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VOLUME 1, NUMBER 12
JULY 1987

MUSIC MAKER PUBLICATIONS, LTD.,
Alexander House, 1 Milton Rd, Cambridge, CB4 1UY, England.
Tel. 011 (44) 223-313722
PAN: Musicmaker.

EUROPEAN AD MANAGER
Graham Butterworth

MUSIC TECHNOLOGY
MUSIC TECHNOLOGY is published monthly, 12 times per year, by Music Maker Publications. Offices at:
7361 Topanga Canyon Blvd.
Canoga Park, CA 91303
Tel. (818) 704-8777
PAN: Musictech.
Applications to mail at second-class postage rates is pending at Canoga Park, CA Postmaster: Send address changes to above address.

Cover photography *Sakamoto courtesy of 10 Records; Matthew Vosburgh (Downes)*

Bob O'Donnell. Rick Davies. Deborah Parisi.

OPENING UP THE BOX

IT CAN BE frustrating, getting to a point where you know just what to look for in an instrument. A lot comes from experience with equipment and knowing just what your needs are.

As we approach the summer NAMM show, I wonder what we'll be hearing about the instruments unveiled at the *last* show. Perhaps there will be more complaints about "no individual outputs", or about the size of the manual, or about how long it takes to load a disk.

But there's a deeper, more fundamental problem lying beneath those considerations. It's not easy for a musician to get into all of the technological music jargon from a cold start – there seems little point in trying to get from A to B if you can't even find A. And even when you *have* found your path through MIDI, samplers, and sequencer programs, there is still a learning curve which can leave you wondering why you ever decided to deviate from the piano.

It's not easy for manufacturers, either. Their instruments are expected to be easy to learn, yet adaptable to just about any situation imaginable at the same time (and at no extra cost, of course).

In the old days, a synth simply had "hidden functions" which could only be accessed if you knew where to look for them. Nowadays, a synth would have so many "obscure" functions by old standards, that it would be difficult to keep them hidden under its cool exterior. It's all very well to pack plenty of processing power into an instrument, but if you don't provide easy access to it, you're likely to limit its appeal.

And now that MIDI requirements are becoming increasingly complex, there also comes a point where each device's set of MIDI parameters needs to be carefully tailored to match other devices' parameters. The simple solution – implementing minimal MIDI – is really no solution at all, since it prevents you from gaining access to so many sophisticated applications. Yet ensuring easy *and* sophisticated communication between instruments is an expensive business, and will remain so until technology improves.

However, there is no conspiracy to prevent musicians from doing what they want or need. No one is trying to withhold the features you demand, and if you can't get what you want out of a $2000 synth, but can get it from a $4000 synth, then that's not to say that one manufacturer is ignoring your needs. It could just be that technology hasn't caught up with your requirements yet.

To us, it seems crazy that new instruments can produce never heard before sounds, and still get panned for not having multiple outputs, built-in signal processing, or an advanced MIDI implementation. The reason why instruments appear with that kind of specification is simply this: technology isn't at a stage where certain features can be implemented such that other, more common features are not compromised.

Sure, any number of features could be buried in "hidden functions," but at what cost? Will the designers have to disallow certain situations in order to avoid possible self-destruct? How much more are you prepared to pay for that extra yard of special functions? And how much longer will you wait while they implement one last feature?

It might *seem* from time to time that manufacturers ignore the needs of the "average musician," but they're often working with the best information they can gather, and always within the constraints of the current technology.

We're not trying to excuse all manufacturers for the occasional bug or design flaw, but it's worth remembering that an awful lot is expected of those little black boxes, and a lot of work goes into them. ■ *Rick Davies*

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Truly Professional

In the short time they have been available, Kawai's K3 and K3M synthesizers have been recognized by musicians and computer enthusiasts alike as truly professional musical instruments with sophisticated capabilities and warm, rich sounds.

The K3 keyboard and its companion K3M synthesizer module have been accepted by leading professional musicians such as Jan Hammer and Tom Coster. They find the K3's unique sound a perfect compliment to their existing electronic music systems.

Computer software companies such as Opcode, Dr. T's, Hybrid Arts, and Compumates also support the K3 with sound editing software for the Atari ST and 130 XE, Apple II and Macintosh, IBM PC, and Commodore 64 computers. These software packages allow graphic editing of the K3's unique programmable user-wave. They also have advanced voice editing and librarian functions that allow patches and user-waves to be easily accessed from disk and via modem.

So take a listen to the surprising Kawai K3 on the "Miami Vice" television show, on tour with Tom Coster and "Vital Information," or at your local authorized Kawai electronic musical instrument dealer.

Kawai America Corp.
Dept. EM
2055 E. University Dr.
Compton, CA
90224-9045

Kawai Canada Music Ltd.
Unit #1
6400 Shawson Dr.
Mississauga, Ontario
Canada L5T1L8

Jan Hammer

"When you think you've exhausted all possibilities of creating new sounds, Kawai brings out the K3M which enables me to use a whole range of sounds that compliment both my FM and analog instruments."

Tom Coster

"The Kawai K3 and K3M Rackmount is a powerful combination: analog punch and digital sounds of the 80's...a welcome addition to my keyboard setup. Thanks, Kawai!"

Tom Coster can be heard on two new CBS releases: 'Global Beat' Vital Information featuring Steve Smith and 'Songs of Freedom' Santana.

KAWAI
The Master Builder

NEWS DESK

CLARITY UPDATES EFFECTS AUTOMATION

The XLV, an effects automation interface from Clarity, is being updated to enable automation of the AMS RMX-16, the Quantec Room Simulator and the Yamaha REVI as well as the Lexicon 224XL and 480L and all voltage-controlled devices (eg. digital delay lines). The aim is to make the XLV compatible with nearly every high-end signal processor in the studio.

XLV features include factory presets and user programmability, eight control voltage outputs, full MIDI implementation with an

onboard processor, positive and negative scaling, a by-pass mode, and compatibility with the 480L. It has a capacity of 99 programs: 16 factory presets and 83 user programs.

MORE FROM Clarity, Nelson Lane, Garrison, NY 10524. Tel. (914) 424-4071.

KURZWEIL OFFERS ADDITIVE SYNTHESIS

Owners or potential buyers of the Kurzweil 150FS (Fourier Synthesizer) may be interested in the announcement that a new Sound Modeling Program (SMP) has been released which, when combined with the Apple IIe, will transform the preset synthesizer into a programmable synth.

Introduced in June '86, the Kurzweil 150FS has always had a range of resident instrument voices, including piano, harpsichord, four analog synthesizer sounds, acoustic and electric bass, acoustic guitar, clarinet, bells, vibes and marimba. The Sound Modeling Program will allow the user to view representations of these sounds – or synthesize new ones – on the screen of the Apple IIe. The frequencies and amplitude envelopes of up to 64 partials per sound model may be precisely selected and manipulated in a data table or graphic display form.

The 150FS has a pool of 240 independent oscillators called partials, each of which may be a sine wave or any of several types of noise. When a note is played, the necessary number of partials is assigned from the pool. Each oscillator's output is then given its own arbitrarily-shaped amplitude envelope, called a contour, which comes from the sound model memory. The Sound Modeling Program allows representation of these contours as a series of line segments where each break point has a time and a rate or amplitude slope.

The Sound Modeling Program is a free enhancement for the 150FS. It requires Version 1.6 software, which is also available as a free update.

MORE FROM Kurzweil Music Systems, Inc., 411 Waverley Oaks Road, Waltham, MA 02154. Tel. (617) 893-5900

TWISTER MIXER

Twister Engineering of Denmark has introduced their eight-channel MIDI-based computer mixer. The Twister Pac allows computer control of signal level, computer reading of input audio level, and status communications using MIDI protocol. The unit is also capable of dynamic control of levels, mutes and groups as well as an assortment of programmable gain modifiers such as noise gating, compression, and followers. Dynamic control information is sent over MIDI using "keyboard-like" data compatible with all MIDI sequencers.

A front panel control as well as Atari ST fader display allow the unit to be used for recording or live performance. Features include 100 snapshot memories, 64 groups per memory, a 95dB signal-to-noise ratio, expandability to 64 channels, and an eight-in, eight-out configuration. The list price is $1599.

MORE FROM Promise Productions, 1416 Dorothy Drive, Glendale, CA 91202. Tel. (818) 242-8582.

MIDI SUMMER NIGHT'S DREAM

Ensoniq has announced what looks to be an exciting promotional campaign just in time for your summer vacation. All you have to do is to visit your local Ensoniq dealer and test-drive one of the ESQIs between now and July 31. Fill out and return the entry form, and your name will be placed in a (probably large) hat. Three lucky winners will be whisked away to Bobby Nathan's Unique Recording Studios in New York City for an evening in the studio with Herbie Hancock.

MORE FROM Ensoniq, 155 Great Valley Parkway, Malvern, PA 19355. Tel. (215) 647-3930

QUAD MONSTER UPGRADES DX7

Monster Memory Co. has announced the arrival of Quad Monster, a small circuit board which replaces the ROM chips of a DX7 in order to expand its memory capacity, function storage, MIDI capability and actual sound-production potential.

New features and functions include internal sound storage of 512 sounds, function storage for each sound, programmable volume for each sound, and direct MIDI access to all sounds.

The new Play mode allows a sound to be layered upon itself up to four times, each layer fine-tunable over four octaves. During live performances, the Double and Quadro modes can be accessed in one move for lead solos or can be switched in by a sequencer over MIDI.

The list price for Quad Monster is $259, with a 10-day money-back guarantee.

MORE FROM Monster Memory Co., 5757 Kirk-wood Pl. N., Seattle, WA 98103. Tel. (206) 526-0540

MESA MEETS MIDI

Roland is looking to set new standards for easy-to-use software with the introduction of MESA (Music Editor, Scorer, Arranger), a software package designed for the IBM PC and compatibles. A fully integrated system with three modes (song, score and print), MESA has several features that today's MIDI users may be looking for in their sequencer programs.

The mouse-based user interface provides access to the functions, and music can be entered using any MIDI instrument, the mouse, or the PC keyboard. The song mode contains advanced sequencing functions (including a 65,000 note capacity), while the screen graphically displays the structure of a composition, allowing "cut and paste" editing and arranging.

Score mode displays phrases in standard musical notation on the screen, which are transcribed and displayed with automatic placement of notes, rests and beams. Complete compositions can be created and individual notes and MIDI events can be inserted, deleted and modified using the mouse or the PC's function and cursor keys.

The program requires an IBM PC, XT, AT or compatible with two floppy disk drives or hard disk, a minimum of 512K RAM, an IBM or compatible monochrome display system, parallel printer port, and a Roland MPU401 MIDI Processing Unit and interface card.

MORE FROM RolandCorp US, 7200 Dominion Circle, Los Angeles, CA 90040. Tel. (213) 685-5141

MIDI COMES TO USC

USC is offering a seminar in Electronic Music, Computers and MIDI from July 15 through July 19, 1987. The School of Music is hosting the five days of training, which will include hands-on experience in computer learning labs and electronic instrument work stations with a variety of microcomputer systems, music software and electronic instruments and MIDI hardware. In addition to the classes, there will be exhibits, forums and performances from USC faculty and industry representatives. An extensive collection of resource materials on MIDI technology will be available to seminar attendees for study and research purposes. And electronic jam sessions are scheduled in the evenings.

Instructors will include David Wheatley, Bill Biersach, Paul La Rose, Bill Alvis and Gilbert Blount of the USC Music Faculty, Clark Spangler from UCLA, film composer Gary Chang, and representatives from Fairlight, Yamaha, Synclavier, Simmons, MIDI Stick, ▶

▶ Bacchus, the International MIDI Association, Dr. T's Software, Club MIDI Software, Berklee College, and IBM. The sponsors are attempting to create an instructional atmosphere rather than a trade show environment.

The cost for the entire five days is $200; for 1 Unit Credit, $351.00. Auditors can opt to pay $50 per day.

MORE FROM USC at (213) 743-3958, or write: Seminar in Electronic Music, Computers and MIDI, School of Music, University of Southern California, Los Angeles, CA 90089-0851 (or PAN: Larose).

N E W S

3.0 SOFTWARE FOR AKG DIGITAL REVERB

AKG Acoustics, Inc. has announced the availability of Version 3.0 software for the ADR 68K Digital Reverb and Effects Unit. The new package was introduced at the AES Convention in London, and shipments to existing customers began in mid-April.

The improvements feature dual digital delay lines, multi-tap stereo processing, and multi-effects (a four-voice stereo chorus, a multi-tap delay line with stereo output, and a multi-tap delay line driving a stereo reverberator). A six-voice stereo chorus program, with controls for equalization, number of voices, voice delay, chorus rate and depth and automatic stereo panning is also included.

The ADR 68K can now sample in stereo – four seconds, or two seconds when used concurrently with other processing programs. New AKG digital reverb units will come with the 3.0 software as a standard feature. Owners of units with older packages can obtain the new software free of charge.

MORE FROM Dave Ogden, AKG Acoustics, Inc., Two Calvin Road, Watertown, MA 02172. Tel. (617) 924-7697

POSHEK PRESENTS TURBO TX81Z

Poshek Productions has just announced the availability of Turbo TX81Z, a memory resident editor/librarian for the Yamaha TX81Z module and IBM PC/XT and compatible computers. The program is said to allow complete editing of all parameters of the TX81Z.

The four main modes of the program allow for editing voice parameters, creation and editing of custom voice banks, and editing performance data. An active three-drive directory is accessible from each mode, and effects, micro tunings, system data and program change data can all be stored.

A demo version with the MIDI functions disabled is available for $5, refundable upon purchase of the program. The list price is $79.

MORE FROM Poshek Productions, 838 Van Dyke Drive, Laguna Beach, CA 92651. Tel. (714) 497-7210

SEQUENCER CAKEWALK

Twelve-Tone Systems has announced the release of Cakewalk, a sequencer package for IBM and compatible personal computers. Cakewalk features 256 tracks, low-level and global editing, DOS command shell, pull-down menus, on-line context-sensitive help, and more in a $150 package.

Each of the 256 tracks has its own name, play/mute switch, pitch and velocity transposition, and optional forced MIDI channel – all changeable during record/playback. Each track can hold any number of MIDI events bound for any number of MIDI channels.

Other features include aural editing with three programmable tempos, event filtering, an ASCII file conversion format and a flexible transport which allows you to move around your work at will.

MORE FROM Gregg Hendershott, Twelve Tone Systems, PO Box 226, Watertown, MA 02272. Tel. (617) 924-7937

SOLID SUPPORT STANDS UP

A new home keyboard stand from Solid Support Industries is now available which almost any home keyboard may be attached to with

EARLY TREMORS

Previews of Innovations to be Announced at the Chicago NAMM Show . . .

16-BIT SAMPLING COMES TO DRUMS

Forat Electronics will unveil its 16-bit sampling drum computer, the F16, at the June NAMM show in Chicago. The rack-mount F16 consists of a main controller which accommodates up to 16 modules. Rather than use one block of memory to store several samples, as is the norm on most samplers, each module stores a single sample in half a million bytes of memory. Thus, individual samples can be as long as six seconds at full audio bandwidth, or 25 seconds with reduced high-end response. For example, crash cymbals sampled at 50kHz play for up to six seconds.

Each F16 module has independent tuning, volume and pan controls, and is available through direct outputs or the controller's stereo mixer outputs. And each module features a large amount of memory – half a million bytes for each sound.

The F16 modules may be triggered using drum pads, pre-recorded drum tracks, a drum machine's trigger outputs or MIDI (which permits hookup to any MIDI sequencer). Prices start at under $2500.

MORE FROM Forat Electronics, 11514 Ventura Blvd., Unit 1, Studio City, CA 91604. Tel. (818) 763-3007

ROLAND RACKS

Roland will be introducing rack-mounted versions of their S50 digital sampler and D50 digital synth at the upcoming NAMM show in Chicago. Dubbed the S550 and the D550 respectively, the new units will include all the capabilities of their keyboard counterparts as well as the ability to operate in MIDI Mono Mode (Mode 4). While exact price information is not available yet, both promise to be a few hundred dollars less than their keyboard-equipped equivalents.

Roland will also be re-entering the pro audio market with the addition of two mixing consoles to their product line. The 16-channel M160 will be a rack-mountable, line level mixer and the non-rack mount M240, will have (any guesses?) 24 channels.

In other Roland news, the company will be sponsoring a D50 sound developers' workshop at the NAMM show. The workshop will be of a technical nature, with specific discussions planned on the new L/A synthesis method (as implemented in the D50), and the voice architecture of the popular new instrument.

Product specialist Eric Persing and some of Roland's engineering staff will be on hand to answer questions. In addition, the company will outline its D50 sound developers' support plan. The meeting is scheduled to take place on Monday, June 29, from 6-7.30pm in the Roland display area (the Lindheimer Room of McCormick Place).

MORE FROM Roland Corp US, 7200 Dominion Circle, Los Angeles, CA 90040. Tel. (213) 685-5141

DAWN OF THE EIII?

First there was the eight-bit Emulator. Then there was the EII with its polyphonic sequencer, individual outputs, and memory expansions and extensions, followed by the low-cost Emax. Now E-mu's EIII outdoes them all, with 16-bit stereo sampling at the center of all the commotion. What else could E-mu Systems call their newest, and most impressive sampling keyboard? (Don't answer that; there was a pattern emerging for a moment.)

the use of 2" velcro strips. This stand is 27¾" high by 25" wide, with two cross braces which allow even small keyboards to be supported at the correct playing height. Some keyboards may be bolted directly to the stand through slots provided in the top cross-support.

The stand is constructed of ¾" steel tubing and weighs 12 pounds. The list price is $59.95.

MORE FROM Solid Support Industries, 2453 Chico Avenue, So. El Monte, CA 91733. Tel. (818) 579-6063, or out-of-state Tel. (800) 782-6377

MIDI PERCUSSIVES

Electronic Percussion Through MIDI, published by RolandCorp US, is a new guidebook for drummers and percussionists interested in the expanding world of MIDI percussion. The book presents an overview of Roland electronics percussion products with photos, product descriptions, various system configurations, and actual demonstrations as well as complete songs on a cassette tape which accompanies the textbook.

The guidebook is intended to give a basic working knowledge of a MIDI percussion system and provide suggestions for designing a personalized electronic percussion set-up tailored to your individual musical needs and budget.

The guidebook/cassette tape package for drummers is available through authorized Roland dealers or from RolandCorp US, at a suggested list price of $10.

MORE FROM RolandCorp US, 7200 Dominion Circle, Los Angeles, CA 90040. Tel. (213) 685-5141

SHURE ANNOUNCES COMPACT PA

Shure Brothers, Inc. has announced the introduction of the Shure Audiomaster, a compact modular sound reinforcement system. Components in the Audiomaster system include the Model 1200 Expandable Powermixer, the Model 3200 Loudspeaker, the Model A1200MX Expansion Module and the Model A1200C Portable Case.

The Audiomaster 1200 Powermixer is a 200-watt, six-channel powered mixer that is expandable to eight or 10 channels with the installation of expansion modules. The 1200 features color-coded controls and LED indicators for various functions. Most microphones can be used with the unit, which includes both transformer-balanced low impedance pin XLR connectors and ¼" unbalanced high-impedance phone jacks for each input channel. The unit is surrounded by steel casing and includes RF protection circuitry to prevent RF interference.

The Audiomaster 3200 Loudspeaker system features the Time Sync crossover network for time-corrected phasing, a constant directivity horn, a compression high-frequency driver, and a 12" low-frequency driver with a 2½" high-temperature aluminum bonded voice coil and a 3½ pound magnet.

Audiomaster prices are: $930 for the Powermixer, $450 for the Loudspeaker, $183.35 for the expansion module, and $70 for the portable case.

MORE FROM Shure Brothers, Inc., 222 Hartrey Avenue, Evanston, IL 60202-3696. Tel. (312) 866-2553

N E W S

Needless to say the initial specs are quite impressive: 16 voices, 16 individual mono-phonic outputs (in addition to two stereo outputs), 16-bit linear sampling, 35.5 secs of monophonic sampling or 17.5 seconds of stereo sampling at a 30kHz sample rate with the standard two megabytes of internal RAM (expandable to 8Meg when new RAMs become available), S/N ratio of greater than 90dB, 40Meg internal hard drive (upgradable to 80+), 3.5" floppy disk drive, built-in sixteen-track sequencer (with advanced editing features including cut and paste), SMPTE cue sheet capabilities, three envelope generators per voice, high-quality VCAs and VCFs, an external SCSI port, an RS422 port and separate DACs for each voice. The EIII price has not been announced, and the instrument will be ready in late fall.

E-mu will also be introducing the SP1200, which is a souped-up version of the SP12 with a built-in 3.5" disk drive and increased memory for up to ten seconds of sampling time. The price of the new drum machine has not yet been determined, but sources at E-mu say that it will be less than a Turbo SP12.

The Emax HD, yet another new entry from E-mu, will be an Emax with a built-in 20Meg hard disk drive. With the new drive, which holds the equivalent of about 35 disks, the load time for samples will be reduced to approximately three seconds. A rack-mount version with a hard disk, as well as retrofits for existing machines will also be available by mid-summer.

MORE FROM E-mu Systems, 1600 Green Hills Road, Scotts Valley, CA 95066. Tel. (408) 438-1921

Dr. T GOES BLANK

Word from Dr. T's is that they have acquired exclusive rights to distribute Blank Software's Sound Lab program (for the Macintosh) and Sound File (for the Mac or Commodore 64). Version 2.2 of Sound Lab (originally designed for the Ensoniq Mirage) now features the MIDI Sample Dump Standard, making the program attractive to owners of other samplers.

Also coming at NAMM is their Esq'apade ESQ1 editor for the Atari ST, and the VDS Mirage editor (also for the ST) which provides FM and additive synthesis tools for editing existing sounds, or for making new samples.

MORE FROM Dr. T's Music Software, 66 Louise Rd., Chestnut Hill, MA 02167. Tel. (617) 969-6657

AURAL EXCITERS RIDE AGAIN!

Another product being introduced at the NAMM show, this one from Aphex Systems, is the newest Aural Exciter, the Type E. Designed expressly for the performing musician, instruments or mikes can be plugged directly into the Type E for stage, recording and PA use without the necessity of a preamp or mixer. The Type E can serve as a low-noise preamp and direct box as well as a sound enhancer.

The Type E features "High Z" ins and outs, plus line level ins and outs, and is intended to be useful at home, on stage and in the studio. Optional rack mounts are available for mounting one or two units in a standard 19" equipment rack.

Like all Aural Exciters, this new model generates musically-related harmonics in an effort to restore natural clarity, detail and brightness. According to Aphex, the effect is especially helpful for digital audio effects, samplers, and synthesizers because it helps counter some of the problems associated with digitally reconstructing a sound.

Suggested list price of the Type E is less than $200.

MORE FROM Aphex Systems Ltd., 13340 Saticoy Street, North Hollywood, CA 91605. Tel. (818) 765-2212

READERS' LETTERS

Your questions answered by Music Technology's team of experts. If you have a query about any aspect of music technology, or some information that might be useful to other readers, write to Readers' Letters, Music Technology, 7361 Topanga Canyon Blvd., Canoga Park, CA 91303.

Dear Music Technology,

I have been looking for a CD with sampled sounds on it for some time. The photo that accompanied your May article on the Casio SK2100 seems to have one pictured in the lower left corner. If at all possible, could you let me know where I can obtain information on it?

Jeff Newman
Smyrna, GA

The CD in the photo is, in fact, exactly what you (and probably hundreds of other people) are looking for. Entitled "Sampling Collection, Vol. I", it is available from Korg, or any Korg dealer, for $41.95. It contains 189 studio-quality samples of 34 acoustic instruments (most of which have multiple samples), including piano, strings, brass, woodwinds and percussion.

Dear Music Technology,

I recently read your May issue and tried to make sense of many of your articles. The reason they didn't make sense was because I don't know the definitions of such terms as MIDI, sampler, sequencer and the roles that MIDI plays in interfacing with computers or other instruments. Can you help me with the definitions of the terms that are used in your magazine? Thanks.

Bronko Gligich
Santa Ana, CA

Take heart, Bronko. At some point we were all in the same confusing position in which you find yourself right now – caught up in a mass whirlpool of technological terminology which seemingly leads nowhere. There are a number of books available on the different technologies which you might look for, but we would also suggest that you begin haunting music stores, talking to fellow musicians and (of course) reading Music Technology. Once you discover which specific areas of technology best apply to your own musical needs, there are various users' groups which can help. Most of all, hang in there! The effort will pay off in superior music capabilities.

Dear Music Technology,

Ha. Ha. Your article, "Heads, Hands, and Feet" (MT, April '87) mentioned the Zlatna Panega cooperative. I swear it's all a joke. I don't know. Do they really manufacture such wonderful interfaces? If so, would you give me their addresses? Your report mentioned that the UK edition of ***Music Technology*** *covered this; can you give me the magazine dates?*

Is it true Dimitrov can't figure out what to do with it? I have a number of good ideas, but I have to know if the reportage is fact or fiction.

Wind O'Neal
Arnoldsville, GA

Boy, we certainly can't fool some of you out there. Wind, here, has obviously caught on to some of the more esoteric developments that are being worked on by the famous Bulgarian MIDI engineers, Boris Dimitrov and Harri Limeski. Unfortunately, no further details have reached us since we reported the initial developments and we have a sinking feeling that we may not hear from them again until around the 1st of April, 1988. (Iron Curtain nations only allow sensitive information to be leaked every so often, you know.)

Dear Music Technology,

I recently read about the Audio Optics QED pickup in your April '87 issue. Can you send me any info on this company or product? Is it a pitch-to-MIDI converter for guitar?

Also, can you send me any info on the Vortex MIDI converter mentioned in the same article. I am looking for a pitch-to-MIDI converter that will work with a nylon string guitar.

Any information on the above would be most appreciated.

Julian Catford
Seattle, WA

We often receive requests for specific information about products (ie. product brochures) and companies but we don't have that kind of info on hand (except for our file copies), so if you're looking for specifics we suggest that you contact the manufacturers directly. (We always put the address and phone numbers of manufacturers at the end of reviews for that very purpose.) In case you lost the issue with the products in question, here are the appropriate addresses and phone numbers: Audio Optics, PO Box 691, Santa Barbara, CA 93102, ✆ (805) 563-2202; Beetle (Vortex), 130 N. Victory Blvd., Suite 101, Burbank, CA 91502, ✆ (818) 841-9922.

