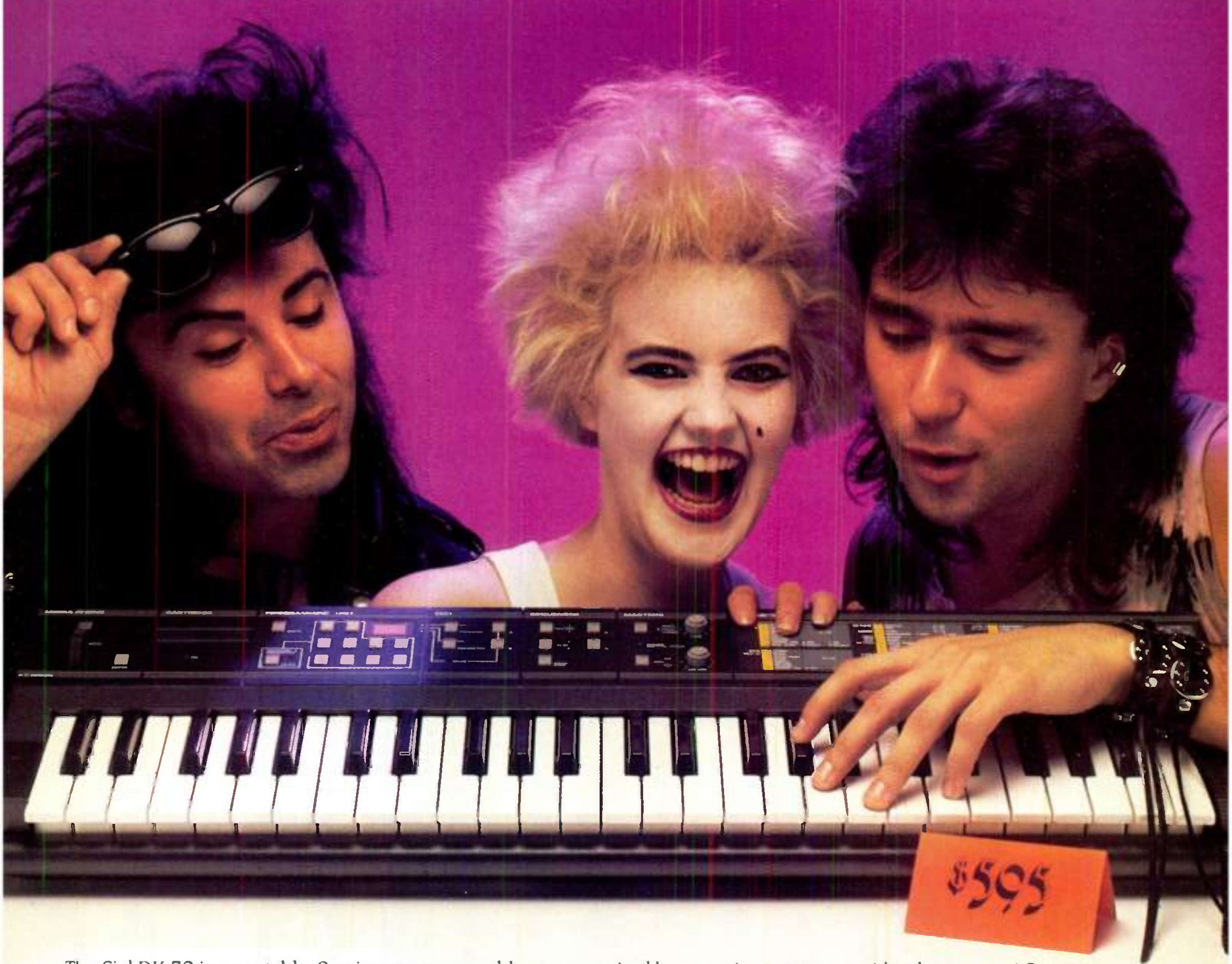

The Siel DK-70 is a portable, 8-voice, programmable synthesizer with that big, fat sound you've been searching for. But a real lean price of $595.

With it's two-track, real-time sequencer, you can overdub, remember program changes, loop musical sounds. Even outboard the sequencer to drive another keyboard.

The optional "guitar neck" puts a ribbon-controlled pitch bender at your fingertips. Not to mention control over modulation, sustain, transpose, on-off to your on-board sequencer and quick step through to your programmable sounds.

The DK-70 has full MIDI and receives touch sensitivity.

And because it operates on either battery or AC power, you can jam within your existing keyboard set-up. Or with the DK-70 strapped around your neck.

So go to your local Siel dealer for a truly mind-blowing experience. Check out the DK-70. You'll hear why more and more musicians are turning to Siel.

For our complete color catalog, send $1.00 to: Siel Keyboards, 3000 Marcus Ave., Suite 2W7, Lake Success, NY 11042. Or call: (516) 775-5510.

SIEL

SIEL IS TURNING HEADS.

GET ON TRACK!

Whether you're recording original music scores, layering up sound effects, or synchronizing to video or film for audio-post sweetening, you need a tape recorder that's built especially for your new and exciting business. The MX-70 is the perfect multitrack for the synthesizer oriented studio tied together with MIDI.

The "70's" three-way design gives you 7.5, 15 and 30 ips in a 1" 16-track, a 1" 8-track, or a 1" 8-track prewired for 16. (An optional ½" 8-track is also available.) Noiseless, gapless, punch-ins and punch-outs provide quiet, inaudible inserts into pre-recorded program material. The MX-70's wide dynamic range makes for quiet, clean recordings. In fact, you'll find the "70" at 30 ips is so quiet that noise reduction just isn't necessary. And to complete this perfect package, you can add an optional autolocator to the standard full function remote for complete session control.

So if synthesizers with MIDI, or SMPTE with film and video, is part of your business, check the specs and don't settle for less than the MX-70. Call your nearest Otari dealer for **"Technology You Can Trust".**

Otari Corporation, 2 Davis Drive, Belmont, CA 94002,
415/592-8311, Telex 9103764890

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