To answer your questions, the QED is not a MIDI guitar converter, but rather, a sophisticated audio pickup system. The potential is there to make it part of a guitar-to-MIDI conversion system, but as of yet, such a system does not exist. As far as a nylon-string guitar converter goes, we suggest you look at the Ovation and Takamine GTM6 acoustic guitar-to-MIDI systems, which are available from Kaman Music Corp., PO Box 507, Bloomfield, CT 06002. ✆ (203) 243-7941. The Photon System, which is now being distributed by Gibson Guitar Corp., can also be used with nylon string guitars. Information is available from Gibson's parent company, Phi-Technologies, 4605 N. Stiles, Oklahoma City, OK 73105. ✆ (405) 521-9000. Roland's new GM70 converter can be used with acoustic guitars as well, but they must have metal strings. ■

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ROM 1 (BANK 3)	ROM 2 (BANK 4)	ROM 3 (BANK 5)	ROM 4 (BANK 6)	ROM 5 (BANK 7)
1 Brass	1 Up Piano	1 Horn 2	1 Fnk Syn 2	1 Organ 1
2 Horn	2 SPiano	2 Horn 3	2 Fnk Syn 3	2 Organ 2
3 Trumpet	3 Piano 2	3 Horns	3 Syn Organ	3 COrgan 1
4 Lo string	4 Piano 3	4 Flugelh	4 Syn Feed	4 COrgan 2
5 Strings	5 Piano 4	5 Trombon	5 Syn Harm	5 EOrgan 3
6 Piano	6 Piano 5	6 Trumpet 2	6 Syn Clar	6 EOrgan 4
7 New EP	7 Ph Grand	7 Brass 2	7 Syn Lead	7 EOrgan 5
8 EGrand	8 Grand	8 Brass 3	8 Huff Tak	8 EOrgan 6
9 Jazz Gt	9 Dp Grand	9 Hard Br 1	9 So Heavy	9 EOrgan 7
10 EBass	10 LPiano 1	10 Hard Br 2	10 Hollow	10 EOrgan 8
11 Wod Bass	11 LPiano 2	11 Hard Br 3	11 Schmooh	11 Steel Pipe
12 EOrgan 1	12 LGrand 2	12 Hard Br 4	12 Mono Syn	12 Mid Pipe
13 EOrgan 2	13 Honkey 1	13 Huff Brs	13 Chirpy	13 Big Pipe
14 POrgan 1	14 Honkey 2	14 Perc Br 1	14 Syn Bell	14 Sft Pipe
15 POrgan 2	15 Pf Bell	15 Perc Br 2	15 Syn Plnk	15 Organ
16 Flute	16 Pf Vibe	16 String 1	16 EBass 3	16 Guitar
17 Piccolo	17 New EP 2	17 String 2	17 Rub Bass	17 Folk Gt
18 Oboe	18 New EP 3	18 String 3	18 Sid Bass	18 Pluck Gt
19 Clarinet	19 New EP 4	19 String 4	19 Pluk Bas	19 Brite Gt
20 Glockens	20 New EP 5	20 Solo Vio	20 Upri Bas	20 Fuzz Gt
21 Vibes	21 EPiano 1	21 Rich St 1	21 Fretles	21 Zither 2
22 Xylophn	22 EPiano 2	22 Rich St 2	22 Flap Bas	22 Lute
23 Koto	23 EPiano 3	23 Rich St 3	23 Mono Bas	23 Banjo
24 Zither	24 EPiano 4	24 Rich St 4	24 Syn Bas 1	24 Sft Harp
25 Clav	25 EPiano 5	25 Cello 1	25 Syn Bas 2	25 Harp 2
26 Harpsic	26 High Tin	26 Cello 2	26 Syn Bas 3	26 Harp 3
27 Bells	27 Hart Tin	27 Lo Sng 3	27 Syn Bas 4	27 Sft Koto
28 Harp	28 Perc Pf	28 Lo Sng 4	28 Syn Bas 5	28 Hit Koto
29 SmallSyn	29 Wood Pf	29 Lo Sng 5	29 Syn Bas 6	29 Sitar 1
30 Harmoni	30 EP String	30 Orchest	30 Syn Bas 7	30 Sitar 2
31 Steel Dr	31 EP Brass	31 Sth Str	31 Marimb 2	31 Huff Syn
32 Timpani	32 Clav 2	32 Picnic 1	32 Marimb 3	32 Fantasy
33 Lo Sng 2	33 Clav 3	33 Picnic 2	33 Xylopn 2	33 Synvoice
34 Horn Lo	34 Clav 4	34 Flute 2	34 Vibe 2	34 M. Voice
35 Whistle	35 Fuzz Clv	35 Flute 3	35 Vibe 3	35 VSAR
36 Zing Pip	36 Mute Clv	36 Flute 4	36 Crocken 2	36 Racing
37 Metal	37 Mute Cl 2	37 Pan Flt	37 Tube Bel	37 Water
38 Heavy	38 Syn Clv 1	38 Slow Flt	38 Tube Br 2	38 Wild Wir
39 Funk Syn	39 Syn Clv 2	39 Sth Flt	39 Bells 2	39 Ghostie
40 Voices	40 Syn Clv 3	40 Oboe 2	40 Temple G	40 Wave
41 Marimba	41 Syn Clv 4	41 Bassoon	41 Steel Dr	41 Space 1
42 EBass 2	42 Harpsi 2	42 Reed	42 Elect Dr	42 Sp Chime
43 Snare Dr	43 Harpsi 3	43 Harmon 2	43 Hand Dr	43 Sp Talk
44 RD Cymb	44 Harpsi 4	44 Harmon 3	44 Syn Timp	44 Winds
45 Tom Tom	45 Harpsi 5	45 Harmon 4	45 Clock	45 Sounds
46 Marmbo	46 Circust	46 Mono Sax	46 Heifer	46 Alarm
47 Storm	47 Celeste	47 Sax 1	47 Snare Dr	47 Helicop
48 Windbel	48 Squeeze	48 Sax 2	48 Snare Dr	48 Sin Wave

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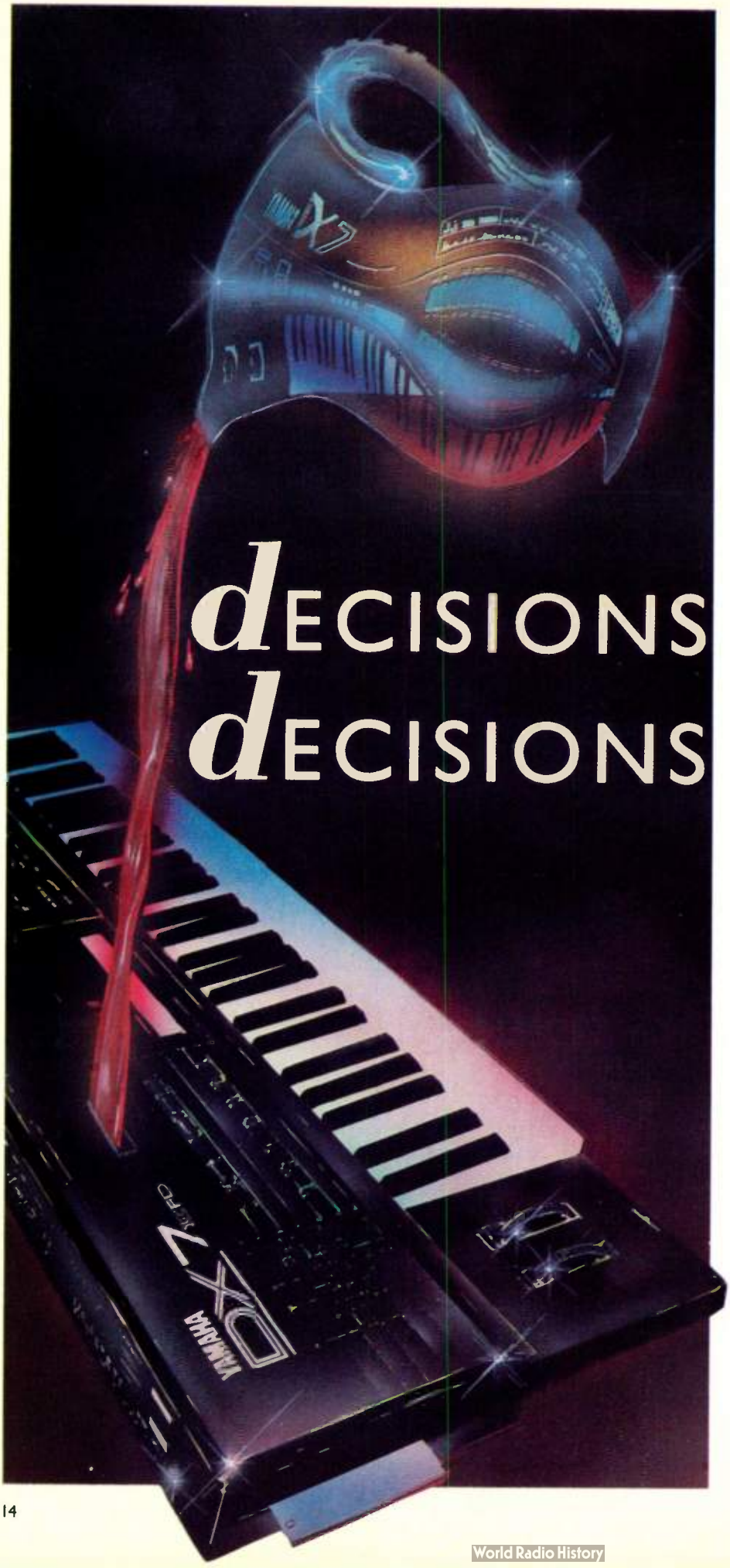

Illustration *Stuart Catterson*

The new DX instruments have a number of features which the old ones lacked, but if you already own an "old" DX7, the important question to ask is, are the new features enough to justify a new purchase? Compatibility may be the key.
Text by Howard Massey.

DX7 OWNERS EVERYWHERE face a tough decision these days: upgrade to the newer DX7IID or DX7IIFD (the difference being simply that the latter contains a built-in 3½" microfloppy drive for data storage) or stick with Old Reliable?

Compounding this dilemma is the fact that many DX7 owners may just be starting to feel a little more comfortable with the gremlin of actually programming their instrument and/or modifying presets. The word on the street is that there was initially a good deal of resistance by DX7 owners to the idea of buying yet another, even more complicated DX instrument, but the fact remains that many important improvements have been made in the second generation DX7II.

So the question remains: should you or should you not lighten your wallet by buying one? Here are some important facts that may help you make this decision.

The Improvements

FIRST OF ALL, let's discuss some of the improvements that everyone has been talking about. Probably the most important one is the fact that the DX7II, unlike its predecessor, is multi-timbral. It can make two sounds at once – sort of like the DX5 (or a DX7 and TX7), but better, because there are many more possibilities for interaction between the two sounds. Like the DX5 (or most bitimbral synthesizers, for that matter), you can split the keyboard or layer the two sounds over the entire keyboard. Unlike the DX5, however, you can actually edit either sound while hearing both in this layered keyboard mode (called Dual mode) – meaning that you can really approach the DX7II as a 12-operator digital FM synth. This is great for those of you who may think you have explored all the possibilities of creating sounds with six operators (and, honestly, I haven't run across anyone yet – including myself – who has).

The interactivity between whichever two sounds you call up at any given time goes well beyond this, however. The DX5 (and the DX7/TX7 combination alluded to earlier) allow you, of course, to send the audio outputs to separate channels, making for a pleasing stereo effect. The DX7II takes things a step beyond, by offering programmable *panning* of the outputs. This is an area of the instrument that I am personally very excited about, because it takes what is normally a very directional, focused sound and allows you to turn it into an ambient sound – almost like

SurroundSound – all from the DX7II's front panel. If you normally play your instrument through a single amplifier/speaker combination, this will have little impact. But if you work in the studio with an engineer who is smart enough to record you in stereo, it means a whole lot! Taking the directionality out of the sound means a whole new ball game as far as the kinds of subtle effects you can create. The panning implementation in the DX7II goes far beyond what I have seen in any other synthesizer in this price range. You can pan by key velocity, by note number, or with an LFO – and there's even an envelope generator which is totally dedicated to this function. Furthermore, you can shift the mixed signal of your two patches, or you can alter the output level of one and not the other, or you can alter the output level of both – in a complementary fashion so that as one increases, the other decreases proportionally.

The second most striking improvement in the DX7II is that the sound quality of this instrument is much higher than that of the DX7. You know all of that digital fizz that accompanies so many DX7 patches? Well, it won't be gone completely in the DX7II, but it will be much, much quieter and virtually unnoticeable in 90% of your patches. The actual clarity of the sound has also been improved, thanks to 16-bit digital-to-analog conversion and higher-speed processing. Not only are sounds crisper and more defined, I find the bass end to be improved a great deal as well. This, along with the many different detuning options offered by the DX7II, allow you to create those big, beefy sounds (hold the onions, please) that DX7 owners everywhere have cried out for. Well, maybe I exaggerate a bit – but there is no question that the DX7 carried the often unjustified label of being "thin-sounding". Have no fear – this is something the DX7II absolutely cannot be accused of.

Another big improvement is that the "function controls" of the DX7 are, happily, no more. Instead, all of these parameters (which include things like pitch-bend wheel assignment, portamento, and real-time controller routings) are, in the DX7II, voice edit parameters. This means that you program them individually *for each voice*, and they will change per patch just as surely as operator output levels will. At long last, you can customize your controller routings for the particular sound you've created. What this really means is that there is virtually no need to alter anything about a sound in live performance – you really can set everything up in advance.

Speaking of controllers, it's worth noting that the DX7II allows the use of many more controllers than does the DX7. You can have two sustain pedals, for example, as well as two foot controllers. One of the sustain pedals can take on some pretty nifty functions, like acting as a soft pedal, or duplicating the effect of a piano *sostenuto* pedal. There are also a couple of continuous sliders on the front panel of the DX7II (an idea borrowed from Yamaha's MIDI keyboard controllers like the KX88 and KX76). These allow you to actually alter voice edit parameters (things like EG settings, operator output level or frequency, or even the algorithm) directly from play mode. Furthermore, the movement of these sliders is transmitted via MIDI, so that they can be used to alter various continuous controller parameters in slave synthesizers. You can also link these via MIDI to controllers from other master keyboards (assuming you're using the DX7II as a slave instrument), so that, for example, moving the modulation wheel of your Emulator changes the LFO speed in your DX7II – or almost anything else about the DX7II patch or patches called up!

"You can actually edit either sound while hearing both in the dual mode, meaning that you can really approach the DX7II as a 12-operator digital FM synth."

Ah, MIDI. Here's an area where DX7 owners have been deprived for years. In Yamaha's defense, it should be pointed out that the original DX7 was designed in 1983 – prehistoric days, by MIDI standards. Well, the DX7II moves into the modern MIDI age, with all the bells and whistles you can hope for. In fact, MIDI is so well implemented that this instrument is actually well-suited for use as a master keyboard controller, something that the original DX7 definitely could not claim. In fact, the keyboard itself has a better "feel" than the DX7 keyboard.

Got some MIDI questions? We got some answers: Yes, you can transmit on any channel (though, unfortunately, you can't transmit on more than one channel). Yes, the two patches can receive on the same or different channels. Yes, you can receive in Omni mode. Yes, Local On-Off is supported (meaning that you can disconnect the DX7II keyboard from its own sound-generating circuitry). Yes, all 127 velocity increments are transmitted. Yes, volume can be externally controlled. Yes, program changes can be *prevented* from being transmitted (the DX7 always sent them) and you can even re-map these program changes (so that, for example, calling up patch #22 on your DX7II causes it to send out a "change to patch #47" command to slave synths). You can even transmit the contents of the DX7II edit buffer via MIDI – meaning that you don't have to necessarily store sounds in memory before you can ship them over to your patch editor. In short, you want MIDI – you got MIDI.

"There was initially resistance by DX7 owners to the idea of buying yet another, more complicated instrument, but important improvements have been made in the second-generation machine."

Speaking of memory storage, there's a lot more of it in the DX7II than in the DX7. Twice as many voices, to be exact, can be stored in the internal memory, along with 32 performance setups, a couple of micro tunings (more about this below), and the MIDI setup. The same amount of data can be stored in the DX7II RAM4 cartridges, which are larger – both physically and in memory capacity – than the DX7 RAM cartridges. These can also be used to store micro tunings alone, or data for an exciting new function called Fractional Scaling. Those of you who ever used the DX7's keyboard level scaling function will be interested in this. In a nutshell, keyboard level scaling allows you to increase or decrease an operator's output level (thus causing a change to the volume or timbre of the sound) as you play different areas of the keyboard. The original DX7 used a system of preset curves to initiate these changes. This system is implemented here as well (the DX7II, as you are no doubt aware, is upwardly compatible – for the most part – with the DX7, as we'll see shortly), but the DX7II takes things a step further with the introduction of this fractional scaling parameter. The fact of the matter is that the so-called "curves" used by DXs for level scaling are not curves at all. Instead, the output levels of various operators are altered in groups of three semitones at a time according to one of four preset tables. Fractional scaling allows you to dig right in and change this table, or even to create your own tables. This gives you much, much finer control over changing volume or timbre on areas of the keyboard, and also allows you to create true hard splits within a single voice –something that becomes, admittedly, less vital when you are working with a multi-timbral instrument that already has a split keyboard mode.

If you go for the "FD" model of the DX7II (about $400-500 more), you'll also have microfloppy disk storage at your disposal. Beyond holding DX7II data, this drive can also be used for generic MIDI data storage. The maximum file length of 20 kilobytes makes its use somewhat limited, but I am told that Yamaha plans to increase this maximum file capacity in an upcoming ROM update.

The much-vaunted micro tuning capabilities of the DX7II – none of which were implemented in the stock DX7 – allow you to do all kinds of weird and wonderful things to the intervals between semitones. You can create microtones, macrotones, inverted keyboards, or "dead" areas of the keyboard where the pitch remains the same no matter what note you play (great for percussive patches). You can also work with any of the 11 tuning presets (which include those created by such luminaries as Kirnberger and the unforgettable Vallotti and Young –weren't they a song-and-dance team in the fifties?), though we have become so accustomed to equal temperament in modern music, it remains to be seen if these alternate tunings will be used a whole lot. Still, hats must go off to Yamaha for at least offering the user the option of getting away from the norm.

There are many, many other improvements that are less significant, but may nonetheless be important to you. For one thing, those horrible membrane switches of the DX7 have ▶

▶ been replaced by real switches –some of which are even equipped with LED status lights. The LCD is much larger, allowing for two lines of 40 characters (as opposed to the two lines of 15 characters in the DX7), so that you can see much more data at any given time. For example, you can view all the EG settings for a particular operator in one single display, without having to cycle through all the rates, then all the levels. Cursor switches, similar to those used in the DX5, allow you to get around from parameter to parameter within the display. The LCD is also back-lit, making it much easier to use on stage. The master tuning display now shows you numbers – so you can actually see if you're at A440 or a bit sharp or flat, though as with the DX7, there is still no reference A440 tone. Physically, the DX7II is just a little bit smaller (though not so much smaller that it won't fit perfectly in your existing flightcase) and a little bit lighter than the original DX7, but appears to be just as roadworthy.

Compatibility

AS MENTIONED ABOVE, the DX7II was purposely designed to be upwardly compatible with the DX7. This means that it can do everything the DX7 could, and, really, a whole lot more. What will be important to the DX7 owner who has invested a lot of time and possibly money into creating a library of sounds is that all DX7 sounds can be played by the DX7II – and will, in most instances, sound exactly the same (or slightly better, due to the higher bit resolution and faster processing times). This is accomplished in one of two ways. The first method involves simply shipping voice data from a DX7 to the DX7II via MIDI (the DX7II will respond to the DX7 SysEx ID code). You then, of course, have the option of storing that received data in internal, cartridge, or disk memory. The second method involves the use of a special adapter (the ADPI, available from your nearest friendly – or unfriendly, as the case may be – authorized Yamaha dealer). This adapter sits in the cartridge slot of the DX7II (which, you will remember, is larger than that of the DX7), and your DX7 RAM cart is then placed in the adapter. This will allow you to directly access whatever patches are stored in the cartridge. One thing to be aware of – and that is that this is *not* a two-way street. The DX7II is compatible with the DX7 – but it is specifically *upwardly* compatible. That means you cannot send DX7II patches to a DX7 – it simply won't understand them, because the System Exclusive code is entirely different and contains much more data. More importantly, that DX7 RAM cartridge that is sitting in the adapter in the DX7II cartridge slot has now become a ROM cartridge – because you cannot write DX7II data to it.

Bear in mind, also, that in many instances you may need to tweak a couple of parameters in order to get the DX7 voice to sound exactly the same once it is in the DX7II. There are really only four areas of slight incompatibility, so let's take them one at a time.

Function controls – As we've seen, these no longer exist in the DX7II. Instead, they have grown somewhat in complexity and evolved into standard voice edit parameters. When you ship a DX7 voice into the DX7II (either via MIDI or via the ADPI cartridge adapter), no function control parameters are transmitted, simply because they are stored in a separate, "non-MIDI" area of the DX7 memory. Therefore, these particular parameters will all reset to initialization defaults in the DX7II, which simply means that everything will essentially be off (except the pitch-bend wheel, which will default to a range of two semitones). So, if you created a DX7 sound with a specific controller in mind to, let's say, route an LFO signal, you'll need to put the DX7II into edit mode and set up that routing again from scratch. The big difference is that, this time, when you store the sound, the controller routing will be stored with it as well. Therefore, the rule of thumb here is that *all* DX7 function controls that you may have used will need to be re-entered into the DX7II – a tedious if not particularly difficult process. Besides the old function controls, the DX7II also contains many new edit parameters, such as the new Pitch EG parameters discussed below. All of these parameters which don't exist in the DX7 will simply assume initialization defaults when a DX7 voice is sent to the DX7II.

"Bear in mind that in many instances, you may need to tweak a couple of parameters in order to get the DX7 voice to sound exactly the same once it is in the DX7II."

Amplitude modulation sensitivity – In the DX7, this parameter has a range of 0-3, with 3 representing maximum sensitivity to both LFO and EG bias modulation. In the DX7II, the range of this parameter has been increased to 0-7. Therefore, when shipping over DX7 patches that use either EG bias or LFO amplitude modulation, you'll have to retweak this parameter. Operators which had a sensitivity of 3, for example, will still have a sensitivity of 3 – but "3" will no longer mean "maximum".

"The DX7II is compatible with the DX7 – but it is specifically upwardly compatible. That means that you cannot send DX7II patches to a DX7."

Pitch EG – The DX7II offers several improvements in the pitch EG, including rate scaling, a keyboard velocity control, and the addition of a variable Range parameter. In the DX7, the (non-adjustable) range of the pitch EG is always eight octaves (four sharp and four flat). When shipping DX7 voices over to the DX7II, the Range parameter will automatically set itself to eight octaves, so it seems like everything should work perfectly, right? Wrong. The problem is that the absolute speeds of movement within the pitch EG change (even though the rate values remain the same) when the Range is altered – just as operator EG speeds change as output level is altered (as the envelope "squeezes down", it takes less time to get from level to level). Unfortunately, a DX7 pitch EG rate of, say, 50, is the same speed as compared with a DX7II pitch EG rate of 50 only when the Range is *one octave*, not eight. Therefore, you will find that all of your DX7 pitch EG rates are considerably faster when the sound is exported to the DX7II. This will have little effect on pitch EG rates which are quite fast, but can be drastic for those which are quite slow. For example, a favorite trick of mine when synthesizing "acoustic-type" sounds on the DX7 is to put in a very slight pitch EG movement at an extremely slow rate (for example, traveling from an L2 of 50 to an L3 of 45, at an R3 of 0). This adds a barely perceptible instability to the sound – and slight instability is a hallmark of virtually all acoustic sounds. The first time I shipped some of these patches to the DX7II, I was shocked to hear what sounded like the instrument becoming violently ill. The R3 of 0, which was really too slow to hear clearly in the DX7, was now a very obvious slow drop in pitch. The solution was simply to change the Range to one octave, and to alter the levels accordingly. The bottom line is that you may well have to make similar kinds of alterations to patches which use the pitch EG.

Delay effects incurred by operator envelope generators – One of the advanced tricks sometimes used by DX7 programmers is to build in "delays" by cleverly taking advantage of a deliberate anomaly in the DX7 operator EGs. As is shown in my first book, *The Complete DX7* (plug, plug), when traveling from a level of below 20 to another level of below 20 (in other words, when traveling at inaudible EG levels), the rates of movement act as digital delay lines. This works in the DX7 even if you are traveling between two inaudible levels that have the same values. In the DX7II, however, the EG logic has been changed somewhat (possibly because editing changes made to DX7II EG levels are assimilated by the microprocessor in real time). Here (as is shown in *The Complete DX7II*, plug, plug, *plug*), there will be no delays incurred whatsoever when traveling between two levels that are exactly the same. Therefore, if you try to play a DX7 sound (like the "Water-garden" ROM preset) that uses this trick in the DX7II, you won't hear the delays. All you'll need to do to fix this is to simply alter one of the two levels so that it's slightly different – even a single increment will work – from the other.

Conclusions

SO – TO DX7IID or not to DX7IID? That is the question. With luck, this article helped to provide you with an answer. The deciding factor will simply be how many of these new features are important to you in your own application. It's a tough decision – but one that Yamaha has made easier for you by deciding to implement upward compatibility. My opinion? I love the new instrument – and I still love the old one. ■

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THE COLLECTOR

MUSIC

Geoff Downes, once one half of the Buggles, then a respected progressive rock keyboardist with Yes and Asia, and now a solo artist with a forthcoming album, owns one of the biggest synthesizer collections in the world. How has he acquired them, and does he still find a use for them all?

Interview by Paul Wiffen; photography by Matthew Vosburgh.

SO THERE WE WERE . . . An equipment-laden photographer and I, walking innocently into London's Townhouse Studios. Suddenly, we were surrounded by keyboards. They filled the control room as well as the studio floor. They were standing against the walls, on the floor, racked up in threes and fours.

We soon gave up trying to count them, and settled down to the serious business of trying to cover them all in our discussions and photographs. Bear in mind that what we have here is the pick of 8000 words of interview transcription, and no fewer than 10 rolls of film . . .

I suppose, to be logical, we should talk about the first keyboard you ever owned.

Well, that's one of the few I don't still have. It was a Farfisa . . . No, before that there was a Vox Jaguar, which cost me $60 (about $100) when I was 13. That was an absolute fortune in those days. But I soon graduated to a Farfisa Compact Duo and then eventually bought my first Hammond when I was 16, a J122. The first two are long since gone (they're the only two keyboards I've ever sold), but the Hammond still exists today. It still works but it's been cut down and modified by Bill Dunn, London's longest-established organ customizer. It's now just a top keyboard and a box with all the gubbins inside it, to make it more transportable.

So you started gigging on the organ, rather than the electric piano?

Well, there was actually one other keyboard I sold, a Hohner Pianet, which was the first electric piano I had. Of course, I'd been playing piano for years, but acoustic piano was a bit inconvenient for gigs. Having studied classical organ as well, the portability of electronic organs made them the ideal main instrument.

In those days, it didn't matter what keyboard you had; if you could get it to the gig, you were in. Having a Compact Duo was the equivalent these days to a rack of DX7s, and the Hammond was really upmarket, you know, like having a Fairlight or Synclavier now.

The electric piano started to emerge as a viable alternative to the organ in the late '60s, like on the Doors records or the Zombies or 'I Heard It Through the Grapevine.' That was the new sound – if you had that then you were really cool, which is what inspired me to get something that sounded vaguely like that. The organ was great for covering orchestral lines; you could get that big sound out of it, but you needed the percussive edge for those sort of cover versions.

So I got the Pianet, which was an absolutely dreadful machine. It worked by plucking the strings with sticky pads. Mine just wore out, because the pads started getting less and less sticky as the dust got in.

When the Pianet died, what did you move onto?

That's when I got the Fender Rhodes, which I still use occasionally. At the time that was the standard setup: Hammond and Fender Rhodes. Of course, the Moog systems were starting to come out – you know, the wall-to-wall telephone exchange systems built into wardrobes, but you couldn't use them live unless you were Emmo (Keith Emerson).

What was your first synth?

Well, I waited 'til the Minimoog came out to get into that, simply because of the convenience. A lot of bands started using that – Yes, ELP, and so on – so I figured I'd better get one. I was at college at the time and we were "studying synthesizer" on that early EMS thing, the VCS3, which Floyd and Roxy were using but mainly in the studio. But no-one at the college knew how to get a decent sound out of it. We were always experimenting with it, waggling the joystick, but all it made was horrible noises. It's difficult to believe that was only 12 years ago, but that was the leading edge of synthesizer technology.

You've now got two Minimoogs, plus numerous other Moogs. I notice that with them and the numerous Prophets, you tend to favor the American sound.

Yes, I've always been an American (or at least an English-speaking) synthesizer man. It's the warmth of the oscillators I like, which even today the Japanese can't seem to produce. The Prophets and Moogs, in particular, excel at this. Take the Taurus pedals, for example. You still can't get that huge sound out of anything else.

The next thing I bought was a Polymoog, which, although pretty terrible by *today's* standards, was the first polyphonic machine. That really did open up a whole new bunch of possibilities, because then you had instant string sounds and all those other pad sounds. And although the Prophet was probably far closer to a polyphonic Minimoog, the polyphony was still a major thing.

"I've always been an American synthesizer man. It's the warmth of the oscillators I like, which even today the Japanese can't seem to produce."

Of course, when the Prophet 5 came out that was a whole new thing. The sort of changes you could make to a Minimoog patch you could now do polyphonically. The Polymoog was really quite rigid in its tonal alterations, but the 5 was a dream machine in terms of flexibility. It was like five Minimoogs plus a programmable memory, which was really revolutionary.

Was there anything between the Polymoog and the Prophet?

Yes, the Yamaha CS80. That was a glorification of the Polymoog approach. It did have four memories you could set up (if you include the two under that lid with the fiddly little sliders), but basically it was still mainly presets.

The CS80 has lasted a lot longer than the Polymoog in terms of its sound.

It's still being used, despite its limitations . . .

Yeah. I bought mine around 1980, to help out with the Yes thing, but I was never that into its sound. I suppose it's part of my prejudice against Japanese synths. They just don't have quite the right sound for me. I've still got it though, and I'll be using it somewhere on the new album I'm sure. I'm going to get everything on there somewhere.

So what was the next major purchase in the Geoff Downes arsenal?

The next thing must have been the Solina. I got that around 1978. That was the string machine to beat all string machines. At that time, on sessions, I was using that, the Clavinet D6 (bear in mind that when I got that in the late '70s, funky disco music was in demand), the Hammond, the Rhodes and the Minimoog. That was like the state-of-the-art setup at the time. Theoretically, you could get any sound you would be asked for out of that lot.

It was when we were doing the Buggles stuff that I got the Polymoog, the CS80 and then the Prophet 5. Then, when I went out on the road with Yes, which was the first really major live stuff I'd done, I added a Novatron and a set of Taurus pedals.

It was just before the Yes tour that I bought the Fairlight. In fact, I used it on the album on a track called 'Man in a White Car.' I'd literally just gotten it and I was still experimenting with it. I'd never spent anything like 15 grand on a synth before.

There had been the Yamaha GX1 before that, which Emmo and Stevie Wonder had got, which was supposed to be worth over twice that, but I don't know if they ever paid for them or whether they got them on some sort of endorsement deal. It was funny, when we were in Japan a few years ago – we saw quite a few of them stuck in resorts and hotels and places like that.

But a lot of people couldn't understand it when I spent that much on a "synth." The thing was, the Fairlight was so revolutionary that nobody could understand what it was about.

I had one of the first Fairlights in the country, certainly the first that went out on sessions. When word got around that there was an instrument which could sound like timps one minute and strings the next, then people wanted to use it on records. They'd been dreaming of it for years. So my name came up on record sleeves in connection with the Fairlight.

What was next after the Series II?

Well, the Prophet 10 came along shortly before we formed Asia. I bought it in 1981. I'd always loved the 5, and when I heard the 10 it just seemed a devastating sound. You'd just put it on the track and instantly the track sounded big – you know, that sort of epic hugeness . . . Because you had that instant stereo output, it made a great tonal enhancement of whatever else was playing with it. If you had the whole rhythm section going and you had a statement of the harmonic structure from the 10, then it was a great base to build on. People often referred to my sound in Asia as being a wall of keyboards, and the 10 was really the foundation that was built on, the backdrop against which everything went.

"Even today, there's nothing that sounds quite like the PPG. You can hear its individual quality no matter how much other stuff you MIDI in with it."

Was the Memorymoog another instrument that came into that "big analog sound" category?

To a certain extent, but the 10 was more fundamental. I used the Memorymoog a lot monophonically, so it didn't fit into the "instant pad thing" the way the 10 did. But it still had the warmth of those Moog oscillators, so it was great for bass sounds and so on. It's one of those synths that'll ▶

▶ cut through anything. The problem was that to start with, it kept breaking down. Fortunately, it's reasonably reliable now because I had the 'Plus' update – that gave it the sequencer (which I never use) and the MIDI (which I use all the time). It also seems to have helped the tuning, which was all over the place originally.

When did the PPG Wave 2.2 arrive?

I think that came along about '82. That was another one I got because it had a sound all of its own. It was away from the normal analog synths, yet away from the samplers as well. It was such a good idea to have a machine with the variability and sharpness of digital sound, but still with the warmth of filters and envelopes. There wasn't a need to learn a complete new way of working, but you got a completely new sound. Even today, there's nothing that sounds quite like the PPG. You can hear its individual quality no matter how much other stuff you MIDI in with it.

Moving back to Sequential machines, you're still using the SixTrak, which a lot of people reckon was not one of the company's better ideas. What is it about it that you like?

It's great for basslines, using the Stack mode to get lots of different elements and then combining them to make one big sound. I know on its own each voice sounds a bit thin, but you can't argue with six oscillators all going at once. It's actually better for bass sounds than the Prophet 5, because Stack mode gives you different sounds combined together, whereas Unison mode – which is what the Prophet had – only gives you five identical voices on top of each other.

But basically I'm an American synthesizer freak. I'll buy anything that comes out of the States.

Yet at about the same time you got the Korg Poly 800. That seems a bit out of place among predominantly American gear . . .

Well, I've always quite liked Korg stuff. I've got a Polysix and a Mono/Poly which I use a bit, and then most recently a DW8000. They're the only ones I find suitably different-sounding from anything else. And that's the reason I buy machines, after all – because they don't sound like anything else.

The Polysix and the Mono/Poly in particular have a warmth you don't find in most Japanese stuff, although with the Poly 61 it was back to the cold Japanese digital sound. The DW8000 and the DVP1 are getting back onto the right track, and the DW does have some very nice things on it, like the aftertouch and so on. It has a nice, wide encompassing sound.

The only thing I find lacking in Korg stuff is bite. As soon as you put something next to them, it tends to impoverish their sound a bit. But they are definitely my favorite among the Japanese makes. I don't own a DX7 and wouldn't want to.

The DX must have something going for it, though. It has sold more than any synthesizer in history, after all.

Well, I think it's really a case of "more is less" as opposed to "more is more." If you stack a bunch of Japanese stuff together, you end up sounding like Howard Jones. I don't mean that in a derogatory sense, but you can't actually differentiate between the different synths he's using. I remember hearing Dave Bristow doing a great demo with the TX816s controlled from the QX1, but the problem was it all started to sound the same. The horns started to sound like the cellos which started to sound like the saxes, simply because everything was being done by DX7 voicing.

I don't think you can get a wide range of sounds without a wide range of synths. Add eight DXs together and it doesn't sound like very much, which is what I meant when I said more is less. Add together eight different American and European synths, and suddenly you have more than the sum of the individual machines – and that's what I mean when I say more is more.

"I don't think the DX7 has a very interesting sound. Sure, it does good superficial impersonations of lots of different sounds, but they all have the same quality to them."

"People referred to my sound in Asia as being a wall of keyboards, and the Prophet 10 was really the foundation that was built on, the backdrop against which everything went."

Would you say that's because each machine is creating the frequencies differently, so they complement each other instead of cancelling each other out, as they would if they were all using one type of synthesis?

I think it's more than that. I don't think the DX7 in itself has a very interesting sound. Sure, it does good superficial impersonations of lots of different sounds, but they all have exactly the same quality to them . . .

After the Poly 800 came a couple of less well-known machines - at least in world terms - the Elka Synthex and the OSCar. What attracted you to them?

Well, when I said I was into American synths, I meant as opposed to Japanese. I'm really into European stuff as well. That dates back to the Solina string machine, I suppose: I've still got that and use it occasionally.

The Synthex was just a great-sounding machine. Before I got the Solina I used to use an Elka Rhapsody on sessions to do the same job, so when the Synthex came along I didn't have a big prejudice against Elka because of the home organ thing. It's actually a very, very good synth. I think of it like a programmable Solina. It does those warm strings sounds that only the preset string synths used to do, but it has loads of other good sounds as well, all with the same warmth.

I've always thought of the Synthex as a sort of cross between the Prophet and the Solina. Something with the programmability of the Prophet but with an extra warmth and size that you only got with chorusing and a stereo output. Sequential got that by doubling two 5s to make the 10, but it was an expensive way to get the same effect. The Elka had some other great things too, like digital ring mod and cross pulse-width mod, which could give FM-type sounds but with warmth. You didn't have to go through a lot of complicated procedures to get a good sound.

The Elka was worth its weight in gold, partly because I got it just as the Emulator II was coming out. I got the EII a day or two before we started work on the third Asia album and it didn't have a sequencer on it at the time, and the MIDI hadn't come through on either the Fairlight or the Synclavier (which I was on the point of buying), so the Synthex's MIDI sequencer was really useful for sequencing the EII.

Then there was the OSCar, a great monophonic synthesizer which was programmable. It came after the majority of the monosynths, and it was the only other monosynth that came anywhere near the original Minimoog in terms of sound. And the great thing was, you didn't have to have all the patch charts that I still have to use with my two Minimoogs. The OSCar has all the features of a classic monosynth, but because it's more recent it's got a lot more, like MIDI, a pretty nifty sequencer with program changes and so on. Then there are nice little refinements, like delayed vibrato and a single-triggering portamento which lets you keep one hand free for a polyphonic accompaniment on something else at the same time.

And on top of everything, its got such a ballsy sound. On lead stuff it'll cut through anything. On several occasions I've got everything but the kitchen sink MIDI'd up for a huge backdrop, and there's the OSCar soaring away on top, cutting through it all.

The Emulator II, which you mentioned earlier, has been a very important instrument for you, hasn't it?

Yes. The EII came out at just the right time, bridging a gap that existed between the low fidelity and short sampling time – but polyphony – of the Fairlight, and the high fidelity and long sampling times – but monophonic playback – of the Synclavier as it was then.

The original Emulator was little more than a poor man's Fairlight with short sample time, and I didn't think much of it. But when the EII came out they had made it a real instrument. And like the PPG, it had standard analog control for altering the sound, so it wasn't another whole new thing to learn. I know that now its quality has been surpassed by much cheaper machines, but in those two years I've built up such a library for it that it would take forever to transfer it to, say, a Prophet 2000.

I'm perfectly happy with its quality. On the sounds (orchestral and tuned percussion, mainly) that I use it for, it has a lovely warm quality, which probably comes from the fact that it's not one hundred per cent faithful.

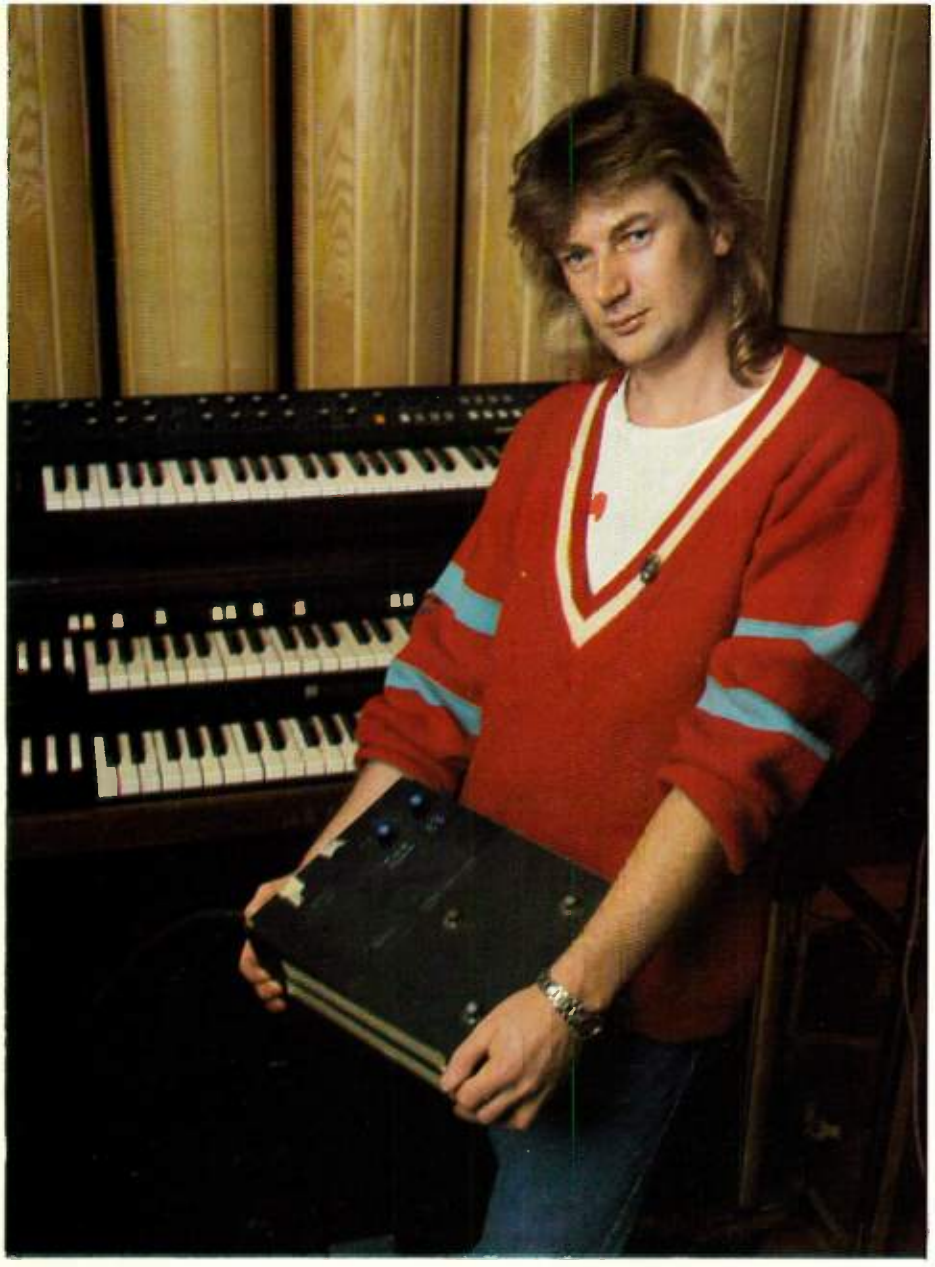
Over the years I've built up enormous libraries of sounds for the Fairlight, the EII and most recently the Synclavier, and I'm not about to throw all that away just because something with higher recording quality comes along. That's another reason why my setup becomes bigger. I still need access to all those sounds. I know what they all are in my head, so if I remember something that works in a particular context, I can call it up, or get my roadie to dig it out of the archives.

"In my setup I have practically unlimited polyphony; it's great for a keyboard player to just be able to go ahead, play the part, and have the machine take care of voice assignment."

You're now in possession of both a Fairlight and a Synclavier, which some people would consider as being over the top. Having been such a champion of the Fairlight almost since day one, how come you're not updating to the Series III?

Well, when the Synclavier came out originally it wasn't that interesting, to me at least. It was pretty expensive just as a synth and then, when it was updated to sample, it was just a straightforward recording-type implementation. The great thing with the Fairlight back then was that it was a complete system. You could play samples polyphonically and mess about with them, get different samples at the same time, and sequence them, first with the real-time recorder and then with Page R.

It was only when the Synclavier's sampling became polyphonic, which was less than two years ago, that it became more appropriate for me. It was very frustrating, being able to play synthesizer sounds polyphonically, but not the samples. And although the quality and bandwidth were a whole lot better, it still wasn't of that much use. You couldn't walk up to the machine and say "right, here's a piano or strings ▶



sample, let's record this piano or string part." We used it on the last Asia album for drum samples but little else.

The great thing on the Fairlight was that you could do all sorts of things with samples: merge, overdub, sequence and so on. But since then the Synclavier has come on in leaps and bounds, to the point where it's a complete recording environment with high-fidelity polyphonic sampling, multitrack recording, and MIDI sequencing. Plus, most of this was there a good year before there was anything but rumors of the Series III. So I got the full Synclavier package, because it was available first.

Now the Synclavier is so far down the road, I don't foresee the Series III ever catching up. In fact, the system as NED foresees it is going to keep expanding into areas that Fairlight hasn't even looked at, tapeless multitrack recording and so on.

And on the Fairlight, you can still only record monophonically on each track like Page R has always been, whereas the Synclavier records whatever you play and assigns voices accordingly. This is particularly useful when you're controlling external machines via MIDI, as in my setup I have practically unlimited polyphony. It's great for a keyboard player to just be able to go ahead and play the part and have the machine take care of voice assignment. There's none of this "I played five notes at one point so I've got to have five tracks of my 16-track sequencer assigned to that part."

Then there's the facility to drop in and record exactly as you would on a multitrack tape machine with rewind and fast forward, which makes the whole system so immediate. It makes life so much easier.

When you first got the Synclavier it was still at the monophonic stage, and you were using it primarily as a super high-fidelity drum machine. Yet presumably you now see the recording side of it as the main thing now, as a central MIDI workstation?

Yes, it's my main compositional tool. I play all my other MIDI keyboards from it and sequence them with it as the master unit all the time. It's great for arranging as well as for writing, because I can change the sounds around. I've actually got the best of both worlds, because I can get all my old sounds from the Prophets (which are all MIDI'd) onwards, but also have state-of-the-art sampling, recording and sequencing in one integrated system.

Do you use the synthesizer side of the Synclavier much?

I do have a few synth sounds on it, but I don't really regard that aspect as terribly important, as most of the sounds I want are easier to get from an analog synthesizer. Making up sounds from scratch on it can really be quite time-consuming, which is a bit of a contrast to how quick it is for sampling and sequencing/recording. I still prefer ADSRs and filters to partials, FM amounts and harmonic addition. The synthesizer side of it is fairly old now, and I find it quite limited, really. I don't think it gives you the weight of sound you can get from, say, the Prophet 5 or a Memorymoog. And you can't tweak cutoffs or jump oscillator octaves as you go along.

It's the newer side of the machine which really interests me: using the sequencer to record parts played using internal samples (drum sounds and so on) and external synths for the most part. It's like a major compositional tool and orchestrator. I record the performances into the Synclavier sequencer with either justified or unjustified recording (more often referred to as auto-correct or quantization on other machines) as seems most appropriate to the part, and then review the sounds to be used in context in the recording studio, so that time is not wasted there getting the part right. I tend to use justification on the drum parts and other rhythmic stuff, and then do the solo parts freehand.

The other thing about the Synclavier sequencer is that you can slide tracks against each other.

Yes, that's great for compensating for the different MIDI delays in the various synths I'm triggering, or the fact that a slow attack synth patch or sample is speaking late because of its rise time . . . If a part is dragging against the rest of the track, then I just advance it until it's in the right place, until it feels right. Then I store that as part of the sequence and it's always right on.

If I decide to change the sound or even the synth(s) playing that part, then I'll slide the track again 'til it feels right. For example, if I record a part with a guitar sound, but then decide I prefer the part on a horn sound, I might advance the track 30 milliseconds to make up for the fact that the horn hasn't quite the attack of the guitar. It's much better than cutting off the front of the horn sample to make it speak faster, which just makes it sound less like a horn. It's the ultimate control, because you can fiddle until it feels right – and that's the sort of control I've always wanted from all this technology . . . ■

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Photography Tim Goodyer

Korg DS8

Digital FM Synthesizer

At last, the world has access to a machine that produces sound via FM synthesis, yet incorporates a familiar, analog-style programming system. Does the hybrid really work? *Review by Simon Trask.*

EVER SINCE THEY acquired the rights to FM synthesis technology, Yamaha has maintained a monopoly over its usage. And as they've chosen to stick with essentially the same programming system on all their DX products, FM has become inextricably linked in musicians' minds with one way of working.

Korg's latest synth, the eight-voice DS8, uses FM technology and clearly sounds like FM, yet presents the system in a different and altogether more accessible format which ensures that any would-be programmer should make friends with FM very quickly. (In case you're wondering how Korg got its hands on FM, the answer lies in the fact that Yamaha now has a majority share in the company.)

Structure

GONE IS THE old FM terminology of algorithms, carriers and modulators. No doubt they're lurking somewhere under the DS8's calm exterior, but what *you* get to work with instead are two oscillators, two timbre and two amplitude envelopes (both four-stage), one pitch envelope (also four-stage) and one LFO (offering a choice of triangle, sawtooth, square and sample-and-hold waveforms) for each of the DS8's voices.

Oscillator 1 offers a choice of sawtooth, square, bright sawtooth and bright square waveforms, while oscillator 2 restricts the choice to sawtooth and square waves but additionally allows you to select cross-modulation (in which case no sound is output from oscillator 1, since it's used to modulate oscillator 2).

Remembering that we are dealing with FM synthesis, the timbre section for each oscillator isn't about filtering in the usual sense. Rather, this is where you alter the relationship between carrier and modulator – as a fixed value, or dynamically with a four-stage envelope. Setting the overall "timbre" value (which can, of course, be varied in time by its associated timbre envelope) to its minimum produces a sine wave, ie. an unmodulated carrier. Further parameters for each oscillator are "spectrum" (on a scale of 1-8), which actually allows you to set the pitch of the timbre "operators" (producing an effect vaguely akin to resonance), and ring modulation (on a scale of 0-3). The latter feature is another example of how Korg has taken the effort out of FM programming, ie. they create ring modulation effects in FM for you, using whatever sound you have programmed. It's worth noting that the effect(s) that these two parameters introduce are ultimately controlled by the timbre level of the relevant "operator." If they have little or no effect it's because the timbre (modulation) level is set low. If you understand FM in Yamaha's terms you'll have no trouble understanding that, which raises an interesting point: although Korg has chosen a familiar and simple programming system, the results you obtain from manipulating sound with it aren't always readily understandable unless you understand the real structure which underlies it. On the other hand, there's nothing wrong with an element of unpredictability.

Now, I'll try not to labor the DX comparison (Korg doesn't even mention FM in their manual, let alone Yamaha's jargon), but the DS8 seems to employ a four-operator FM architecture. In fact, experimentation using a DX7 side-by-side for comparison suggests that the DS8 employs two four-operator FM algorithms (numbers four and five on the DX9, but seemingly minus the feedback loop on five; the second algorithm is selected by choosing cross-modulation for oscillator two). This conforms well with the general observation around these offices that the DS8 sounds more like a DX9 than a DX7. Those of you who are familiar with Yamaha's terminology may find that you can't shake it off when you approach the DS8 – and that, ironically, the organization of Korg's synth works against you until you find a happy working medium. It is possible to understand the DS8 in Yamaha's terms if you really want to, though there are one or two lingering questions. Those of you who have steered clear of FM because of the way it has been presented up 'til now will ▶

Structure *"The DS8 uses FM and sounds like FM, yet presents the system in a more accessible format which ensures the would-be programmer will make friends with FM quickly."*

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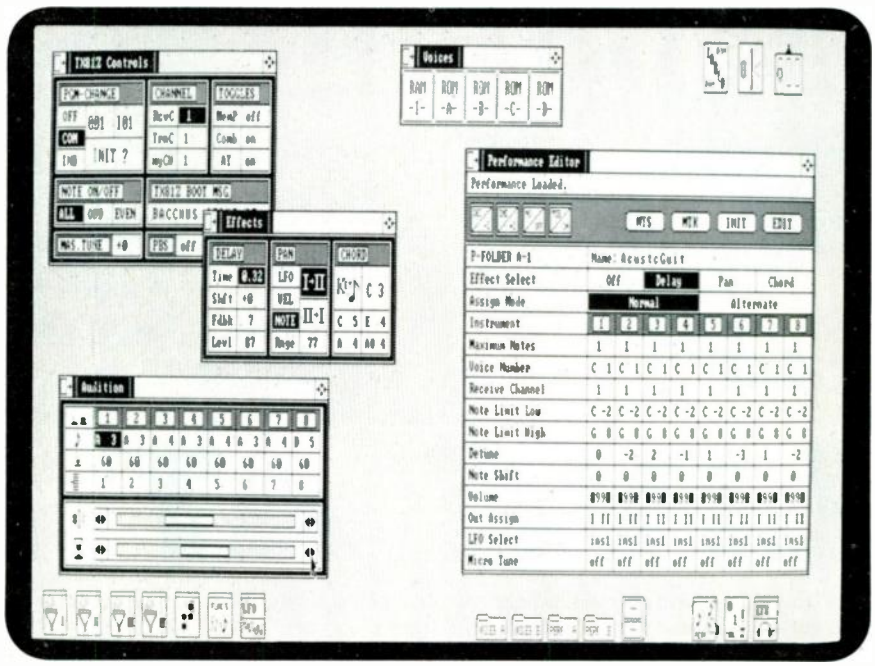

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Control

KORG HAS ALSO set about giving FM a new degree of performance flexibility by providing front-panel timbre and envelope sliders which allow you to alter "brightness" and the overall shape of the timbre/amplitude envelopes in real time. In fact, these sliders have no effect in edit mode, and when you enter this mode the associated values revert to those programmed – which seems like a sensible idea. The timbre slider functions along FM lines, and appears to work with reference to the currently-set timbre level(s) – but its effect isn't exactly analogous to altering timbre level.

The DS8 provides four keyboard modes: single, layer, double and multi. Layer is what we would normally call dual mode (and includes the ability to heavily detune the two sounds and to set an interval of up to an octave in semitone steps), while double is split mode with the ability to overlap the split-points. Double mode additionally allows you to specify the number of voices on each side of the split (with the total not exceeding the DS8's eight-voice limit). Multi mode allows the DS8 to flower into multi-timbral glory: eight patches (known as groups) can be selected, each one assigned its own MIDI receive channel (1-16) and number of voices (again, within the eight-voice limit). You can also play any one of these patches from the DS8's keyboard (within the assigned polyphony) by setting the synth's MIDI transmit channel to the receive channel of the relevant group. Thus, for instance, you could play (and perhaps record) a monophonic lead synth sound on the DS8's keyboard while an external sequencer played back a monophonic bass line and a four-voice chordal string part – and you'd still have two voices free for another part or parts. A further useful feature of double, layer and multi modes is the ability to pan each patch to either left, right, or equally to left *and* right (center) audio outputs, enabling sounds to be processed individually.

Programming *"The ring mod feature is another example of how Korg has taken the effort out of FM programming . . . they create the effects for you, using whatever sound you have programmed."*

Now, these modes wouldn't be of too much use if you couldn't store relevant data which could be readily called up. This is where Korg's "combi-programs" come in – performance memories under another name. Ten of these can be stored onboard the DS8, embracing the keyboard mode and related features like voice allocation in double and multi modes.

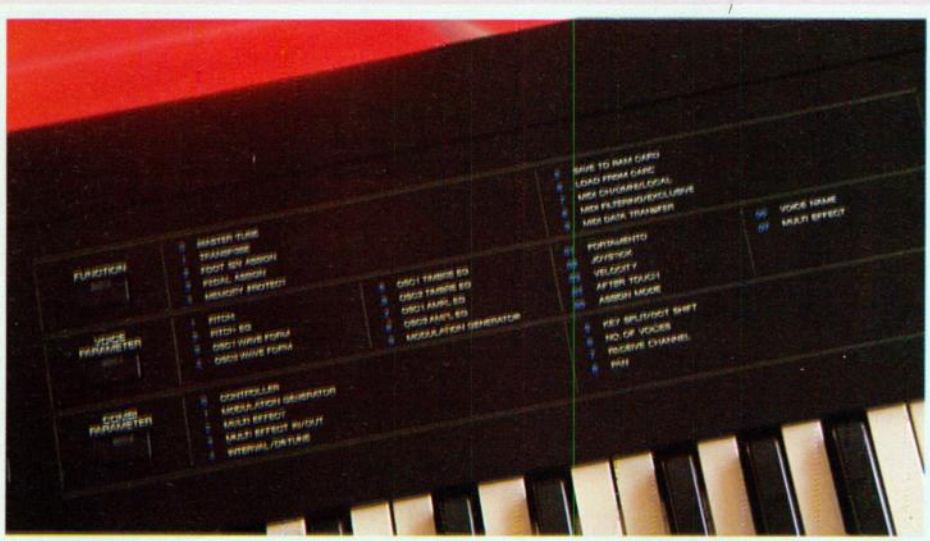
The DS8 has 100 patches onboard, while external storage is taken care of by a choice of three credit-card RAMs (offering 100 patches + 10 combinations, 200 patches + 20 combinations and 400 patches + 40 combinations), and of course data transfer via MIDI. A quick spot of arithmetic will tell you that you can have a maximum of 500 patches and 50 combi-programs available with a single cartridge slotted into the DS8, which can't be bad. One point about combi-programs: you can't combine internal and external patches. But it's easy enough to transfer individual patches between internal memory and cartridge, as well as transferring complete banks of 100 patches and 10 combi-programs.

The factory sounds offer much that is familiar in FM-land, and that familiar crispness of tone is much in evidence. Clanging pianos (there's even a patch called 'Clang Keys'), tinkling electric pianos, biting clavs, well-detailed organs and pipe organs, percussive vibes and marimbas, clangorous bells, weedy strings (sorry about that), warm horns, funky slapped bass . . . and the usual complement of wind, rain, birds and choppers. Now, a number of these are just fine, but more often they fall short of the instrument's full potential, which is strange, given the expertise that exists in FM programming these days. And one undesirable feature of FM synths up 'til the DX7II is, unfortunately, also evident on the DS8: digital noise on bassy sounds.

The DS8 offers a useful range of parameters for taking advantage of the keyboard's sensitivity (including attack velocity and channel aftertouch). You can define velocity response for each of the timbre and amplitude envelopes (though it's not possible to invert the response) and aftertouch response for pitch modulation generation, timbre level (for both oscillators) and amplitude level (for each oscillator).

Korg has also provided a choice of polyphonic and unison keyboard performance modes. Unison mode stacks up all eight voices to play a single patch, with the ability to choose single- or multi-triggering and to detune the voices – a good way of creating a powerful sound. Korg hasn't forgotten that old faithful portamento, either, which is programmable for each patch and can be switched in/out from a dedicated front-panel button.

The keyboard itself is a particularly effective ▶



compromise between synth-style and piano-style: plastic keys and medium travel coupled with a modestly weighted action which allows you to "dig in" more than on some synth keyboards.

Programming

EDITING SOUNDS ON the DS8 (in fact, doing *anything* on the DS8) is greatly aided by the 2×40-character backlit LCD, which allows you to see several related parameters at the same time. Left/right cursor switches allow you to step through the parameters, while the familiar increment/decrement buttons and data entry slider take care of parameter value alterations. Thankfully, these sort of large and informative window displays are becoming a common feature on synths and samplers nowadays.

The DS8's parameters are listed on the instrument's front panel and are grouped under three headings: function, voice parameter and combi parameter, each of which has its own selector button. In conjunction with the numeric "keypad" under the LCD window, this listing makes it very easy for you to find your way around the instrument and to familiarize yourself with the features it has to offer.

Onboard effects look set to be flavor of the year among Japanese manufacturers. Korg was one of the first manufacturers to put patch-programmable effects on a synth; you'll no doubt remember the DDL on their DW8000 and Poly 800 MKII synths. Now Roland has come up with patch-programmable reverb, and not to be outdone, Korg has included a section called "multi-effects" on the DS8. This allows you to program any one of the delay (manual, long or short), doubling, flanger and chorus effects for each patch and combi-program. These appear to be digitally implemented, and can be switched in and out from the synth's front panel.

One irritation that I encountered on our review model is that when you call up a patch or combi-program which uses any one of these effects, there's a 1-2 second delay before the effect actually comes into operation. By any reckoning, that's one built-in effect too many.

Long delay offers a delay range of 105-729msecs plus a feedback level and effect level, while short delay offers a range of 20-88msecs and similar feedback and effect levels. Doubling, flanger and chorus are useful effects, but to these ears they don't possess the richness that you might expect from the same treatments produced by a dedicated delay line.

While each patch can have its own effect, the DS8 is only capable of generating a single effect at a time. This means that when you select double, layer or multi modes you have to make a couple of choices: which patch's programmed effect you want to use and which patch(es) you want the effect to operate on. What this means is that, for instance, in double (split) mode you couldn't have a lead synth sound with long delay and a string sound with chorus – but you could decide, say, to have the long delay on the string synth sound only.

A similar situation applies with multi mode, and it's worth noting that, although you have access to up to eight different sounds when using the DS8 in conjunction with a sequencer, these sounds aren't totally independent. Just as the DS8 can only generate a single effect at a time, only a single set of controller assignments and a single set of Modulation Generator (LFO) assignments can be implemented by the synth. You can choose which patch (group) will be the source, and turn their effect on/off for each patch individually.

MIDI

THE DS8 ALLOWS you to select the MIDI transmit/receive channel, omni on/off, local on/off, active sensing on/off, patch change on/off, controllers on/off and System Exclusive on/off. MIDI System Exclusive communication allows all 100 internal patches and combinations to be bulk transferred. Individual patch data can also be sent over MIDI by selecting the relevant patch (as on the DX7, you need to have System Exclusive enabled to do this). This should make life fairly simple for those of you who want to use an all-purpose MIDI data storage device or equivalent software program. At the same time, Korg has actually provided a fairly sophisticated System Exclusive implementation (with full details in the manual), which software developers will be able to make use of for the usual patch editor/librarian programs.

The DS8's rear panel offers MIDI In, Out and Thru, stereo and headphone outputs, sustain pedal input, dedicated program up input and assignable footpedal and footswitch inputs. The footswitch can be assigned to program down, oscillator select (1, 2 or 1+2), velocity on/off, multi-effect on/off, aftertouch on/off and portamento on/off. The footpedal can be assigned to control MIDI volume, timbre (equivalent in effect to the front-panel timbre slider), pitch modulation or timbre modulation.

Conclusions

WITH THE DS8, Korg has certainly succeeded in its aim of making FM technology accessible and more interactive in both editing and performance. However, it's a moot point whether they have actually improved the *quality* of FM (as Yamaha has done on the DX7II), and the question has to be asked whether or not the user-friendliness that the DS8 gives to FM synthesis is enough to win it a large following. Fortunately, there are enough contemporary features on the instrument to give it a unique identity, apart from the myriad of FM instruments which Yamaha already has on the market – particularly at a time when a four-operator FM synth for over a grand might otherwise seem a bit dated. The competition is stiff, though, with Roland's D50 just coming onto the market and Ensoniq's ESQ1 still a winner.

Performance *"The keyboard is a compromise between synth and piano styles: plastic keys and medium travel coupled with a weighted action which allows you to 'dig in' more than on some synth keyboards."*

Unfortunately, the DS8's factory sounds don't tell the full story of this instrument, and will do little to bowl over those of you with a jaded ear – but it would be a pity if you didn't give the instrument a chance. A bit of DIY programming will help you to uncover the DS8's capabilities better – *and* you'll discover the benefits of the machine's simple architecture all the sooner.

If you still have time for FM, then you may well have time for this flexible and undeniably well thought-out instrument. ■

PRICE $1295

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STUDIO

Synhance MTS1
MIDI Synchronization Unit

Photography Bob O'Donnell

This new synchronization box combines chase-lock capabilities with a number of MIDI functions, at a fraction of the cost of most SMPTE-based synchronizing systems. Is it the ultimate home studio accessory? *Review by Bob O'Donnell.*

IMAGINE THIS SCENARIO. You're syncing your MIDI sequencer to your four-track cassette recorder, and about three minutes into the song you're working on you want to do a guitar overdub on one of the other tracks. You need to hear the sequenced parts to get your cue so, unfortunately, you have to rewind the tape to the beginning so that the sequencer can lock up to the sync track you've recorded on tape. Then you have to wait for the recorder/sequencer combination to get to the point where you want to record, and pray that you get it right the first time. If not, the whole process has to be repeated over and over until you're happy with the results.

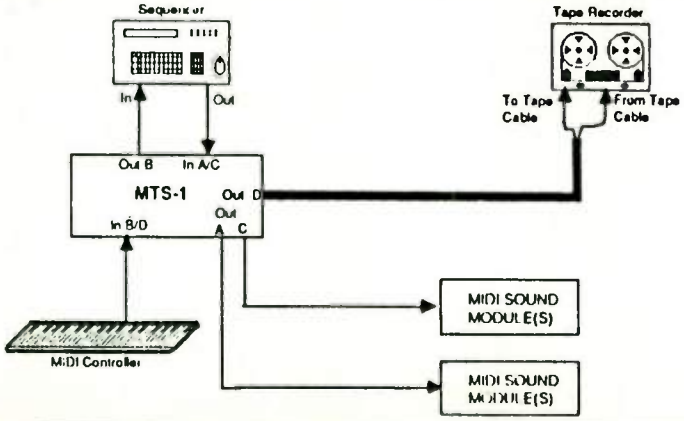
If you have an FSK or pulse-based sync-to-tape setup, this may well be a frustratingly familiar situation. It may have even led you to wonder whether it's worth using synchronization at all, particularly if you have ever endured the horror (and potential embarrassment) of tape dropout. (In case you haven't, what that basically means is that part of the sync track is lost forever and consequently, the sequencer will remain forever out of sync with the other recorded tracks. As you might well guess, it's more than a bit frustrating.)

"But wait," you say, "I know how I can avoid these problems: I'll use SMPTE." Sure enough, using a SMPTE reader/generator and a SMPTE-to-MIDI conversion box will give you an absolute timing reference and, if it's somewhat sophisticated, allow you to overcome tape dropout problems – but have you checked out the prices of some of those units? Granted, many new lower-cost items with SMPTE are becoming available, but if you're working in a home studio with a small budget, most of them are still out of reach.

So what's a poor boy (or girl) to do? Well, I recommend you take a good hard look at the latest offering from Harmony Systems, the Synhance MTSI. For about half the cost of even the most inexpensive SMPTE synchronizing systems, the MTSI offers chase locking of MIDI sequencers to tape, as well as MIDI merging functions. (In case you're a bit unsure, the phrase "chase locking" refers to the ability to start anywhere on the tape and have the sequencer "chase" up to that point and "lock" into the current position of the time code. So, referring to the above scenario, instead of having to rewind the tape to the beginning of your tracks every time you want to do an overdub, you could simply rewind to a point just before you want to punch in. Then all you have to do is hit the play button on the tape recorder, and the sequencer automatically moves to the point where you started the tape. Needless to say, this greatly improves the efficiency of your system.)

Up until now, the only way that you could get chase-locking capabilities incorporated into your system was with SMPTE, but what the clever folks from Harmony Systems have done is taken advantage of the oft-overlooked MIDI Song Position Pointer data that's produced by most of the newer hardware and software sequencers and drum machines, and built a synchronization system around it. (Song Position Pointer, or SPP, determines the location within a piece of music by counting and keeping track of the number of 16th-notes there are in it.) The MTSI converts SPP data into a proprietary timecode which contains continuous information about the tempo and the position of the sequence and sends it to the tape recorder. When the tape is played back, the MTSI reconverts the audio signal back into SPP data and sends it to the connected sequencer via MIDI and thereby controls its location. (For all of you technical types, the MTSI puts out a 2400 bit/sec phase-encoded signal, which is the same format that SMPTE generators use, but the MTSI timecode is *not* compatible with SMPTE.)

The real beauty of the Synhance synchronizer is that the half-rack size unit works invisibly – just hook it up and forget about it. In fact, there are absolutely no controls on it, so there's nothing you can do to alter its operation (except turn the keyboard echo feature off with an internal DIP switch, but I'll talk about that later). While this may sound like a potential pitfall, it really isn't because of the nature of what the MTSI does, and because it's very intelligently designed. The people at Harmony Systems figured out the kind of capabilities that are important for MIDI/tape synchronization and incorporated them into the MTSI. It's that simple.



Operation

SETTING UP THE MTSI is a relatively easy process. The well-written owner's manual explains what each of the two inputs and four outputs are to be connected to, and what kind of data flows through each of them. As you can see in the accompanying diagram, the unit basically fits in between the MIDI controller and the sequencer, with the tape recorder connected in a type of "side chain." The only thing that could be confusing (and one of the two minor complaints I have with the unit) is the labeling of inputs and outputs. When I first

looked at the back panel, I thought there must have been a reason (which I simply couldn't understand) why the outputs were labeled out of order and why the inputs had two letters each. I also wondered why the connection to the tape deck used a five-pin DIN plug, because I knew it could very easily be mistaken for another MIDI output (which it is not). A quick call to Harmony Systems led to the discovery that in an attempt to maintain the lowest costs that they possibly could, they decided to use existing back plates (which were originally used on one of their other products) on the MTSI. As a result, the labeling of the outputs is a bit compromised, but if you read the owner's manual, everything becomes perfectly understandable.

Once I'd properly connected the unit and was ready to record the sync track on tape, all I had to do was set the proper level on our Portastudio, begin recording, and then play my sequence through the MTSI. Setting the level was very easy because whenever the unit is at rest (or "idling" according to the owner's manual) it puts out a signal at the same level as the sync tone. My only other complaint with the unit, though, is that you can't turn off this signal, and if your mixer has poor crosstalk specs it may bleed onto other tracks. Of course, you can always disconnect the tape recorder input, but it would have been nice if they had incorporated an on/off button somewhere on the front panel.

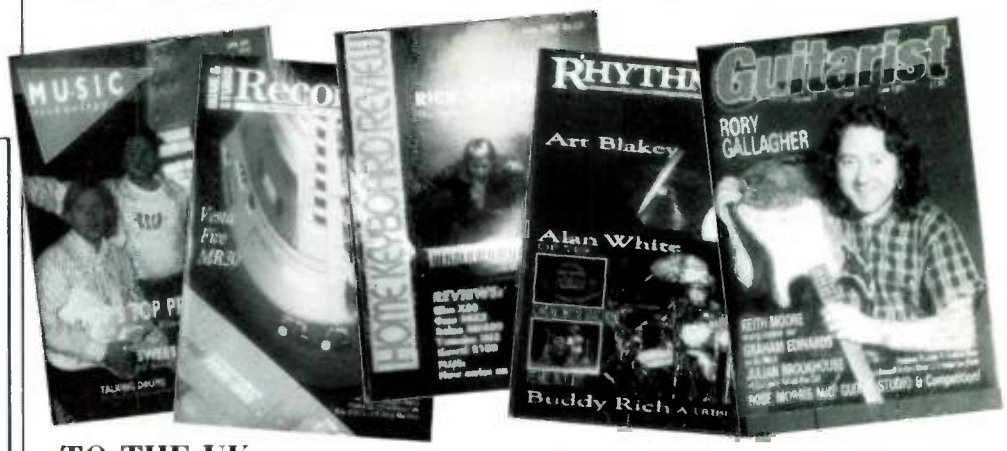
As the sequence plays, the MTSI creates its timecode and writes it to tape. Any tempo changes can be recorded and recreated, but once the sync track is recorded, the tempo changes and song length remain permanent. The MTSI does not give you the same kind of flexibility that SMPTE-based systems allow (they permit you to make tempo changes at any time in the recording process) because they provide an absolute timing reference as opposed to the relative timing reference that the MTSI creates, but again, you have to pay for that power.

If you have an older sequencer which doesn't put out Song Position Pointer data, the MTSI can still work (as long as the sequencer outputs MIDI clocks), but it won't give you the ability to chase lock. Thanks to a well-designed error correction system, however, it will give you the ability to overcome tape dropout problems. The way the system works is that once it redetermines the time that has passed during the dropout, it computes how many MIDI clock ticks were missed, and sends that many out instantaneously so that the connected sequencer can catch up. This all occurs very quickly, however, so errors generally go unnoticed.

OK. So how well does the chase locking and synchronization work? Very well, as a matter of fact. When I rewound the tape to the beginning and played it back, the sequencer kicked right in and the sole LED on the front panel flashed at the tempo which the sequence had been recorded at.

Even more importantly, when I stopped the tape, fast-forwarded to a point near the end of the sync track and started the tape again, the sequencer followed. The actual lock-up time was about two seconds, which admittedly is▶

COME ON OVER.



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▶ not the fastest, but it's a *huge* improvement over a straight FSK or pulse system. (The MTSI can actually determine the current tape position in about 100msecs, according to Harmony, but they included a two-second delay in the firmware because many existing sequencers incorporate a slightly different implementation of MIDI Song Position Pointer that requires this much time to properly lock up. The SPP portion of the spec was fleshed out at the January NAMM show, however, and this problem was addressed, so Harmony may take this self-imposed delay out of later versions of the unit.)

The MTSI is also very consistent; I went back and forth on the tape and it locked up the same way every time.

MIDI

IN ADDITION TO its synchronization capabilities, the MTSI includes two different MIDI merging functions. The first, which is commonly referred to as keyboard echo (and which can be turned off if you want to use the keyboard echo features of your sequencer), merges the output of the sequencer with the output of the MIDI controller so that they can both play the same sound module at the same time. If you connect two different controllers to the MTSI's inputs, you can also use it as a normal MIDI merger, which is an extremely nice "extra" feature for such an inexpensive unit. In general, however, these MIDI capabilities are not extras, but are important functions which allow the MTSI to work invisibly. (In fact, without them, the unit might actually create as many problems as it solves, because of the extra step in the MIDI chain that it adds.)

The second merging function combines timing information from the tape with the output of the MIDI controller, so that you can overdub onto your sequencer as you sync to tape. This is also a nice touch which works well, and is yet further enhanced by the fact that there are two MIDI Outs. They each carry identical information and in some small systems, may even eliminate the need for a MIDI Thru box.

Conclusions

IF YOU CURRENTLY have or have been considering buying an FSK-to-MIDI converter, or if you have an FSK or pulse-based sync-to-tape setup, you owe it to yourself to check out the Synhance MTSI. Even if you've been considering getting into SMPTE, it may be worth your time to see if the MTSI can do the type of things that you are looking for in a synchronizing system. Many people look to SMPTE solely because they want chase lock capability, and the MTSI can do that and more for a lot less than most SMPTE-based systems.

The unit is a well-designed, useful piece of equipment which can add a number of capabilities to your existing setup without you ever needing to worry about it at all. And for that alone, it deserves a big thumbs up. ■

PRICE $229

MORE FROM Harmony Systems, Inc., PO Box 2744, Norcross, GA 30091. ✆ (404) 662-8788

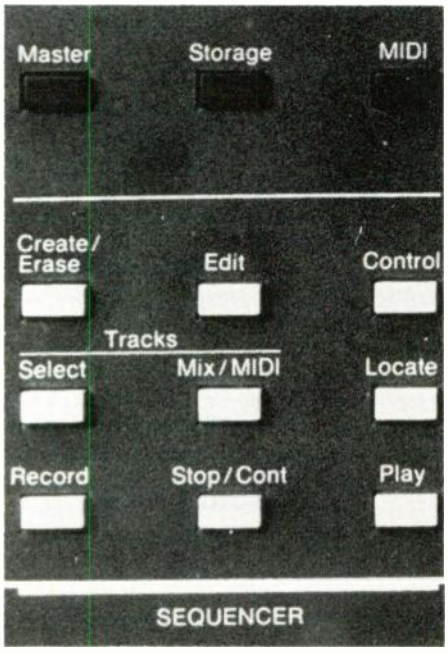
▶on MIDI channel 2 and a lead line on MIDI channel 3. Of course, you can repeat this overdubbing procedure until you start running out of voices on the ESQ1. (I have found that it is possible to have as many as five tracks going without too many problems, even if some of those tracks are polyphonic.)

Now, if at this point you decide that you want to replace the ESQ1's synth bass with a bass sample on your sampler, here's what to do. You could select "Seq" on the "Track Status" page for that particular track but I have found that it's equally easy just to turn a track to "Off" in the "Track Mix" page. Don't forget that you'll have to match your sampler's MIDI channel to that of the ESQ1's track and ensure that Omni mode is "Off" on your sampler. Unfortunately, the ESQ1 lacks MIDI Thru, so if you wish to drive an external device from your sequencer, you'll need some form of MIDI patchbay or thru box.

It could be that you want to double the string part on Track 5 with a DX string sound. Set your DX to MIDI channel 5 (assuming that Track 5 is set to MIDI channel 5 of course) and, if the "Track Status" is set to "Both," you'll get a nice rich string sound, especially if you detune the DX slightly against the ESQ1.

If you've got a keyboardless expander or sampler, you may wish to program parts on channels higher than 8 in order to run them. (I find it more convenient to program the external sequencer from within the Seq page instead of returning to normal mode.) So, select a track on the ESQ1 (let's use Track 6) and set it to MIDI channel 9. Next, program the part into your external sequencer. During the programming process you will hear all tracks on the ESQ1 except Track 6 (if it was being used before), but you'll hear your external synth/sampler on MIDI channel 9. Before you commence playback, you must restore Track 6 to MIDI channel 6 so that when you run your sequencer you'll hear all the parts.

Having said all that, however, it's unlikely that you'll fill up all eight tracks with ESQ1 voices, so you could probably use any spare tracks to run external synths or samplers without having to worry about changing their channels back and forth and turning them on and off. Remember, though, that if you're running a lot of external devices or another multitimbral unit (such as the FB01, the TX81Z, or even the new rack-mount ESQM), then you might run out of tracks (and MIDI channels).



Of course, there is no reason why you can't have two parts on the same MIDI channel. The way to achieve this is simply not to change tracks on the ESQ1 when you overdub your part into your external sequencer. You could also use this technique to overdub other things. Say, for example, you wanted to program a very intricate bass part. Lay down your basic bassline into the external sequencer and, once that is done, staying on the same track on the ESQ1, overdub the bass pull-offs. Repeat the process, overdubbing pitch-bends and vibrato. Now, on your external sequencer, merge all those bits onto one track to form a composite bass track.

Now, assuming that you've spent the last few days programming your new masterpiece into your sequencer or computer and you've gone in and edited the whole thing to perfection, you can now load the whole caboodle into the ESQ1's internal sequencer (assuming that the piece doesn't exceed the ESQ1's note capacity). True, you have to do the whole thing track by track but, other than that, it is very easy. All you have to do is sync your external sequencer to the ESQ1's MIDI clock, set up the ESQ1 to record in the usual way, and it will then think that it's being played from the keyboard. This could be handy if you use the ESQ1 for live accompaniment, since you wouldn't have to bring the sequencer along anymore. It would also be useful if you're going into the studio, since it would cut down on the amount of gear you have to take out and would save on space in a cramped control room.

In view of the fact that the ESQ1's sequencer is so good, you're probably wondering why we would want to use an external sequencer at all – especially as the ESQ1's sequencer is now expandable to 10,000 notes. But there are limitations to using the ESQ1's sequencer. First of all, it doesn't have a step-time input (although you can *edit* in step time) and there is a lot to be said in favor of step-time music entry, even for the best keyboard players. Also, whilst the editing facilities are very good on the ESQ1, the editing facilities on the new breed of sequencers such as the MC500 or QX5 are that much better again, as are the facilities offered by computer-based software packages. If you set the ESQ1 up as I have described, you'll be able to take advantage of its outstanding multi-timbral implementation *and* the exceptionally powerful music-making possibilities of today's sequencers.

If individual outputs ever become available for the ESQ1, the possibilities will be quite staggering. Even as it is, though, the combination of an ESQ1 and a present-day stand-alone sequencer is on par with some synthesizer systems costing far more. ■

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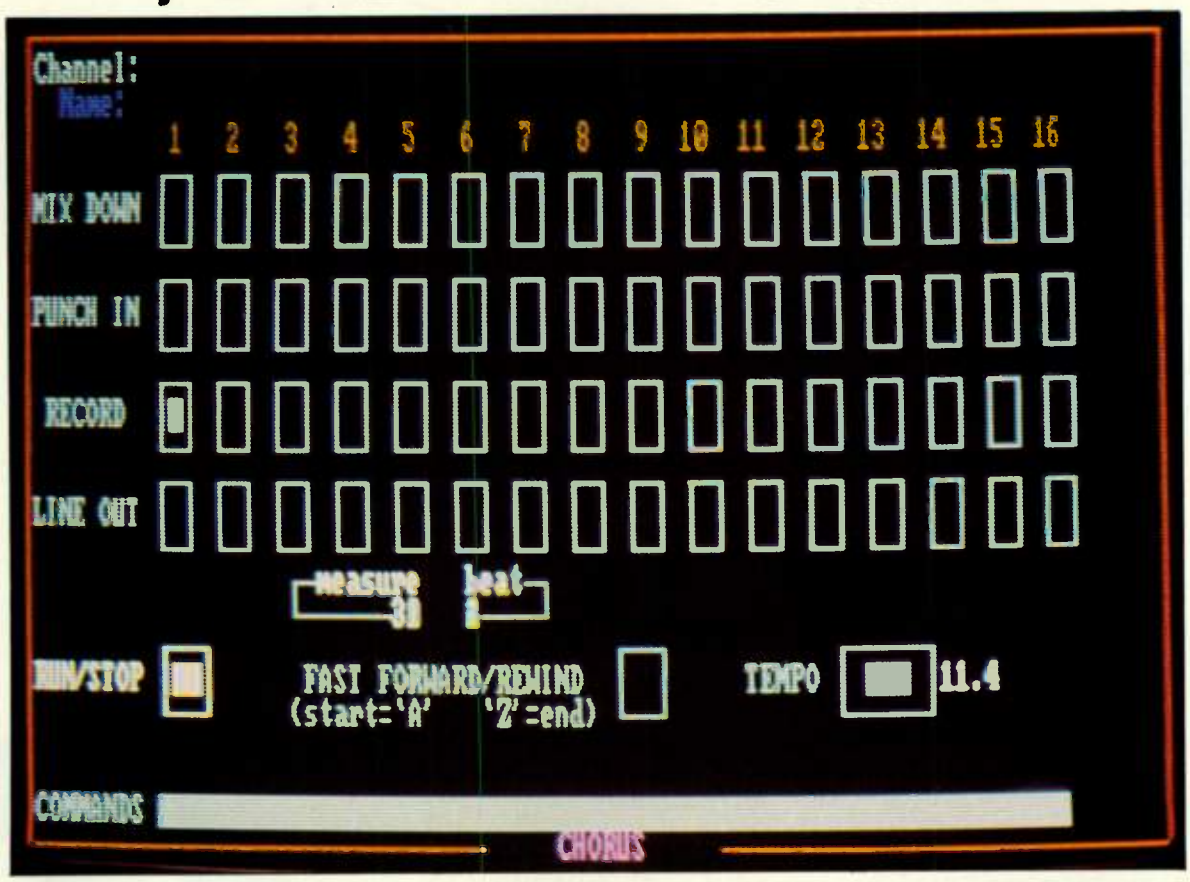


This IBM-based sequencer contains a number of fascinating new editing features and a host of tools for film and video scoring.

Review by Chris Many.

AS I MENTIONED last month, the IBM PC/compatible is coming into its own in the world of computer-based music systems, thanks to the increasing number of MIDI programs written for the machine. The program under review this month, the 48 Track PC II, is yet another fine entry into this field. Originally marketed by Syntech, the program is now being sold by its author, Robert Keller, who retained the rights to his work and has considerably updated the software.

48 Track runs on the IBM PC family, including the new 386 machines, and requires two drives and at least 256K of memory, although you'll need more if you're planning on doing any kind of serious work. It's understood you'll also need a MIDI interface (Roland's MPU401 or Voyetra's 4001). The disk itself is not copy protected, so you can make backups or run it from a hard disk (which speeds up operation of the program by decreasing disk access time – as you'll see later).

Overview

THE MAIN DISPLAY is easy to understand. You see four rows of 16 boxes, each row being labeled according to its function: Mixdown, Punch In, Record and Line Out. Below these rows are three tape transport buttons, Run/Stop, Rewind/Fast Forward and Tempo. Directly below this is an additional command line for entering a variety of editing and other commands. The layout is uncluttered, even if elements of it seem a little on the redundant side. For example, you can only record on one track at a time, so having a whole row of boxes to access the record function seems to be a bit of overkill, particularly as they only serve as a visual reference to the individual track you're recording on.

The boxes act as switches which are selected using the cursor keys, and are turned on and off with the space bar. Recording is a matter of selecting the track you wish to record on, and starting the sequencer with the Run button. You'll have a count-off (which you can alter to meet your needs) and then recording will start.

Rewind your track to the beginning (it's instantaneous, but rewind is used to maintain the multitrack analogy), press the Line Out box for the track you wish to hear, and there's the part you played. You can also program your controls to automatically rewind each time you finish recording something. To overdub parts, simply choose the next track you want to record on and repeat the procedure.

Erasing a bad take is actually required if you don't want it. Unlike some popular sequencing packages, 48TPC does *not* store your takes in a buffer. In other words, if you record something on a track, it is there on the track, and the only way to get rid of it is to erase it and to record again. 48TPC actually will merge a new performance with the old if you don't erase it first, making it a bit cumbersome, especially since erasing a take requires disk access, adding seconds to your waiting time.

You can then name each track or assign individual tracks to their own MIDI channel. (If you have recorded more than one MIDI channel on a track, you can either assign a new channel to all recorded notes, or only assign a new channel to notes recorded on any one channel.)

So at its most rudimentary level, 48 Track PC does a fairly good job of emulating a multitrack tape deck; it's easy to use without much hocum to work through. But in this day and age, most sequencers perform their fundamental duties well, so what else can 48TPC do?

Song Functions

NOT ONLY WILL the program function as a linear sequencer, but it also allows drum machine-style programming, à la section/song links. Let's say you've just recorded your basic rhythm tracks for the first 16 bars of your song using 24 tracks of the sequencer. Save this to disk as "versel." Go to the command line and

type in the name of the next section you wish to work on, let's say "chorus1," and record this next section of your song. The names and MIDI channel assignments don't change when you enter a new segment, so it makes it easy to put the same parts on the same track. When you're finished, save to disk, then add a bridge or any other parts or sections you'd like. If you remembered a line you wanted to add into the verse segment, just call that file backup and edit it until you're satisfied.

Next, go to the command line and link them all together. Commands are given in a form such as "verse1 + chorus1 + verse1 + bridge + chorus2 = song1." Doubling sections can be done by multiplying the number of times it loops around (verse1×3) and any combinations can be done in the same manner (chorus1×2 + verse1×2 + bridge = song2). A Comment window is available for making notes, such as a description of the current layout of your song (automatically updated as you enter it), and to do any other necessary administration that you need to.

All the sections are stored separately on disk, so if you had a nice little four-bar groove that you sequenced a few months ago that would be perfect for the end of the bridge, just insert it. I used the above song structure to illustrate the use of section/song features mainly because this kind of sequencing lends itself to that format, but you are in no way limited to that structure. Your sections can be any meter, tempo, length, and so on – it's just as easy to transcribe an orchestral score as it is to write a pop tune using this method.

For faster editing, you can work with 10 different sections in memory at one time, and flip back and forth between them with the press of a key. You can also chain together as many sections as you want. A word to the wise, though – if you are doing a long piece, make sure that your computer is configured with enough memory. When you combine these sections into a new file, the new file actually contains the full amount of data and becomes one long section. Unlike some other programs offering this kind of sequencing, this one leaves you with a linear translation of the commands you've given – so (verse1 + verse2 + chorus = song1) means that the file song1 is the sum of the other three files. So if verse1 adds up to 5K in memory, verse2 equals 12K and the chorus is 7K, then the file song1 will contain 24K. In other words, your sequences may very well fit within your computer's memory as you're working with them, but you can quickly run out of memory once you begin chaining them together – a frustrating experience, to say the least. On the other hand, once you're done, you do have a straight-ahead version of your song instead of a patchwork of loop commands that you need a map to navigate through.

It would have been nice, however, if there was an option for choosing between a complete file and one that just contains commands and doesn't eat up memory. But since there isn't, you'll need to keep track of the amount of memory you're using per section and have an idea of what your layout is going to be, so you don't have any surprises at the end. Of▶

MUSIC

READERS' TAPES

Reviewed by Yung Dragen.

THIS HAS BEEN the hardest column to write so far. No, not because of my foot – the cast's off and it's healing nicely, thank you. I'm tempted to blame my recent passion for rum and grapefruit juice, or even my renewed interest in 'old' analog synthesis (mid-'70s Carlos, Gleeson, Tangerine Dream et al). But the real reason is that I know I'm about to come face to face with just how jaded I've become.

Rock 'n' roll has been around since many of us have been born (at least for my generation). Those who would choose to record rock 'n' roll have quite a history to compete with – originality is highly prized and just as hard to come by. This is the month I try to clear out the backlog of 'normal', '70s-style rock tapes – and see if I can overcome my personal boredom with the style.

To start things off on a bad foot (if you'll pardon the pun), two of the three tapes came without any notes besides a contact address and song titles. For future reference, please include a little more documentation, such as equipment used, people involved, religious or citrus fruit preferences, and so on – I need an angle!

Busker, a duo comprised of Billy Shultz and Sharon Fletcher from Billings, Montana certainly have the most memorable haircuts of any tape I've reviewed so far – but more from a '50s/'60s sock hop angle than the multi-hued '90s look. The music is hard to describe in that it's a salad of many vegetables – two of the tracks ('Kept Alive' and 'Lonely Without Me') feature multitracked backing vocals, fast tempos, high energy, and Stax horn sections on synthesizers. I would have to call it a cross between *Something/Anything*-era Todd Rundgren, Motown, and (on 'If It Wasn't For You') 'Eleanor Rigby' Beatles of the same period. The lyrics are of the shallow love song variety. They included a lyric sheet, which is nice, but it also allowed me to verify such lines as "We were kept alive/Ooh, it was burning us/We were kept alive/With our love." They played all of the instruments themselves and recorded it more than competently on a Fostex B16 and TAC Scorpion board – as a matter of fact, both performance and recording quality were fabulous.

Suggestions, then? Erase some of the tracks; thin it out! To put it into zen terms, clear your mind of all distractions other than delivering the tune itself, and you will find your efforts shine with an inner peace and purity. There's so much going on vying for equal attention that I had great difficulty remembering anything specific about the three cuts.

Next up is a four-song demo by **While We Sleep**, which I assume is actually the solo act Todd Regn from Mt. Holly, New Jersey. Todd plays keyboards (including a Sequential Six-Trak – I recognized the factory organ patches), drum machine, guitar (acoustic and electric), bass, and sings. The styles presented are varied – semi-quirky dance music ('Unfold' and 'Don't Deny It'), hyped-up Cat Stevens ('The Test Before The Lesson'), and a nice synth-and-electric guitar rock instrumental ('Rhythmical Coition'). The guitar playing and overdubs are quite good, the vocals range from weak to adequate, and the keyboard work is, uh, a bit "loose." The synth programming is short on originality (except on 'Coition') – perhaps a limitation of the equipment. Engineering is clean but the sound is quite dry (Todd – one word – 'Microverb').

Unfortunately, the two dance pieces both use the same, busy, two-bar drum pattern (without break) throughout their entire lengths, and in general are weak where the other two are quite good. In short, this is the classic bedroom tape – a person with a limited budget who is quite good on a couple of instruments and at least adequate on all others, cranking out some tunes. In that regard, it is to be lauded in itself. If you like rocking acoustic guitar or guitar/synth instrumentals, check it out (and ignore the first and last tunes).

And then we move to **Ti Cubed**, featuring Timothy Keiningham from Owensboro, Kentucky. Tim has an interesting deep voice that grabs your attention and a very good sense of song structure – lots of breaks, and so on. Instrumentation on the tape is electric guitar (which is good), synths (solid bass work, fair to good rhythm/backing string synths, and a hair weak resonant comping and lead work), and energetic drumming – lots of rolls, breaks, and crash cymbals. This three-song demo ('Sailing on the Wind,' 'Crazy,' and 'Hear Me Calling') is over before you know it, being in the mid-to-late '70s style of medium/high energy rock with generous amounts of keyboards replacing the second guitarist. Engineering and production is again somewhere between rather and very good, needing only a touch more clarity to perfect.

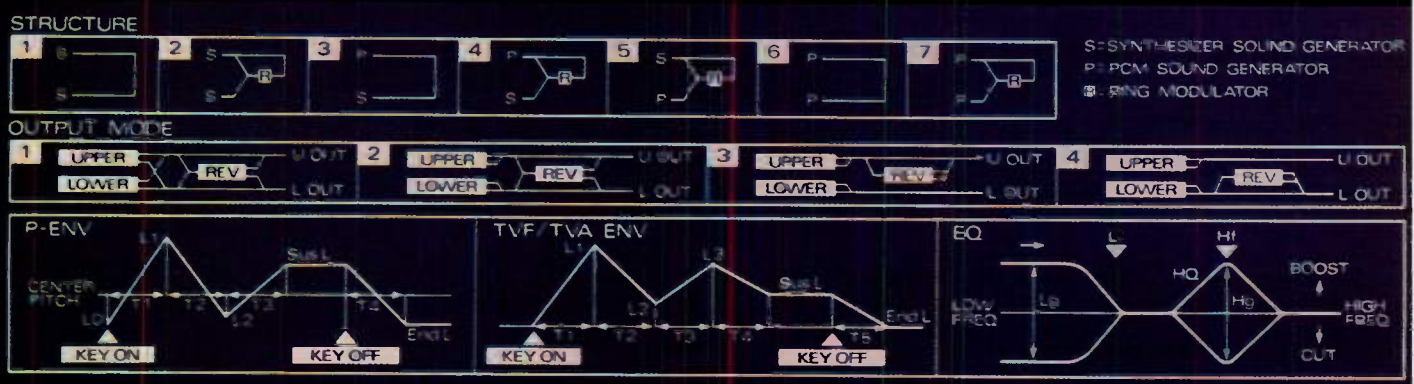
I can't help but think that I've heard a lot of this type of music before, but I can't put my finger on the exact references (a rockier Kansas is the closest I can come – I really don't own albums like this). It has been a decade, though, and nobody has been copying this style to death in the meantime. The end result is that the tape sounds kinda fresh while invoking a bit of nostalgia. On the several listenings I've given it, I've never *dis*liked it, and considering my most recent driving tape has been *The Best of Grand Funk*, I happened to find it rather nice on the most recent listening. And it's so nice to hear a live drummer again (particulary after Todd Regn, above, saw fit to use the same drum pattern twice without change).

This is a free country – people are still allowed to play rock – and as long as it's approached with honesty (such as Ti Cubed, or While We Sleep's 'The Test Before The Lesson' and 'Rhythmical Coition') as opposed to just cleverness or slickness (Busker, Sleep's 'Unfold' and 'Don't Deny It'). It's still valid and it still works. Viva la consistency in the universe. Until next month . . . ■

Contact addresses:
Busker c/o Billy Schultz/Sharon Fletcher, 4427 Palisades Park Dr., Billings, MT 59106.
While We Sleep c/o Todd Regn, 2 Beulah Ave., Mt. Holly, NJ 08060.
Ti Cubed c/o Timothy Keiningham, 2520 Middleground Drive, Owensboro, KY 42301.

Send your demo-tape, along with some biography/equipment details and a recent photo if you have one, to: Reader Tapes, Music Technology, 7361 Topanga Canyon Blvd., Canoga Park, CA 91303.

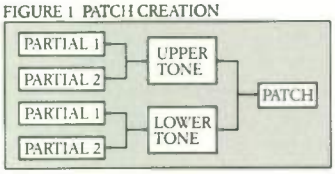
A NEW TECHNOLOGY IS CREATING A POWERFUL STORM IN THE WORLD OF SOUND SYNTHESIS



INTRODUCING THE D-50
THE BOLD NEW FORCE IN DIGITAL

To the Player It's a Dream, To the Programmer It's a Miracle/Imagine a new technology that is so sophisticated that it offers totally new and unparalleled sound creation possibilities, combined with a programming method so logical that it actually builds upon the knowledge you currently have of sound synthesis. That is the essence of the D-50 Linear Synthesizer, a completely new, fully-digital synthesizer realized by Roland's Proprietary LA Synthesis Technology. The sounds created by the D-50 are simply breathtaking, resonating with character, depth and complexity, but with a warmth and completeness digital synthesis has never had before. The reason is that no sound has ever before been created in a manner so complex and rich with possibilities, and yet ultimately so very logical. Linear Arithmetic (LA) is normally used for computing complex mathematical problems in the field of science. In the area of sound synthesis it is an ideal creative method, offering superb predictions, analysis and control capabilities. Roland engineers have spent years developing a new highly sophisticated LSI chip, code-named the "LA Chip," that utilizes a linear arithmetic technique to digitally synthesize sounds. The "LA Chip" is the heart of the D-50.

FIGURE 1 PATCH CREATION



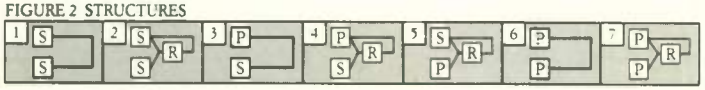
LA Synthesis Explained/LA Synthesis is component synthesis on the highest order. To create complex sounds, the D-50 starts with a very simple premise—build sounds from the ground up by combining different types of sounds together, and then experience the interaction of these sounds on each other. We start with individual elements of sound called Partials. Two Partials are combined to create a Tone, and two Tones are combined to create the Patch. (Figure 1) The D-50 can hold 64 Patches and 128 Tones. Each of the two Tones can be processed individually by on-board signal processing that is sophisticated enough to rival a rack-full of equipment, and includes digital reverb, digital parametric eq,

LINEAR SYNTHESIZER
SOUND SYNTHESIS TECHNOLOGY

digital chorus, digital delay and more. But before we go too far, let's get down to the basics, the building blocks of LA Synthesis—Partials.

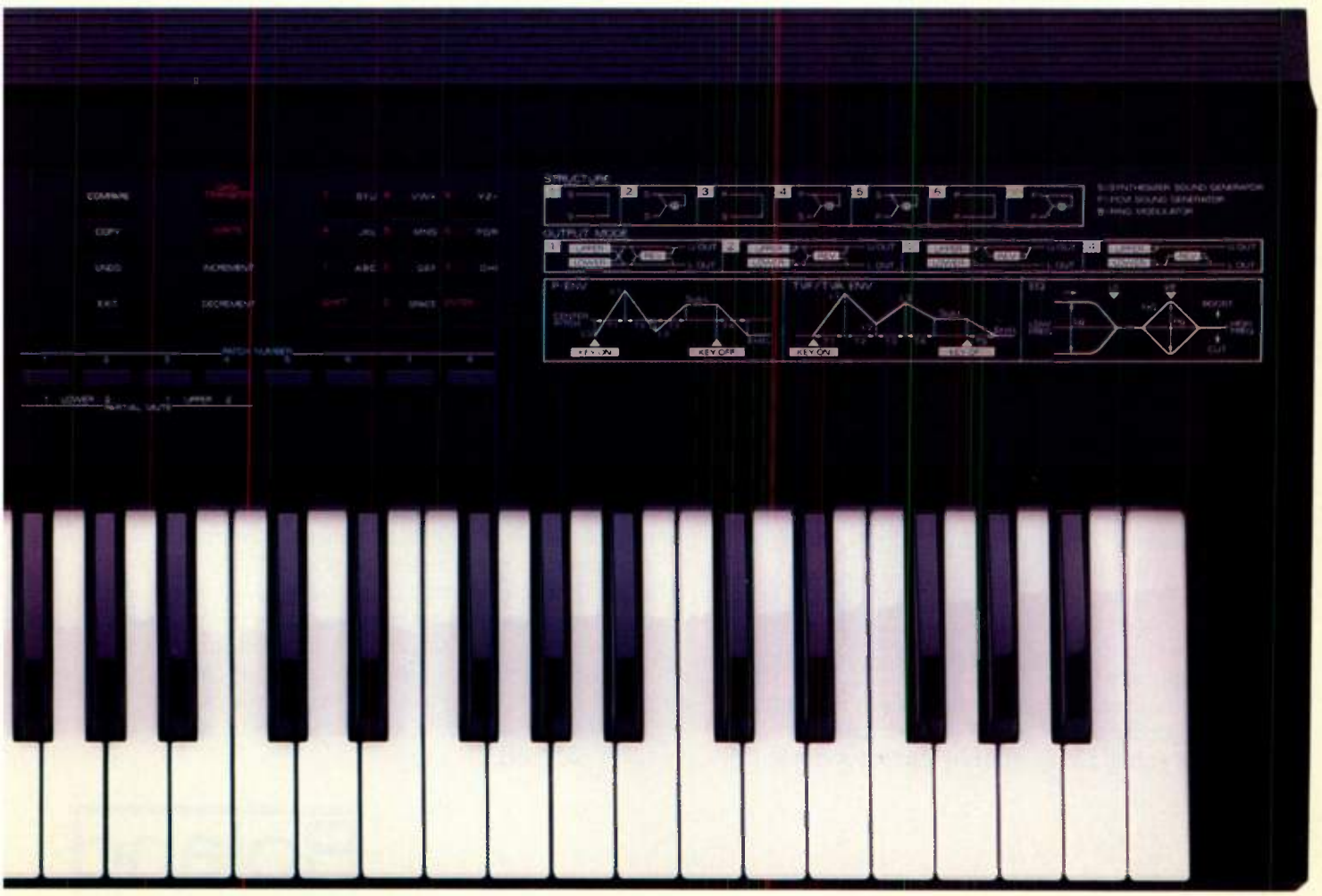
Synthesizer Partials/What is a Partial? A Partial can be either a digitally synthesized waveform, or a PCM sample. Each of the thirty-two Synth Partials contains all the components usually found in the hardware of an analog synthesizer, presented here as digital software. This includes the Wave Generator (to create a sawtooth or square waveform), the Time Variant Filter, the Time Variant Amplifier, three five-stage Envelope Generators and three digital LFOs.

FIGURE 2 STRUCTURES



In this way, even though the D-50 is a digital signal, programming the Synth Partial is very similar to programming on an analog synthesizer, (as these components react in the same way as VCO's, VCF's and VCA's on analog synthesizers) while offering sound synthesis capability beyond the most advanced digital synthesizer.

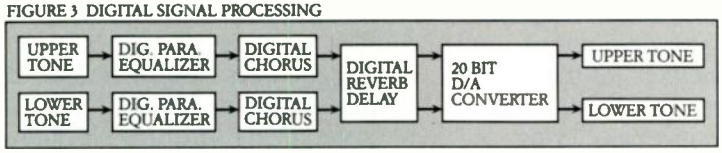
PCM Sampled Partials/A Partial can also be more than a digitally synthesized signal, it can also be a PCM sample. Resident in the memory (ROM) of the D-50 are over 100 carefully selected 16 bit PCM Sampled Wave Tables which can be used by themselves, combined with Synth Partials or combined with each other. The PCM Partials are carefully selected, and digitally processed so that they combine well with other Partials. Some of the sounds include a wide variety of the attack portions of percussive sounds: marimba, vibes, xylophone, ethnic instruments, grand piano hammer attack (with the fundamental removed), a variety of flute and horn breaths, a range of different string plucks and bows, nail files, guitars, and many more. The Wave Table library also includes Loop sounds and long samples, such as: Male and female voices, organs, pianos, wind and brass instruments, and also Harmonic Spectrum sounds, which are created by removing all of the fundamentals of a sound, isolating its harmonic components.



The sounds created by the D-50's PCM Waveform Generator are far superior to wave table samples found in other synthesizers, which are usually only one looped cycle in duration, and are usually no more than 5 milliseconds. In contrast, many of the PCM Partials on the D-50 are up to 256 milliseconds.

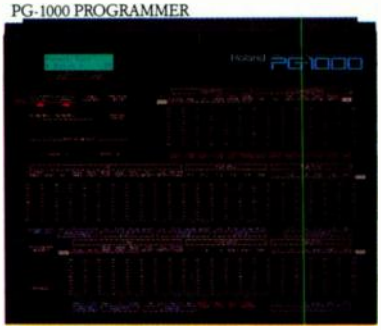
Structures/The combination of the Partials' operation modes can be set by selecting one of the seven Structures. (Figure 2) By choosing one of these Structures it is possible to combine two Synth Partials, or two PCM Partials, or a combination of the two in several different relationships. In addition, the Partials can be cross-modulated by the digitally-controlled Ring Modulator, which helps to create the complex harmonic environment for the resulting Tone. Unlike ring modulators of the past (which tended to be interesting yet unpredictable), the Ring Modulator in the D-50 is designed to track with the keyboard, ensuring the proper harmonic relationships as you go up and down the keyboard.

FIGURE 3 DIGITAL SIGNAL PROCESSING



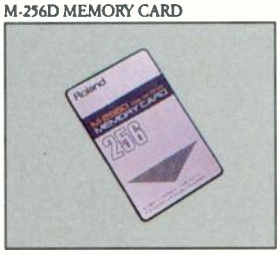
Built-In Digital Effects/The final routing of the signal before it reaches the output is through the digital effects circuitry. (Figure 3) But, far from being merely an add-on, the D-50's effects are as carefully thought-out as the rest of the instrument, and likewise just as integral to the creation of new and unique sounds. The first effect is the digital Parametric Equalizer, used to contour the equalization curve for the tone before it passes into the digital Chorus, or we should say Choruses, as the D-50 fields an arsenal of eight chorus circuits—all available simultaneously, configured in any of 16 modifiable presets such as panning chorus, tremolo, flanging and much more. Within each chorus there are parameters set up as to how these choruses interact for maximum effectiveness. Lastly, the signal passes through the digital Reverb, which can also function as a digital Delay, offering various room and hall sizes, gated (non-linear) reverb, reverse, stereo panning effects that can be routed to either or both of the stereo outputs. The awesome power of these built-in effects means that the D-50 requires literally no outboard effects processing. And just as important because all the D-50's effects are processed in the digital realm, they are completely noise free.

PG-1000 PROGRAMMER



A Mother of a MIDI Keyboard/The D-50 is also an excellent mother keyboard for your MIDI system, as it is totally dynamic, offering 61 keys in four different key modes (Whole, Split, Dual and Separate). In the Whole mode the D-50 is 16 voice polyphonic, while in the other modes it functions as two 8 voice synths, one for each Tone. All mother keyboard functions are programmable per patch including a separate transmit channel and a separate program change transmit. As the D-50 is truly bi-timbral it can function as two MIDI sound modules as each tone can receive on its own MIDI channel. All D-50 parameters and programs can be saved on Roland's new M-256D memory card which offers 32K bytes of storage in the size of a credit card. All of the D-50 functions can be programmed internally, or externally with the use of the optional PG-1000 programmer, which combines visual clarity and speed for the programming professional.

M-256D MEMORY CARD



Put It All Together/Taken as a whole, the D-50 represents more sound creation potential than most of the leading synthesizers combined. And just as important, it comes at a price that you can afford—$1895.00*. Of course, the only real way to find out for yourself is to play the instrument, but we'd like to suggest you do a little more. Go to your dealer, but before you try the D-50, try three or four other synthesizers first—really give them a good going-over. Then spend some time on the D-50. We think you'll find that the world of sounds you knew before, now seems to be black and white— while the D-50 has just exploded you into a universe of color. The new force has taken you by storm. RolandCorp US, 7200 Dominion Circle, Los Angeles, CA 90040 (213) 685 5141.

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SOUNDS natural

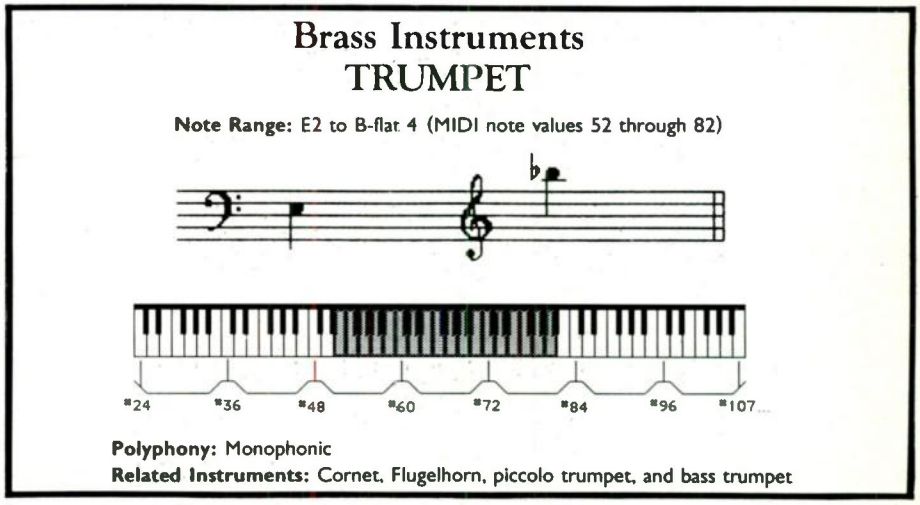
Part 1: Brass Instruments – The Trumpet

In the first of a series of articles, we describe the various ways in which acoustic instruments produce sound, and explain how to synthesize and sample them. *Text by Howard Massey with Alex Noyes and Daniel Shklair.*

PROGRAMMING

BEGINNING THIS MONTH, MT is reprinting excerpts from a fascinating new book written by three staff instructors at the Center for Electronic Music, a non-profit organization based in New York City which offers educational services in music technology. Entitled *A Synthesist's Guide to Acoustic Instruments*, the book presents a discussion of 25 acoustic instruments from the point of view of the synthesist – no matter what kind of axe he or she is using. (*A Synthesist's Guide to Acoustic Instruments* is published by AMSCO and is being distributed by Music Sales.)

Each instrument is examined in the same general way – with a discussion of its physical construction, commonly employed playing techniques, timbral analysis, and envelope characteristics, after which are presented tips as to how to go about synthesizing these types of sounds with generic analog synths, digital phase distortion synths (a la the Casio CZ instruments), and digital FM synths (like the Yamaha DX and TX synths). Last, but by no means least, each instrument chapter concludes with guidelines as to the best ways to record the acoustic sounds themselves and capture and process them in the sampler of your choice. We begin the series with the trumpet.



LIKE ALL BRASS instruments, the trumpet is essentially nothing more than a mouthpiece, some brass tubing, and a bell. The mouthpiece has a cup and a tapered back-bore, its tubing is cylindrical (as opposed to conical), and its bell is abruptly flared. It is one of the smaller cylindrical bore horns. A system of three valves allows the player to change the length of the tubing, by switching in extra sections in the middle – thus yielding a wider range of notes than would be available with tubing of a fixed length.

This is probably a good time to define cylindrical and conical tubing, since these are terms used to describe all brass instruments. Cylindrical tubing changes little in diameter from mouthpiece to bell – and the bell itself is usually not very pronounced. Like the trumpet, the cornet and trombone are cylindrical. Conical tubing increases in diameter from the mouthpiece to the bell – and the bell itself is very pronounced. The family of conical brass instruments includes the tuba, baritone horn, French horn, and flugelhorn.

How It's Played

A TRUMPET PLAYER'S lips channel air from the lungs into the mouthpiece. The lips themselves act as a sort of valve, regulating the flow of air into the instrument. The oscillating air, set into motion by the vibrating lips, travels down the entire length of tubing. When this column of air reaches the bell, part of the resulting wave is reflected back into the instrument. This then combines with incoming waves to produce what are known as standing waves (that is, waves whose frequencies remain constant due to this push-and-pull motion).

Changes in volume are achieved by the trumpet player's blowing harder into the mouthpiece. This also results in a change in timbre. Other changes in timbre and changes in pitch are made by altering the tension of the lips (thereby altering the air flow) or by changing the length of the instrument with the valves. As the lips vibrate at a given frequency, multiples of that frequency (harmonics) are generated and reinforced in the instrument: essentially, the trumpet acts as a resonator for these vibrations. When the lips vibrate faster than about 1500 times per second, the bell acts more like a megaphone than a resonator – resulting in little or no wave content being redirected into the tubing. This sets the limit of the highest pitch that can be derived from this instrument.

The trumpet, like the trombone (and sometimes other brass instruments), is occasionally played with a device called a mute which is inserted in the bell to alter the sound. One of the more common types of mute is called the straight mute. This device changes ►

Acoustics *"The timbre of the trumpet sound changes dramatically – not only with changes in pitch, but also as the volume of the sound is altered."*

PROGRAMMING

the radiating patterns of the trumpet's sound by inhibiting low frequencies. The resulting sound is typically thin and edgy.

Other types of mutes include the cup mute, bucket mute, plunger mute, hat mute, mica mute, and harmon mute. Some may cause virtually no fundamental frequency to be transmitted by the instrument. Conversely, others may decrease the upper overtone content of the trumpet sound. The harmon mute, in particular, is among the more tightly fitted of the mutes – meaning that virtually all of the air column that leaves the instrument passes through its fitted stem-and-plunger mechanism.

As we have said, when the trumpet player blows harder, the timbre changes significantly – and a brighter sound results because more overtones are generated. At high amplitudes, as the strength of the fundamental is doubled, that of the second harmonic quadruples, the third harmonic's loudness increases by a factor of eight, and so on. This increase in harmonics with an increase in amplitude will always result in a much richer timbre.

Timbral Analysis

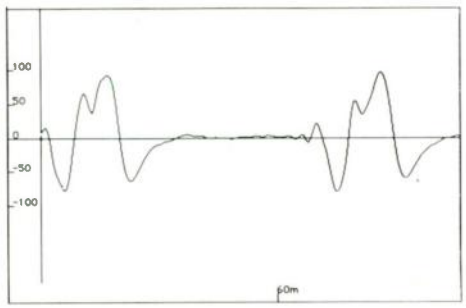
AS INDICATED ABOVE, the timbre of the trumpet sound changes dramatically – not only with changes in pitch, but also as the volume of the sound is altered. Played very quietly, the instrument will yield the fundamental, second, and third harmonics – the fundamental being about twice as loud as the second harmonic. When the trumpet is played at a slightly greater volume, the second overtone begins to dominate, and harmonics up to about the sixth are introduced. At a medium volume, higher harmonics are generated; the second, third, and fifth overtones dominate; and – with the exception of a peak at the ninth harmonic – the sixth through eleventh overtones steadily decline. When the instrument is played loudly, the second harmonic is significantly louder than the fundamental, the third through sixth harmonics increase in strength, and the tenth and eleventh harmonics drop rather sharply. Finally, at very high volumes, the second harmonic will be much stronger than the fundamental. Those up through the eighth increase drastically in strength, while the ninth, tenth, and eleventh harmonics drop off radically. The general pattern is clear – louder volumes yield brighter or more piercing sounds.

The bell of the trumpet concentrates the high frequency components of a sound into a narrow beam, while it spreads the low frequency components more or less equally in all directions. Among brass instruments, this phenomenon is unique to the trumpet. The effect is that the timbral content of the sound is quite different when heard while facing the bell than when standing to one side of the instrument. (We'll talk more about this in the section below on sampling the trumpet.)

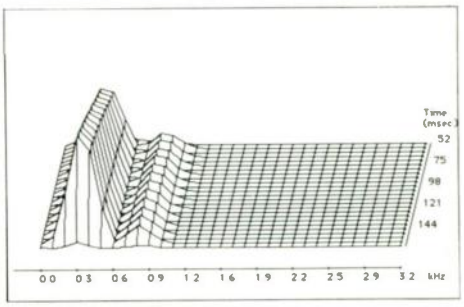
When a trumpet player first vibrates his or her lips, it takes a short period of time for the wave of air to travel the full length of the tubing, to be reflected back by the bell – and to then be combined and to interact with incoming air. Thus, at the beginning of a trumpet sound, there will be a short period of time during which no standing wave exists. Instead, the vibrating air leaves the bell unreinforced and unresonated – producing a short burst of inharmonic overtones. This accounts for the characteristic "spit" at the beginning of a trumpet note. A rapid increase in the overall loudness of the sound follows, as the standing wave builds in intensity and finally reaches a steady state, which results in a smooth, stable timbre.

A well-trained trumpeter playing a well-constructed instrument produces less of a "spitting" effect at the onset of each tone than would a stylist or a beginner. This is because a trained player takes less time to adjust the lips to the internal pressure of the instrument. Still, the "spit" effect will, to some degree, always be a characteristic of the trumpet sound.

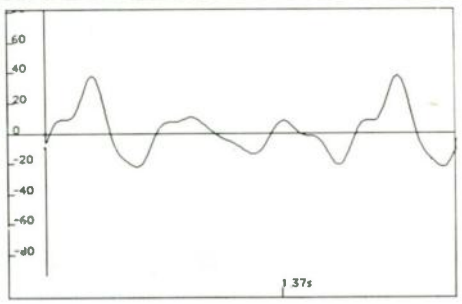
The trumpet is a sustaining instrument – meaning that as long as you blow into it, the sound will continue. Its after-ring, or release time, ranges from short to quite short, depending upon the acoustics of the playing environment.



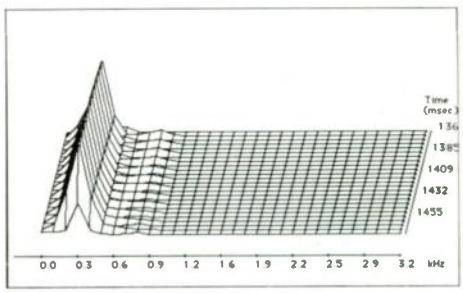
Start of trumpet sound: *After the initial "spit" and attack, this very unusual waveshape manifests itself; note the very long "dead" areas between periods.*



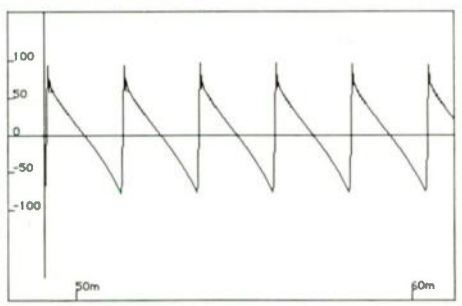
Start of trumpet sound (FFT): *This shows the sound just after the "spit," as the standing wave begins to manifest itself note the clear harmonic content and the presence of many low overtones.*



Sustaining portion of trumpet sound: *Here we have a different complex waveshape, though one which is quite regular.*



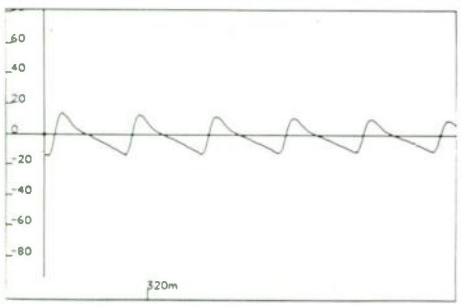
Sustaining portion of trumpet sound (FFT): *Note that upper harmonics are still present and that there is a good deal of timbral stability in this sound.*



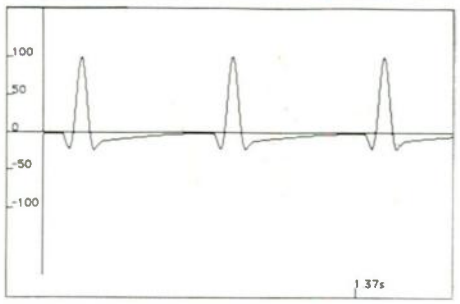
Start of subtractive trumpet patch: *These are very angular, somewhat "spiky" sawtooth waves; note the absolute timbral regularity.*

Subtractive Synthesis

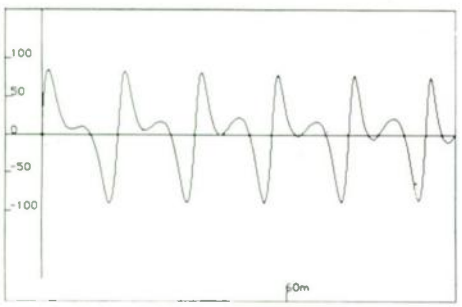
MOST BRIGHT, BRASSY sounds created by subtractive synthesizers start with the sawtooth wave, since that is the waveshape with the richest harmonic content. So, you'll want to start by setting both oscillators to a sawtooth wave, tune them in unison (at the middle octave, or 8′ setting), and then synchronize them for the strongest basic sound. You



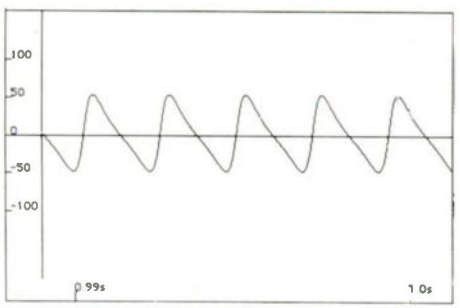
Sustaining portion of subtractive trumpet patch: *The sawtooth waves at this point are considerably smoother, reflecting the actions of the filter EG on the lowpass filter; note also the decreased amplitude, similar to that of the original acoustic sound.*



Sustaining portion of digital Phase Distortion trumpet patch: *Note the very long "dead" areas between wave cycles, similar to that found in the beginning of the real trumpet sound.*



Start of digital FM trumpet patch: *An unusual but stable waveshape, slightly similar to that produced by the real trumpet but without the long "dead" areas between wave cycles.*



Sustaining portion of digital FM trumpet patch: *These very sawtooth-like waveshapes are actually very similar to those produced by the subtractive trumpet patch.*

might also try routing a bit of one oscillator's output to the other's input (a technique called cross-modulation) – or to the filter's input – to induce a slight inharmonic content in your sound.

In any event, send approximately equal amounts of signal from each oscillator to the mixer. The low pass filter's cutoff frequency should be set at a bit less than half, but you'll need to tune this parameter by ear, as it is critical to the patch. Add a small amount of resonance, and route a moderate amount of keyboard voltage to the filter. This will cause the higher notes to be significantly – and realistically – brighter than lower ones.

The real key to this sound, as with most synthesized sounds, lies in the actions of the envelope generators. The filter EG, in particular, contributes quite a bit to the naturalness of the sound. So, you'll need to route a fair amount of controlling signal from the filter EG to the filter itself. Set its attack time at just less than instant (about 12msec); the decay should be fairly long (about 50msec); the sustain level, medium; and the release time slightly longer than the attack time (about 15msec). With these settings, you will get a good simulation of the natural timbral movements in the trumpet sound. Your amplifier's EG should be set in much the same way – perhaps with slightly faster attack and release times (about 5msec each).

If your synthesizer has a third EG that can be utilized for pitch change, send a little controlling signal from it to the master oscillator. This will cause the pitch of the note to bend upward slightly after the initial attack. This subtle pitch change will help make the sound more "real."

If you have the luxury of yet a fourth EG, you can use it to shape a little noise with a sharp attack and short decay (no sustain or release) to simulate the "spit" effect at the beginning of the sound. If your keyboard is velocity-sensitive, you'll probably want to use the sensitivity to open the filter a bit more as the notes are struck harder. This makes the louder notes sound brighter – a hallmark of the trumpet sound. Finally, a small amount of LFO signal (sine or triangle wave at approximately 5Hz) can be routed through a controller to give you the option of adding a light vibrato effect when needed.

The accompanying diagram shows the subtractive system patch for a trumpet sound.

PD *"DCA envelopes are both given very complex shapes. Both are undergoing rapid triple attack-reattacks so as to simulate the natural sound of the trumpet player's 'tonguing'."*

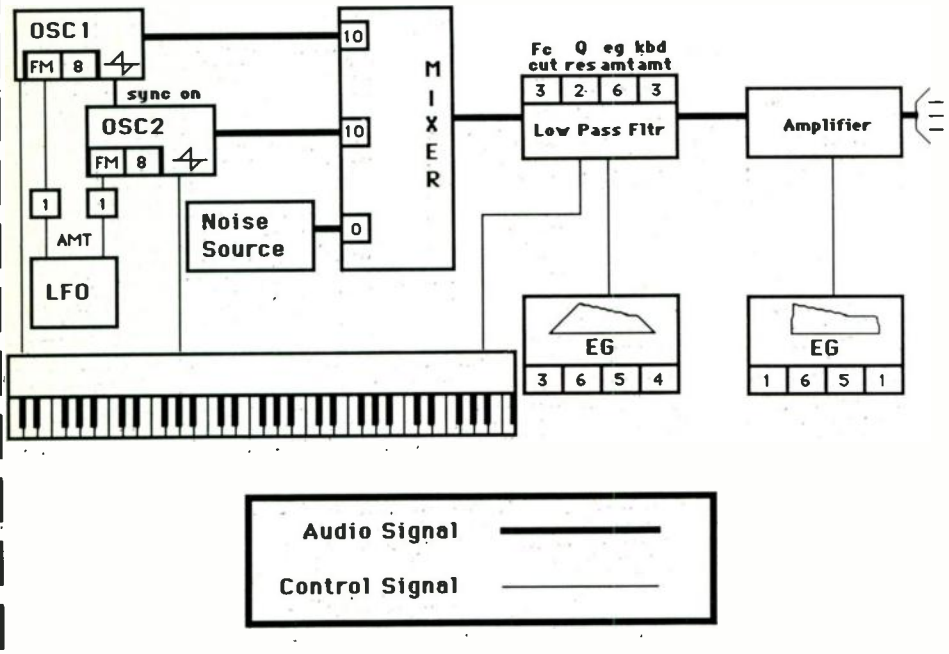
PD Synthesis

THIS SOUND IS created via digital Phase Distortion synthesis on the Casio CZ101, with the 1+2′ configuration, but with no detuning between the two lines. The best waveform to select for both oscillators is the pulse wave (waveform option 3 on this instrument), because it is one of the richer waveforms available (it contains the highest harmonics in the greatest strength). The DCO envelopes should be completely inactive, as we won't need any kind of pitch change in this particular sound.

The DCW envelopes will also be fairly inactive, providing only a simple fast attack and slow decay configuration, with no sustain. Add moderate amounts of keyboard following to both so that the high notes are not un-pleasantly over-bright.

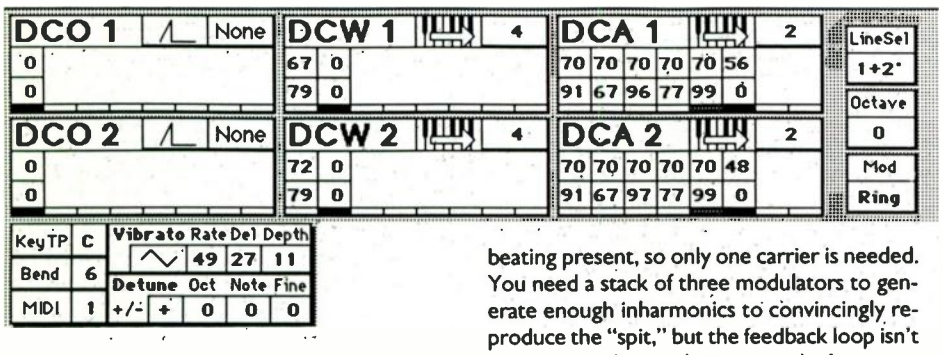
In contrast, the DCA envelopes are both given very complex shapes. Both are undergoing rapid triple attack-reattacks so as to simulate the natural sound of the trumpet player's "tonguing." Set them both at the maximum sustain level, since the trumpet is a sustaining instrument. Apply moderate amounts of keyboard following to both DCAs so that higher notes will produce this tonguing effect a little more rapidly than lower ones.

The application of ring modulation is of par- ►



► ticular interest here. Because both DCOs are precisely the same frequency, the ring modulation will add a few high overtones to the sound. This helps to increase the overall brightness of the sound. A small amount of delayed vibrato helps to simulate the typical pitch waver that occurs in long, sustained trumpet notes.

Finally, if you were working with a velocity-sensitive keyboard, you would want to route a small amount of its control signal to the DCW – and slightly less of the same to the DCA. This causes sounds to be brighter when the keys are struck faster – an accurate simulation of how the trumpet sound changes when played louder.



FM Synthesis

BECAUSE A TRUMPET sound is very rich in overtones – inharmonically at its outset (during the "spitting" portion) and harmonically during its sustain portion – you need quite a few modulators to simulate its sound. But the trumpet has a strong fundamental, with little beating present, so only one carrier is needed. You need a stack of three modulators to generate enough inharmonics to convincingly reproduce the "spit," but the feedback loop isn't necessary in that stack since you don't want to go so far as to induce noise. It will be far more useful to make the feedback loop available on a single modulator to act as a phantom, "seventh" operator. A quick glance at the available algorithms indicates that algorithm 18, with its single carrier, stack of three modulators, and single modulator with feedback, is the best choice.

In this algorithm, operators 2 and 3 are both acting as single modulators, and both have been set up in a 1:1 frequency ratio with the carrier (operator 1). This will produce sawtooth-like waves (with all harmonics present). However, operator 3 has been detuned slightly, and its envelope settings are substantially different from those of operator 2. These EG settings provide some sustaining timbre from operator 3, while operator 2 dies away rapidly to a sustain level of 0. You'll also want to open up the feedback loop on operator 3 to maximum in order to increase the brightness of the sound. The stack (operators 4, 5, and 6) is also detuned, and the upper modulators (operators 5 and 6) are set to non-whole number frequency ratio values so that they contribute the inharmonic content of the sound (which will simulate the "spitting" part of the sound). Note that the envelope settings for operator 6 (which causes the highest overtones to be generated) ensure that it fades away the fastest, in simulation of the acoustic sound.

You should also use sparing amounts of keyboard rate scaling in order to simulate the rapid timbral changes that occur when high trumpet notes are played – with a small amount of keyboard level scaling added in order to slightly "roll off" the effects of the feedback operator (operator 3) at the topmost registers. There is also a small amount of velocity sensitivity on the carrier and on modulators 2 and 3, to give you a more performance-responsive patch; that is, as keys are struck more quickly, the resulting sounds are slightly louder and at the same time much brighter.

Of particular note is the use of the LFO – it is set to a square wave running at its highest possible frequency (about 60Hz). This relatively fast-moving periodic signal is then routed directly to the amplifiers in operators 5 and 6 (the modulators responsible for the inharmonic content), and, to a lesser extent, to operator 2 (the detuned sawtooth wave). The purpose here is to simulate the characteristic "growl" of the trumpet – an effect derived from a specific lip technique. Because operator 6 has maximum velocity sensitivity, this effect will only become apparent when a key is struck quickly.

Sampling

BECAUSE OF THE broad dynamic range of the trumpet, it is best to sample this instrument with a dynamic, rather than condenser, microphone. Be sure, however, that the mike you choose has a good high-frequency response – as this instrument characteristically produces such a bright sound.

You'll probably want only a little bit of room ambience, so try to sample this instrument in a small room, and place the mike six to twelve inches away from the bell. As mentioned above, the bell of the trumpet has the effect of beaming the high-frequency components straight ahead, while it disperses the lower-frequency components of the sound more or less omnidirectionally. With this in mind, you should position the mike directly in front of the bell if you want a very bright sample – and a bit off to the side if you need a warmer sound. Again, this beaming and dispersion effect is

more pronounced in the trumpet than in any other brass instrument.

In general, little signal processing is required when sampling a trumpet sound. If your mixing board is clean – and your recording system can handle the high sound pressure level (spl) that emanates from the instrument – you shouldn't need any compression or limiting. If your mike placement is good, you will probably not need any equalization either. It's best to take at least four samples of the trumpet per octave (that's one every three semitones for those of you who weren't born with internal calculators). These samples need not be particularly long – but just long enough to get past the initial attack and into the stable, sustaining portion of the sound. One-second samples should be long enough – and since the trumpet is a sustaining sound, looping can do the rest.

Needless to say, the loop points have to be chosen carefully. You should be able to hear (or see, if you're using a waveshape editing program) a clear attack portion that contains the "spit," followed by a stable sustaining portion as the standing waves are finally generated within the instrument. You'll need to loop in that latter area, but be careful to avoid any vibrato points. We found that the best looping points generally occurred at least 300msec into the sound. The loop itself should be moderately long – on the order of at least 100msec. You can then either truncate the sound at the end of the loop and use a slight amplifier-envelope release time to simulate the natural ambience – or set your sampler to jump to the end of the sample after key release, to give you the actual ambience you recorded.

Because of the relatively small amount of processing involved, you should find the trumpet to be one of the easiest acoustic instruments to sample. ■

On/Off	1 Car ☒				2 Mod ☒				3 Mod ☒				4 Mod ☒				5 Mod ☒				6 Mod ☒			
EGRate	59	41	24	47	41	51	15	33	46	32	22	50	66	39	22	77	48	55	22	52	77	56	20	70
EGLev	99	86	90	0	90	0	0	0	99	86	86	0	99	61	62	0	98	61	62	0	99	0	0	0
Scaling	0	A-1		0	0	A-1		0	0	A*4		10	0	A-1		0	0	A-1		0	0	A-1		0
Curve	-lin		-lin		-lin		-lin		-lin		-lin		-lin		-lin		-lin		-lin		-lin		-lin	
Output Level	99	Vel 2	RSc 2	AM 0	82	Vel 1	RSc 2	AM 1	80	Vel 1	RSc 1	AM 0	40	Vel 0	RSc 0	AM 0	72	Vel 0	RSc 0	AM 3	47	Vel 0	RSc 7	AM 3
Freq	M	Coar	Fine	Det	M	Coar	Fine	Det	M	Coar	Fine	Det	M	Coar	Fine	Det	M	Coar	Fine	Det	M	Coar	Fine	Det
	r	1.	00	0	r	1.	00	+6	r	1.	00	0	r	1.	00	-4	r	.	79	-1	r	16.	35	0

PEGRat	94	67	95	60	LFOWave	Spd	Del	PMD	AMD	PMS	Sync
PEGLev	50	50	50	50	Square	99	0	0	47	0	Off

Algorithm	18
KeyTP	C3
Feedback	7
KeySync	On

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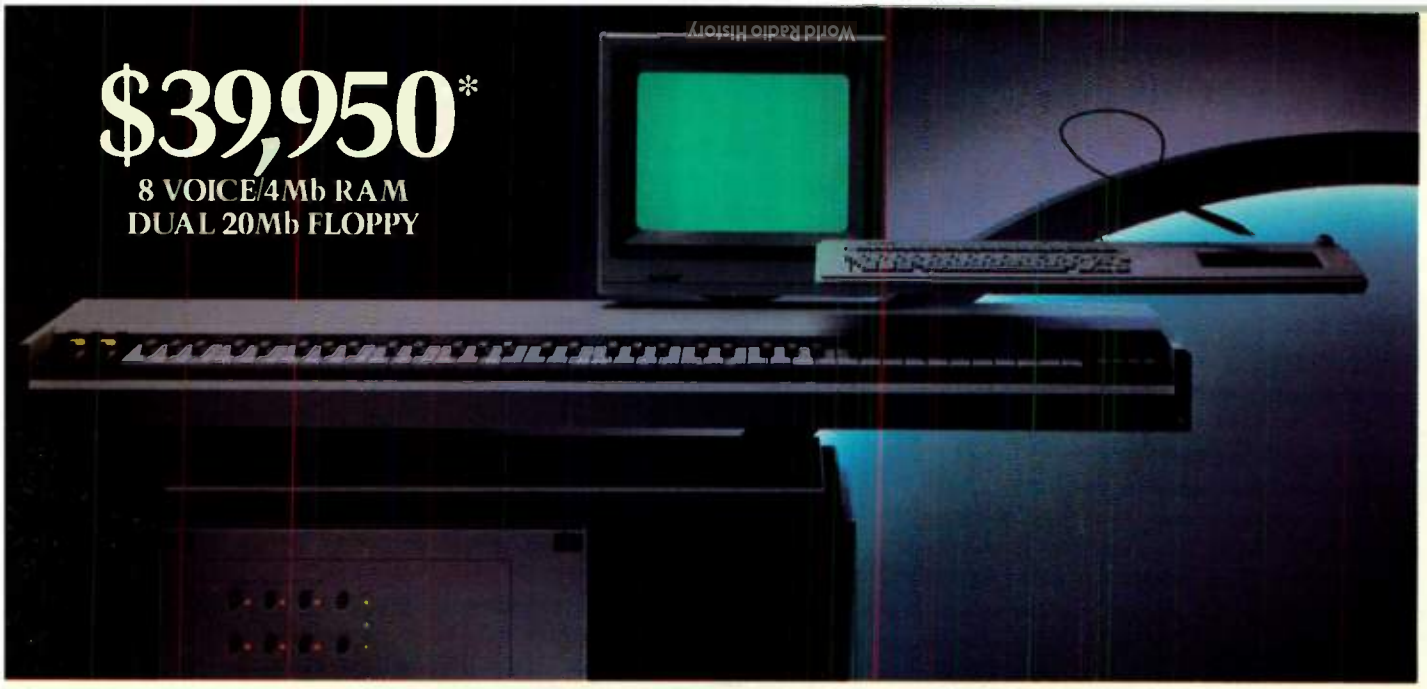

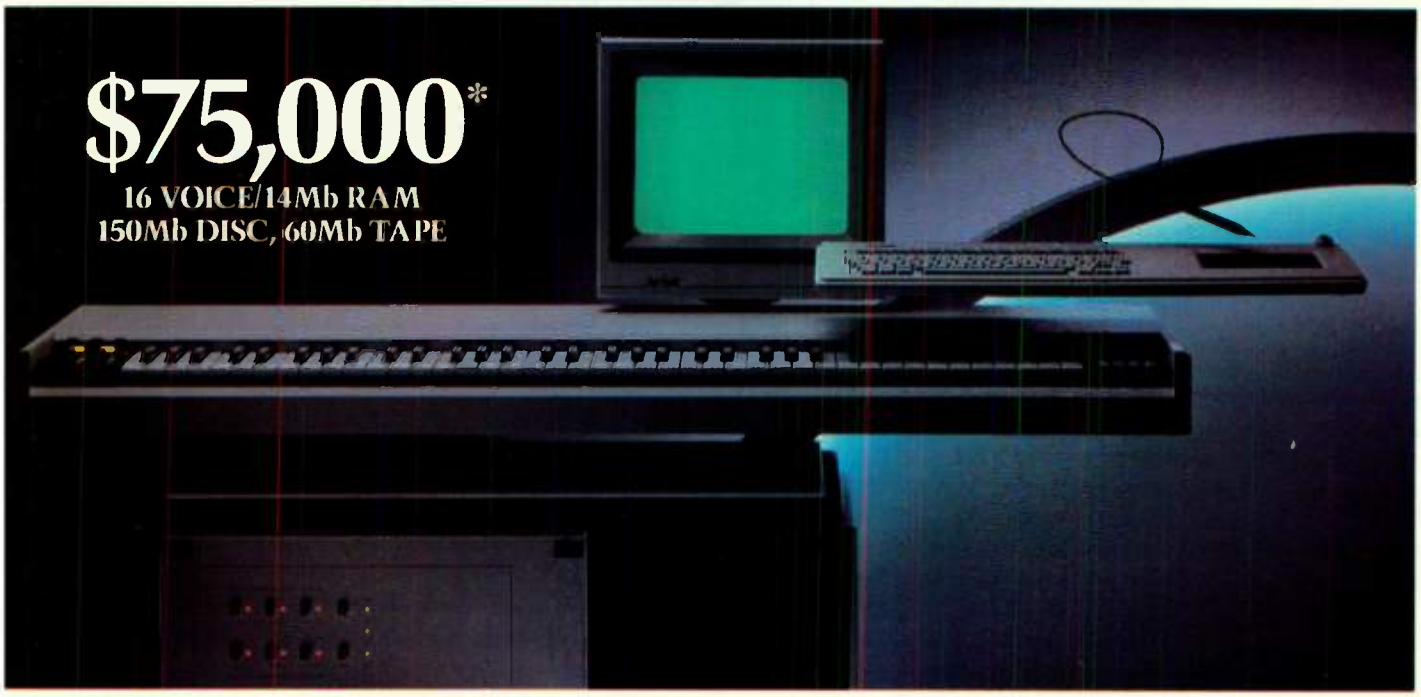

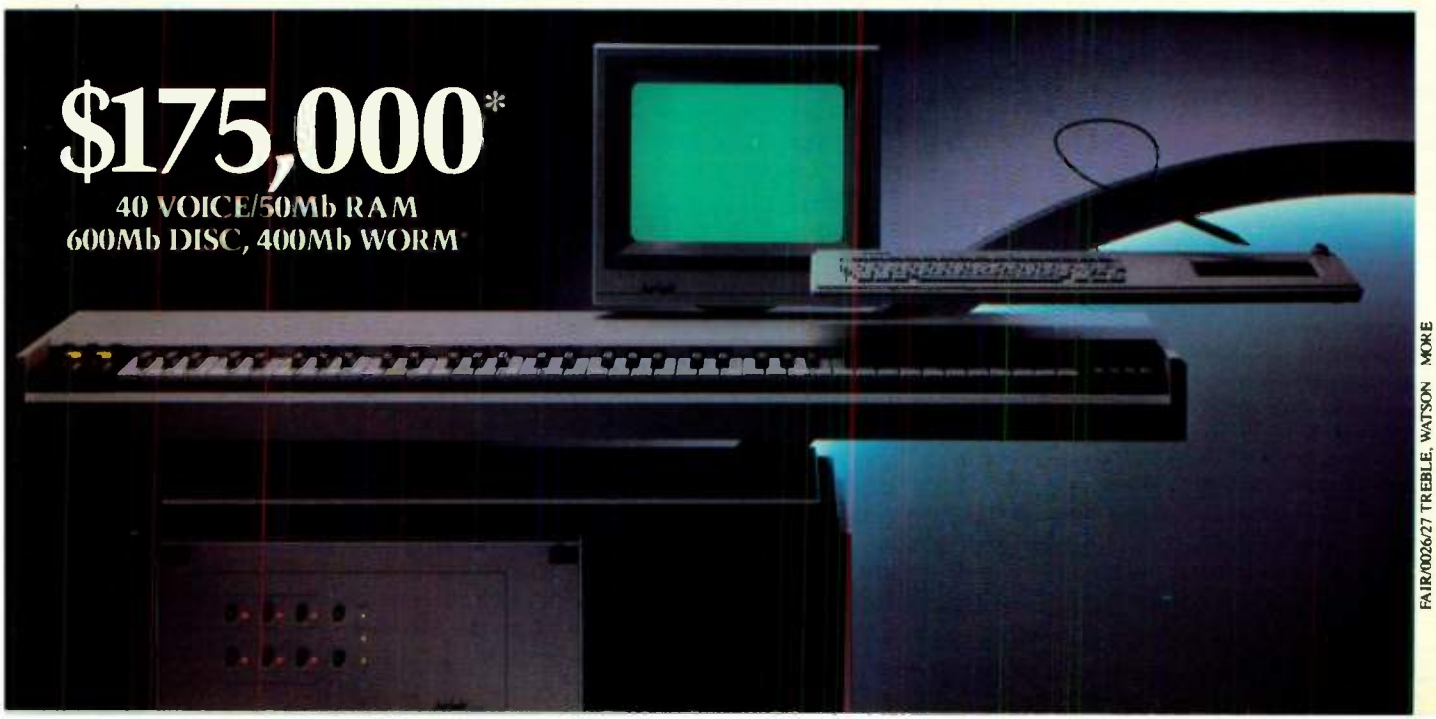

FAIR/0026/27 TREBLE, WATSON MORE

REGENERATION

SAMPLING

Many electronic instrument manufacturers are working to avoid obsolescence by offering updates to their existing products. We look at three such revisions in the rapidly changing world of sampling. *Text by Chris Meyer and Bob O'Donnell.*

IF WE WERE asked off the cuff whether synths or samplers become obsolete more quickly, we would have to say, "Samplers, of course." New ones are announced every few months that are leaps ahead of the previous generation – witness the likes of the Casio FZ1, which will undoubtedly lead to a preference for 16-bit machines and perhaps a bloodbath for the 12-bit instruments.

But one advantage that samplers have over synths with regard to obsolescence (which, in fact, may be due to the current state of affairs) is that synths don't seem to get updated very often, whereas virtually every sampler currently available has received (or is receiving) both software and hardware updates – from facelifts to overhauls. The capabilities of the updated instruments are generally *much* greater than the originals and consequently, deserve a second look. Here are a few of the most recent revisions, to keep you up to date.

Oberheim DPX1 Version 1.3

WE BEGIN WITH Oberheim, who have been busy adding features. We now have the version 1.3 software release for the DPX1 (which came out very quickly, I might add), that adds a whole group of features, plus hardware modifications that add individual outputs and provide the capability to use Optical Media International's CD-ROM unit.

Before going any further, I have to admit to being among the first to knock the DPX1 – who needs playback only? A person needs to be able to sample, and edit those sounds! However, here in the real world of trying to do MIDI sequencing with a sampler, I often find that I could definitely use more than one disk's worth of sounds to pull things off – and I certainly didn't need to spend extra money on features redundant with my existing sampler (ie. sampling and editing).

The DPX1 sounds better than an EII or a Mirage on their own disks, and almost as good as my Prophet 2002 (I swear – I wanted it to sound just as good, but it falls a hair short). It is also rack-mountable, and lists for $1000 less than another identical 2002. So, I was curious to see how the new features swayed my vote.

The most eye-raising software change in version 1.3 is the ability to write your own disks. Oberheim claims this is just to ease archiving of sounds. What you (and I) the user are more anxious to learn is that it actually

loads its own disks faster than the source disk (since some of the conversion to its own internal format has already been done – particularly significant in the case of the EII, which took some 10-15 seconds extra to convert).

The new software also makes the DPXI "more compatible" with the samplers it emulates (no joke intended). For the Prophet 2002, it now also supports Fixed Output mode and alternate release on envelopes; for the Mirage, it now implements looping and fading of samples more accurately; and for the EII, the hold pedal now reacts as it would on an EII itself. There are also a handful of other small but useful enhancements, such as the ability to have single samples requested and received via the MIDI Sample Dump Standard.

The individual outputs modification was something that was requested from the very day the DPXI was announced, and users will be glad to see it finally there. Yes, it indeed works just as it would on the source sampler, but it actually has a few enhancements. For one, the base channel may be rotated around if using a 2000 disk in Mono mode (see *Getting the Most from Mono Mode*, MT September '86 for further reference). Second, pitch-bend and modulation may be received per sample in Mono mode, which the 2000 could *not* do. Since I intend to get into guitar synthesis, I was particularly excited about this one – Mono mode operation, including individual articulation per string/voice, is all but essential to most players. However, there is a slight bug in the current software revision concerning this feature – it is possible for the voices to become confused and start getting detuned from each other if they are stolen while being bent. Thankfully, Oberheim is already working to cure this one.

And finally, the CD-ROM interface. For those who do not know, Optical Media makes a CD-ROM unit (the CDS3) for the EII that allows about 500 disks' worth of information to be held on a single CD. (They now have two disks in this series.) It transfers sound via the RS422 port on the back of the EII, and the DPXI, with this revision, also sprouts such a port. This means that "disks" get loaded faster than a normal floppy would, and handling them is much easier – just call them up on the CDS3's handy-dandy hand-held remote.

Load time with the DPXI is not quite as fast as the EII, since it must spend the aforementioned 10-15 seconds converting the special EII format into 12-bit linear. The DPXI loads all of the OMI banks correctly, but there seem to be problems with some of the banks that have positional crossfades – holes are left in a couple of keyboard maps. Also, if one attempt to abort a transfer in progress from the OMI hand-held remote, the DPXI will lock up the system and both units need to be switched on and off again. And finally, the CD-ROM system is pricey – $1995 for the player and remote, and $795 to $995 per CD. However, if all you intend to do is play back presets (I mean, that *is* what the DPXI is all about), you can buy a DPXI, a CD-ROM, both CDs, and a mother keyboard for what an EII costs – and have a smaller, better feeling, better sounding system, with 1000 disks thrown in. Of course it will lack any editing features, but it is an angle to think about.

Combining the DPXI's current updates with the newly announced ones (built-in hard disk and the ability to read S900 disks), it is obvious that Oberheim is committed to keeping this child updated and healthy. Will I buy one? A very strong "maybe," leaning towards "yes." Will you? Well, like I've said before about Oberheim's slightly off-base offerings, "If you need one, it's great." You decide. ■ CM

Sequential Prophet 2000, 2002, 2002+

CONSIDERING THE POPULARITY of rack-mounted MIDI equipment, it is surprising that the Sequential Prophet 2002 and the Akai S900 have been the only full-featured 12-bit samplers available for the past year and a half. Although the Akai lacked some of the features of the Prophet (and the sound quality issue between these two is strictly a matter of subjectives – do you want accuracy or bite?), its friendly user interface and individual outputs have given the S900 an edge over the 2002 in the marketplace.

The indifferent user interface is the same as always (a whole new machine might as well be released as opposed to trying to remedy it in a modification), but Sequential has finally addressed the output issue – and a few others – with a pair of updates and the release of the Prophet 2002+.

The 2002+ is essentially a Prophet 2002 with eight additional ¼" jacks on the back (one per voice), a new keyboard mode called "fixed output" to address these outputs, the addition of crossfade looping, and a script

SAMPL

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►"plus" added to the front panel. "Fixed output" gives a fairly basic, non-editable voice-to-output assignment – samples 1 and 9 always go to voice 1; 2 and 10 always go to voice 2; 3 and 11 to 3, and so on. This is not as fancy as some other units (such as the S900, or even Sequential's own Studio 440), and requires a little bit of foreplanning (to make sure that two sounds which may be playing at the same time are not sharing the same voice and therefore cutting each other off), but to be honest, it actually does what is needed most of the time.

The crossfade looping feature also picks certain defaults – linear crossfade, length as long as possible – which again is not as full-featured as some, but ends up being the best solution 90% of the time. Crossfade looping works for both sustain and release loops in either a unidirectional or bidirectional format. The effect sounds a little rough for fades under 1K or 2K in length, but it is otherwise very smooth. Other features in this update which should be of interest to Prophet 2000 owners include the ability to transmit from the local keyboard and wheels in Mode 4 (Mono mode), and the ability to assign and transmit the second release pedal on the same MIDI control number as everyone else's hold pedal. This hardware and firmware update is available for all Prophet 2000s and non-"plus" 2002s.

The second update expands the memory to 1 Megaword of 12-bit samples. This is not continuous memory, but acts as two banks of 512K (like the EII+). This second bank is just like the first – it can be loaded and sampled to, played from, and remotely edited (via Sound Designer or whatever) just like the first one. Studio bodies may be disappointed at not getting double-length samples out of the bargain (only one bank is active at a time), but it is nice on stage to be able to load up two disks worth ahead of time (nearly a two-minute proposition) and have them instantly available for performance. Having more RAM on hand also makes it nicer to do raw sampling – twice as many samples can be taken before saving to disk or resorting to the terminal support package. This update also consists of both hardware and firmware, and is available for all three variations of the Prophet sampler.

The Prophet 2000 and 2002 are already out of production, and the 2002+ may be out of production by the end of this year. Yet although their 2½-digit LED display looks even more frugal now than it did back in 1985, they still sound as good as any other 12-bit machine, and will no doubt one day be looked back on fondly as good examples of the firsts in their field. Given that, even though the updates have the same "here's all that you really need – come here and figure it out" feel as the original machines, it's nice to see Sequential making some effort to keep their samplers (and their owners) current before they're gone. ■ CM

Prophet *"The updates have the same 'here's all you really need – come here and figure it out' feel as the original machines, but it's nice to see Sequential making some effort to keep them current . . . "*

SAMPLING

Roland S50 Version 2.0

WHEN ROLAND FIRST released the S50, they made a point of explaining that the system would be upgradable, and that updates would be occurring in the not-too-distant future. Like other available samplers, the internal configuration is basically a computer which can be made to perform different functions depending on the software which is loaded into it. (The S50's operating system is stored on each data disk, so updates can be done with software only – thereby eliminating the need to make hardware modifications or to replace EPROMs.)

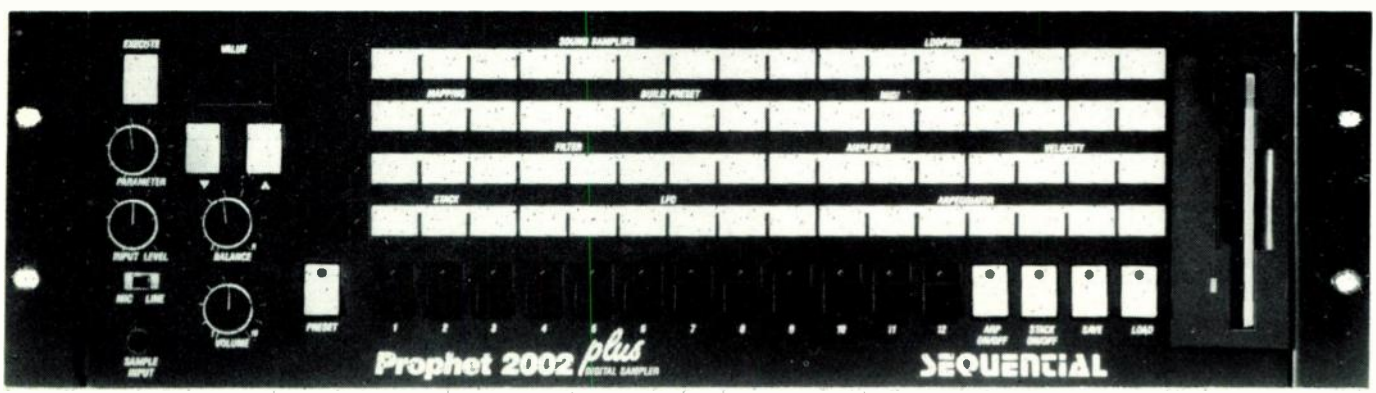
Well, the not-too-distant future has arrived, and with it has come version 2.0 for the S50. The new software adds a number of new functions (including the ability to convert 1.0 sound files into the new format) and some refinements to existing functions. The best part about it, though, is that it is *free* to existing S50 owners; there's not even a minimal update fee. (Roland has been giving away free samples via their Sound Bank program, but this is a whole other category.) Needless to say, this is an encouraging sign, and there is the promise of even more updates to come.

Probably the most important new feature implemented in version 2.0 is the ability to operate on multiple MIDI channels. What this means is that four different patches can be played simultaneously (with the full keyboard range) on four independent channels. In other words, the S50 can function as four independent instruments, one of which can be played from the keyboard itself, if you so desire. Each of these patches can have a certain number of voices allocated to it and can then be assigned to one of the four multiple output jacks. A patch can only be assigned to a single output, however, and as a result, the multiple outs can only be made use of via MIDI. Therefore, unless you assign the different patches to the same MIDI channel and plug the MIDI Out of the S50 into its MIDI In, you can only use one output when you play from the keyboard. (Roland claims that you'll be able to use multiple outputs from the keyboard with the next revision, but this should have been taken care of in version 2.0.) The new multiple channel operation is *not* the same thing as Mono mode, but it *is* a vast improvement over the original.

Another nice feature included in the update is support of the optional DTI00 digitizer tablet, with which you can do free-hand wave editing, wave drawing and envelope breakpoint manipulation. (Hand-drawn waveforms don't always give the most sonically impressive results, but they can be a lot of fun.) To be able to change the shape of the envelopes you obviously have to be able to see them and, not surprisingly, the new software also offers just such a graphic display (on the connected monitor, of course.)

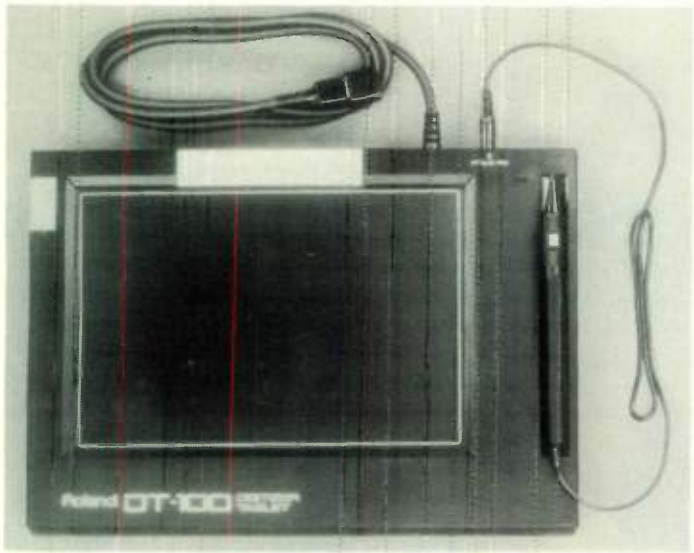
The number of available memory locations for tones has been doubled to 32, even though the total memory remains the same. Roland has simply increased the control you have over available memory by allowing you to split it into finer divisions (this is true for a number of features). And speaking of finer divisions, version 2.0 also introduces the concept of "subtones," which use the same wave data as a regular tone but have different parameters (such

Roland *"Probably the most important new feature implemented in version 2.0 software for the S50 is the ability to operate on multiple MIDI channels."*



as loop points, envelopes, and so on). The benefit of sub-tones is that they *don't* store the wave data a second time and thereby save precious memory.

The looping functions have also been improved with version 2.0. The S50 now has two types of automatic looping and three types of loop displays. In addition to the original display, you can now view the entire wave at once to ease the setting of rough loop points, and also fine-tune the loop by means of a highly specific, oscilloscope-type display.



The sampling process has been simplified with the addition of pre-trigger and previous sampling functions. The pre-trigger option allows you to start sampling 10-100msecs before the threshold point so that you can be sure that you capture a sound's entire transient, and the previous sampling option actually records the sound prior to your hitting the Enter button. Now while this might sound like magic, the way it actually works is that the S50 starts sampling continuously as soon as the "previous" function is engaged and constantly updates the memory location that you've chosen, so that when you hit Enter the sampling process stops and the previous 1.2 seconds (or whatever length you chose) is stored. This clever little feature can be particularly useful if you sample off of CDs or tapes, because you no longer need to rewind the CD (or tape) and then guess when you should hit Enter on the S50. Instead, all you have to do is hit Enter *after* you hear something you like and it's been sampled.

Roland has increased the usefulness of the video display with a tone map feature, which allows you to display the parameters (one at a time) for all 32 tones at once. With it, you can easily keep track of, compare and edit the various tones from both the Tone Play and Patch Play modes.

Other helpful features include the ability to perform a variety of SysEx wave and patch data dumps, combine waveforms, truncate samples with finer resolution, label each disk and add a 48-character note, control pitch-bend via aftertouch, and finally, you can now enter parameter values with the 10-key pad as well as the alpha dial.

With version 2.0 software, the S50 finally does what Roland's initial hype claimed it would and, on paper, it certainly appears impressive. Not many manufacturers are willing to give away *anything*, particularly something that adds this many new capabilities, and so Roland deserves special praise for offering the update at no charge.

The S50 is a powerful instrument, and with new software revisions (or perhaps completely new applications) already in progress, it promises to be an important contender in the sampling marketplace for quite a time to come. ■ *BO'D*

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BROADCAST QUALITY

MUSIC

Though they consider themselves a "guitar-based" band, Cutting Crew, whose hit song '(I Just) Died in your Arms' was number one in eight countries, have made extensive use of synths and guitar synths, and plan on making even more use of them in the future. *Interview by Tim Goodyer.*

ONE RECURRENT THEME in artist interviews is the commercial crossover between the United States and Britain. How many times have we read about "the guys that left the States to crack the British market first?" Or the band that are big over here but unheard of in their homeland?

One of the reasons for the differences between the two markets is undoubtedly their respective media and chart systems. Over here we have MTV, regional radio and the Billboard and Cashbox charts. The British, on the other hand, operate a centralized system based around one national radio station, Radio 1, and the Gallup chart. They also have a weekly chart TV show in *Top of the Pops* that reaches a national audience. And then there's the small matter of national taste . . .

Cutting Crew, with their single '(I Just) Died in your Arms' only recently deposed from the No. 1 slot on this side of the Atlantic, are three Englishmen – vocalist Nick van Eede, bass guitarist Colin Farley and drummer Martin Beedle – and a Canadian, guitarist Kevin Scott MacMichael. Virtually unrecognized in England, they represent one such band of defectors, yet they deny any deliberate policy of "playing the system," as singer van Eede explains . . .

"We aren't a band tailored to the media bandwagon, we won't play that game. And we haven't chosen to make music for the American market or the British market, we just make music."

Theirs is a brand of good old-fashioned rock 'n' roll, but with a liberal dose of

synth technology to bring it in line with the '80s. On their album, *Broadcast*, the keyboards were looked after by van Eede, guitarist MacMichael and a number of guests. But there's another angle to Cutting Crew's synthetic tones: MacMichael's Roland GR300 guitar synth, for which the mild-mannered Canadian has endless enthusiasm.

"I'm a big fan of the GR300 because it's got such an individual sound. It's got its own character. On the album I used it a lot for doubling guitar lines to put more texture in the background; I'd set up an oscillator an octave above the guitar, and then tuck it in behind. Where it really comes into its own is on tracks like 'Sahara,' which is a slow ballad. All the string washes are done on guitar synthesizer. I think they've got a movement, a swirling to them that you just wouldn't get using a keyboard. It's a combination of using the volume pedal and hand vibrato on the strings.

"Textures like that have to be lush, and that's where the guitar scores over a straight-ahead, hands-on keyboard. When a song needs something extra, when it needs more than just notes, that's when I like to use the guitar synthesizer."

Being such a fan of the GR300, it seems strange that MacMichael hasn't gone for any of the more recent developments in guitar synthesizer technology, especially those based around MIDI.

"It *is* way behind a lot of the current guitar synths," he admits, "but some people still like Farfisa organs, and I still like the GR300. Adrian Belew used to do some fantastic things using one, he used it on *Discipline*. I think that's a classic example of what you can get out of it. He's using the GR700 now. I haven't heard any of his new stuff but I found the time lag on the GR700 and the fact that it's just a glorified JX3P, which isn't my favorite synthesizer anyway, meant I'd rather sit around and see what's going to happen next.

"I want to have a look at the Stepp guitar soon. One of the things about the Stepp is that it's a *guitar* synthesizer. I'm not really interested in MIDI because I don't want eight rack-mounted DX7s with my guitar firing them all up, I want something that involves the performance aspects of the guitar. I'm not trying to be Keith Emerson with my Strat.

"One of the most integral parts of my style is the volume pedal. Sometimes it's a bit of a drag because it makes you a bit static on stage, but I don't know how anybody could do much without the volume pedal, especially with bowed-type sounds. I mean, you can fiddle about all day trying to get the attack on a sound just right, but there's nothing like being able to control it in real time with your foot. I'm a big advocate of the volume pedal."

It's reassuring to learn MacMichael hasn't neglected the more traditional side of his guitar playing, and there, too, the GR300 holds his interest.

"I fancy the Roland as a guitar as well as for the synth section. It's got some lovely Strat-type sounds in it. If you put all that through an Ibanez Multi-effects Unit – that's the 405, the one with compression, parametric, stereo chorus and analog delay – it's a very sweet guitar. I've also got one of the new Roland DEP5s, which is a great beast."

FIRST AND FOREMOST, Cutting Crew regard themselves as a "guitar band," a statement borne out by their lack of a dedicated keyboard player. Their use of synthesizers is to add color and texture to their music but, rather than restricting them, they claim this has actually broadened their outlook on synthesized sounds.

"The synths are very important," asserts van Eede. "Because they're used primarily for color, we have to use a very wide variety of sounds. Some bands are recognizable for their synth sounds alone, but we don't tend to stick with the same ones – they all have to be different colors that let the guitar weave around them."

"All the string washes are done on guitar synthesizer. I think they've got a movement, a swirling to them that you just wouldn't get using a keyboard."

MacMichael gives the guitarist's version of the story: "I've sort of reacted against modern technology, because the last band I was in was a real techno-pop band where we had everything MIDI'd up and the drummer playing to a click-track to keep in time with a LinnDrum. But I found the heart and soul of the matter started to disappear, and that's why Cutting Crew is very much a guitar-based band. The keyboards are there but we use them in a supporting role."

Alongside van Eede's and MacMichael's synth playing and Colin Farley's piano work, *Broadcast* includes Peter Woodroffe's Fairlight and contributions from Dave Le Boult (of Laurie Anderson fame) and Jethro Tull's Peter Vettese.

"Peter was a real joy to work with," recalls MacMichael. "He's a real sweetheart. We'd just set up a track in the studio and say, 'what do you think this needs, Peter?' and he'd start playing. No chart, nothing. He'd just listen through once and he'd clocked it, then he'd start throwing stuff in.

"I tend to be the manic one who'll come in and listen to a track and just stick anything down. I'm a big fan of sampling keyboards used for what I think they are – toys, very expensive toys. I'm not trying to upset all those digital fans out there, I like them for dropping in interesting little afterthoughts.

"What d'you do with toys? You have fun with them. And out of the fun comes something that you can work on. I find I need stimulation all the time, otherwise I start to veg out. I need toys around to play with to keep me happy.

"There's one track called 'Fear of Falling' that's got sampled sheep on it. It has relevance to the song because it's about being sold down the river in New York; it's a bit of a personal vindication, so I thought we'd get some sheep . . . But they're in there for all the headphone freaks if they want to find them.

"I'll sit at home with my Commodore 64 and some drum software and let it amuse me. Then something will come from there. I don't like that to form the backbone of a song, because if that's the case I don't think you've got a song to deal with. I'll look for something that I can see as four bars of an interesting idea, then, when I get a good song, maybe I'll use *that*. I'm continually putting things on file for later use."

Between van Eede's singing and MacMichael's guitar antics, responsibility for Cutting Crew's on-stage keyboard playing has had to be delegated. Until now, a series of session players has passed through the ranks, but the vacancy is about to be permanently filled.

MacMichael: "We've just hired our first full-time keyboard player for live work. His setup is pretty extensive – a Prophet, a Mirage, a bank of DXs – and with that lot he manages to recreate the stuff we had on the album. He's had it easy up 'til now – he's just had to walk in on one of the parts that Peter Vettese or myself has already done – but now I'm going to spend some more time with him working on sounds for the next record."

THE COURSE OF recording *Broadcast* was anything but smooth, leading Cutting Crew on a tour of studios in England and America, and through a succession of producers before finally sharing the production credits themselves. Both van Eede and MacMichael agree it was an instructive period, but . . .

"It was the first major project any of us had got stuck into," recalls MacMichael. "We made a lot of mistakes and ended up firing three producers in the process. There were no hard feelings, it was just that what Nick and I had in our heads wasn't getting translated onto tape, and we weren't clever enough to be able to go in and do it all ourselves. What it boiled down to was that we were so opinionated, we needed a brilliant engineer to co-produce us. He was Terry Brown. The single, 'Died in your Arms,' is actually the third recording of that song."

"Certain producers got involved at the 24-track analog stage," continues van Eede, "then we came back to England to a guy who wanted to go 48-track. That didn't work out, so we went to a guy who didn't ▶

Nick van Eede (right) and Kevin MacMichael.

mind if it was 24-track or 48-track, as long as we went digital. So you find you're overdubbing in a digital studio on a copy of stuff that's four months old, recorded on 24-track analog equipment. We ended up working with copies abroad, so the masters were all safe at home, and twice we sat in a studio in LA, put the tapes up and all that was there was a couple of sequencer patterns and a sampled snare drum. There we were, waiting to do a mix, and the producer was saying 'you guys are pretty minimal, aren't you?' "

Eventually the recording settled down to the peace of the English countryside – a peace that was quickly shattered, if Cutting Crew's after-dinner stories are to be believed. The studio there turned out to be well-suited to their needs.

MacMichael: "We worked in a really magical studio called Comforts Place in Surrey. They have every imaginable keyboard there hard-wired in – it's wonderful. You can try anything you fancy. If you want something on the Fairlight, or the Emulator, or the Kurzweil, you just have to push a button and it's through to the desk. They already had the sheep there amongst a lot of animal noise samples for the Emulator II."

Chaotic? Certainly. Educational? Apparently. Satisfactory?

"The first album portrays the songs of Cutting Crew, but the spirit has evolved a lot since then," says van Eede. "That's what we want to concentrate on capturing on the second album. We want something that's not quite so clinical, that's got a bit more air and space.

"For the last album, when I'd written, say, 'Died in your Arms,' we could sit here and strum it on an acoustic guitar. Then, when we'd take it into the studio with the other guys, we'd have to teach them their parts because they hadn't really become part of the band. Now they go out and play wonderfully because they've done two tours, but it takes that time to happen. Kevin's into computers now too, so I'm sure we'll do a lot of work at home before we go into the studio again."

According to van Eede, the band really wants to get into more spontaneity. "On tour I'd love to be saying, 'Here's a song we finished this morning – 1, 2, 3 . . .' It's like when you're playing in bars and the drummer's just written a new song; you don't know it but you're going to play it anyway. You have that feeling of, 'Oh shit, what's the next chord? I've forgotten.' And then the drummer plays the fill in the wrong place . . .

"I want to get some of that back into the band. When you get into the world of recording and hit records, you go out and play your album, then you have to wait until the next tour to play the next album, and that's terrible.

"I want Kevin to be more off the wall. He's the most melodic guitarist I've ever worked with but, at the same time, he's got a really wild streak in him. On tour, when he's had one too many German beers, it'll come out and it's great. I want to capture a bit more of that. I want to be more daring, take a few more risks.

"I think it comes down to tangents. We want the band to be looser, but we still want a lot of 1987/88 technology in there, and that will probably come from Kevin sitting in his bedroom with a computer."

In which case, Kevin Scott MacMichael will find himself in plentiful company. ■

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Yamaha TX81Z FM Synth
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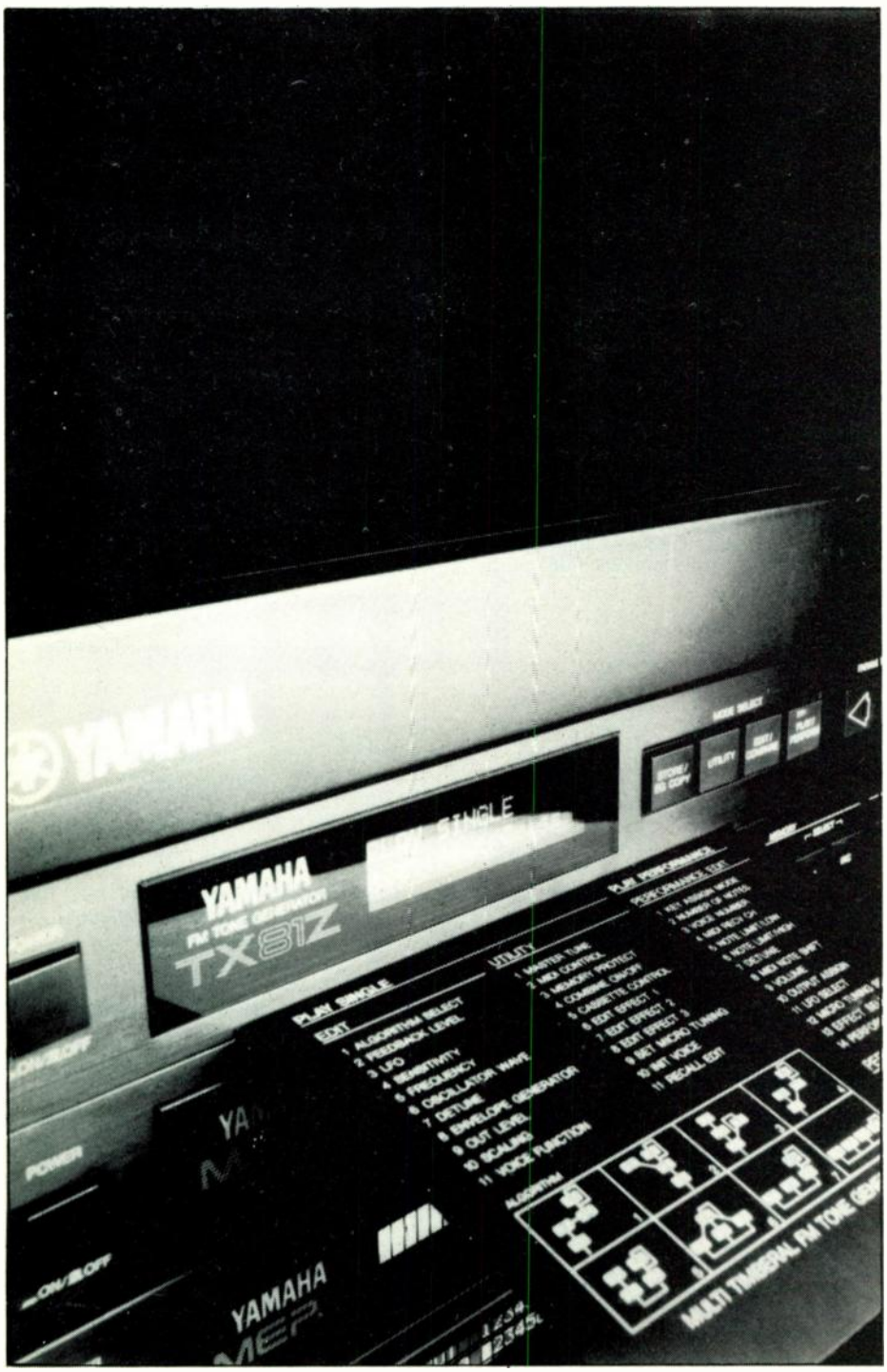
REVIEW

Yamaha's second multi-timbral synth module shows improvements in sound quality, packaging and flexibility. As if that isn't enough, the MDF1 MIDI disk drive caters to the TX81Z's storage needs as well as those of other MIDI instruments. *Review by Rick Davies.*



WHEN YAMAHA INTRODUCED the FB01, many FM enthusiasts were both very happy and very frustrated. Happy because they finally had access to an inexpensive multi-timbral sound module. Frustrated because they needed an external computer and voice-editing software to program it, and because the four-operator voice architecture didn't provide the flexibility that six-operator instruments had.

The newest tone generator from Yamaha, the TX81Z, addresses both of these concerns. It is programmable from the front panel (although the process is a bit tedious) and its four-operator architecture has been beefed up by offering a choice of different waveforms with which you can begin the synthesis process. Like the FB01, the TX81Z is an eight-note polyphonic FM synth, capable of producing eight different timbres at the same time.

The TX81Z chassis is of the single rack space variety, which alone is a significant improvement over the FB01. The back panel features the standard MIDI In, Out and Thru sockets, a DIN cassette interface, a non-detachable power cord, and two audio outputs ("I/MIX" and "II"). Headphones plug into a front panel socket, where they join a 32-character LCD and 11 switches (four for mode selection, two for parameter selection, two for data entry, two for master volume/cursor controls, and a "Cursor" switch which toggles the function of the master volume switch.

Sounds

THE FM SOUND is certainly no longer a novelty, and the way it is produced is no secret: sine waves (generated by "operators") multiply with, and add to one another to produce complex waveforms. The timbre of these waveforms depends on the number of available operators, how they are connected (determined by the voice's "algorithm"), their respective levels and tunings, and how they change each time a note is played.

Sounds simple, doesn't it? I won't try to convince you that it's easy to program, but it *is* possible to learn the system a little at a time. The real trick is in getting access to all of these parameters with minimal hassle (but more on that later).

Given that the TX81Z is a four-operator instrument, one would expect its voices to resemble those of the DX27 or DX100 more than the DX7. But in fact the TX81Z's sound is different from all of those instruments', because its operators can generate seven different non-sine waveforms. To get even a static complex waveform on any other FM synth, you would have to use at least two operators, so there is a chance that the extra waveforms

will compensate for the four-operator restriction. But do they really?

To a limited degree the answer is: "yes." Three of the eight available algorithms have three operators running in parallel, so that you can easily detune three distinct waveforms and get a thick sound, and through creative enveloping you could fade operators in and out – a poor man's additive synthesis, really.

But if you want filter-type effects, you're still going to have to use a couple of those precious operators as modulators, and though the range of timbres is greater with the new waveforms, this is no substitute for two additional operators. (Although two voices can be doubled for the equivalent of an eight-operator voice in exchange for reduced polyphony.)

A nice touch is the TX81Z's "reverb" control which is not actually done with signal processing but gives a reverb-like effect by extending envelope release stages over a variable amount of time. It's not a substitute for the real thing, but it is certainly worth using from time to time.

The TX81Z's 160 factory voices do a good job of showing off the new face of four-op FM – a cross-section of traditional DX-type sounds plus a few surprises. Brass, woodwind and various plucked and hammered instruments are well represented, as are a variety of strings which work remarkably well for a four-op instrument. (Sure enough, examination of the string voices reveals extensive use of the new waveforms.)

Now, 128 of the "Single" voices are arranged in four banks (A to D) of 32, and though they may be edited, they can only be stored in the remaining 32 user-programmable voice locations (Bank "I"). Bank I comes from the factory already loaded with voices (which are not duplicates of any of the other voices), so if you're planning on doing any editing, look for some form of voice storage right away. In this case, your only options are the cassette interface or MIDI.

When in Single Play Mode, the Parameter and Data Entry switches select the voice bank and number, and the TX81Z behaves as a simple, eight-voice synthesizer – no splits or layers. Simple, predictable, and the best place to start to get acquainted with the TX. Two TX81Z's can operate together as one 16-note polyphonic instrument by setting one TX to respond to all even notes and the other to respond to all odd notes. Thus, both TX's can be driven by the same MIDI controller, rather than having one TX's MIDI output "overflow" to the other TX's MIDI input when its eight-note capacity is depleted.

To play more than one sound at the same time, you select one of the 24 programmable "Performance" setups. These automatically select up to eight Single voices, and set them up to suit whatever controller you are using – whether that's a keyboard, guitar converter or sequencer. This is how you can achieve splits, layers, or multi-timbral sequences. Although the options are many, programming Performance programs is very easy, and mainly entails selecting up to eight Single voices in Performance Edit mode, and then assigning them to any combination of note ranges, MIDI channels, audio outputs, transpositions, tuning offsets, and volumes. To make life easy, Yamaha provides several initialized performance setups, including splits and layers, which require you only to select the voices you want to combine – the defaults take care of the rest.

Each performance setup can invoke one of 13 microtonal keyboard tunings (11 preset and two programmable) and one of three programmable "effects" – "delay," "pan," and "chord." Yamaha deserves credit for supporting alternate tunings (even if the majority of TX owners are not likely to make much use of this feature), but the TX81Z's effects are the subjects of my rantings and ravings for this month. Except for the problem of using a preset LFO to automatically pan voices across the audio outputs (which is essentially what happens if you use many of the preset voices in your Performance setups), the effects make this instrument a lot of fun. The "delay" effect is, like the "reverb" feature, not a digital delay, but uses any available voices to repeat notes, and can raise or lower the pitch with each echo. The "chord" feature lets you program a four-note chord for every note you play. This is light years beyond the old "chord latch" approach, and is a blast for single-note soloing.

Needless to say, the Performance mode is where the TX81Z really gets a chance to show off. I tried it out with two guitar-to-MIDI converters, and in both cases I found the TX81Z could adapt to anything I threw at its MIDI input; this makes it equally suitable for MIDI sequencing.

Programming

I DON'T WANT to fill this review with "what do you want for $450?" remarks, but the

TX81Z really would have benefited significantly from a data entry slider and better parameter organization. After reading the manual you get the impression that the TX81Z's designers left no stone unturned, but programming this instrument leaves some question as to how well some of the features interact.

The Single Utilities section is where the MIDI controls reside for the TX81Z's transmit and receive channels, and its response to note-on/off, program changes, pitch-bend, continuous controller, and System Exclusive messages. System Exclusive dumps are initiated from this area. The effects settings are also programmed here, though they can only be used in Performance mode. This is an example of the sometimes frustrating placement of some functions within the TX81Z's system. There are several other MIDI-related controls which are accessible only in Edit mode since they affect each voice individually rather than the whole instrument. These include the keyboard mode (poly or mono – not to be confused with the MIDI modes), and ranges for pitch-bend, breath controller and mod wheel MIDI messages.

When I was testing the pitch-bend response in Performance mode with a guitar controller, I had to enable the pitch-bend parameter in the Single Utilities area, and then make sure that all affected voices had the pitch-bend range set appropriately in the Function section. The trouble came when I tried to save preset voices after editing their pitch-bend ranges. The only place where edited voices can be stored is in the "I" bank, and those 32 locations can get used up *really* fast.

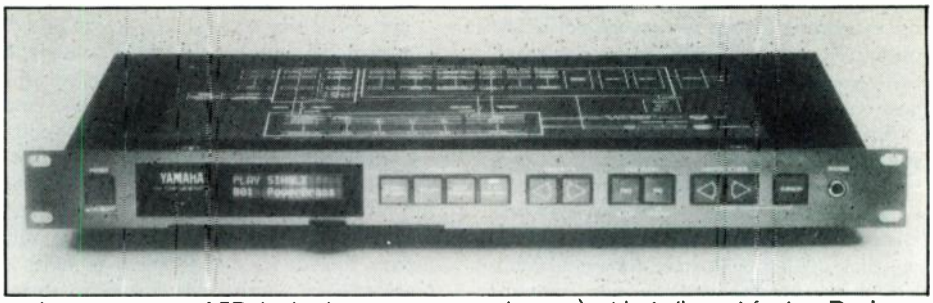
If you have a MIDI storage device such as Yamaha's MDFI (coming up in a moment), you can get around the TX81Z's limited RAM by storing banks of voices, effects, and other system data separately. It's not as fast as random access, perhaps, but it's faster than reprogramming the TX81Z each time you need new voices.

If you're going to do any serious programming, you'll want to look into software support (Bacchus Software and Poshek Productions have both announced editor programs already), but considering how few memories there are in the TX81Z, the computer could end up working overtime just as a librarian when it really could be doing other things. That's where dedicated MIDI storage devices come into the picture.

The MDFI

QUESTION: HOW DO you get excited about a disk drive? Answer: try writing a review on the same computer that you'd normally use for backing up synthesizer patches. Not everyone can afford a dedicated computer for every application, so there is obviously still room for specialized devices like the MDFI, Yamaha's first dedicated MIDI Data Filer. It is intended for the storage of patches, drum patterns, sequences, samples – or whatever the occasion calls for – on 2.8" Quick Disks. While this may not be everybody's favorite format, it did allow Yamaha to produce the MDFI for well under $400, making it worthy of a parking space in a MIDI switcher where it can reach everything in a system.

It takes a while to get used to the way that the MDFI communicates its messages with a single seven-segment LED display, but it eventually becomes clear that there's not a whole lot for it to tell you. Actually, that's fine because the less time spent with a disk drive, the more time there's left to make music.



The key to the MDFI is its simple control layout. The number of functions does not heavily outweigh the number of controls, so life stays simple. You'll want to keep a note of the files you store, however, since the MDFI doesn't supply file names or types when you prepare to load from disk, and you can save several types of data from several instruments in one file, and up to 19 files can be stored on each side of a quick disk.

There is one very nice feature on the MDFI: because it sends MIDI data back out over the channel on which it is received, you can connect it to any number of identical devices, and as long as they're on different MIDI channels, they will only accept data intended for them. This does mean that you have to look out for instruments like the TX81Z which can receive and transmit on different channels, but this feature can be very helpful in large MIDI setups. (Fortunately, the TX81Z puts both of these parameters in the same MIDI Control section of the Single Utilities, so there's no need to switch back and forth between modes.)

There are two stages to saving data to an MDFI disk. First, you select the File function, then press the Save switch, which prepares the MDFI for an incoming SysEx transfer (this must be initiated from the other instrument). Once a file is received, it is not saved to disk until you press Save a second time, which gives you the option of either sending more files to the MDFI or aborting the operation. Since you have a 19-file limit on each disk side, it is desirable to get as much mileage out of each file as possible. The MDFI facilitates this by allowing several instruments to dump programs (sequences, or whatever) onto the same disk file. When the disk file is reloaded, the instruments receive their respective data files in the same order they were recorded. This makes the MDFI particularly useful in live performance, where you might want to load fresh data into several devices for a particular track.

As I mentioned earlier, the TX81Z has a limited number of memories, but it only takes five seconds or so to reload effects settings, and seven or eight seconds to load either the I-bank with 32 voices or the Performance memories with 24 new combinations. This really does help the TX81Z out substantially, since its files are so small.

The MDFI doesn't discriminate against non-Yamaha products, either, so you can also connect effects units, drum machines (or whatever) with similar satisfaction. Don't get your hopes up if you own a sampler, though – Sample Dump Standard messages may be accepted, but 60,000 bytes (the size of the MDFI's input buffer) doesn't add up to much in today's world of 12-bit samplers; chances are you'd cause the MDFI's input buffer to overload if you tried to save anything but the tiniest of samples. (I verified this with a Prophet 2000.) To help you plan your files, Yamaha have been thoughtful enough to include a list of file dump sizes for a dozen or so of their MIDI products.

If you run out of room on a disk, just flip it over. But keep those disks protected – it's easy to get careless once you've become accustomed to 3.5" disks.

One small point worth mentioning: if you're thinking of buying one of the new DX7s and an RX5 drum machine, you'd be better off going for the DX7IID (which has no disk drive), the RX5 and the MDFI. Reason being that the DX7IIFD (which does have a disk drive) won't store data to disk sent to it by the RX5, whereas the MDFI – helpful beast that it is – will accept files from both the DX7IID and the RX5.

Conclusions

AFTER A COUPLE of weeks with the TX81Z, I decided it was indeed worth the effort to overcome my initial confusion about the parameter organization. The TX81Z is fun, but like a lot of my favorite albums, this instrument took several listenings before it started to really grow on me.

The four-operator sounds are clean and quite usable, and all the features are there to let you do just about anything you could want with them. The built-in selection of 128 preset voices acts as a good starting point, but in the end, the 32 programmable voices will probably get used up quicker than you might imagine.

The TX81Z will appeal to anyone who wants to get started with sequencing but hasn't enough money to justify a keyboard instrument, particularly if a multi-timbral FM synth sound is called for. I found that it is ideally suited for control by guitar-to-MIDI converters, and though the complexity of programming may be a put-off initially, in the long run it's well worth the time and money. ■

PRICES TX81Z $450, MDFI $350

MORE FROM Yamaha International Corporation, PO Box 6600, Buena Park, CA 90622. ✆ (714) 522-9011

Illustration: Stuart Culkeen

we can't go on

BEATING LIKE THIS

part 4

If you're going for realism in your drum programming, why restrict yourself to tapping little switches with your fingers when you could be whacking electronic drum pads with sticks?

Text by Matt Isaacson.

BY NOW, HAVING faithfully read every installment in this series, you are fully aware of the psychoacoustic subtleties of percussion instrument sounds, and why drum machines are limited in their ability to keep up with natural acoustic percussion in all of its crude mechanical glory.

You also know that some of the inherent sonic shortcomings of drum machines can be overcome by using samplers in their place as your percussion sound source, especially if the sampler is carefully chosen and properly set up (and you've picked up a good many hints in those areas as well).

You've even endured a whirlwind tour of a number of currently available MIDI percussion interface devices which can be used to drive that carefully chosen, properly set up sampler.

Read on, now, as I forge another link in the Great Chain of Electronic Percussion Being with a discussion of the gadgetry used to trigger MIDI percussion interfaces.

They Hit Pads, Don't They?

AND THERE AIN'T much to 'em, really. At their most basic, they're just these things you hit that produce a trigger pulse which, in turn, triggers a sound. So what's the big deal about them? Well, they provide a means to get around the inherent *mechanical* rather than sonic limitations of drum machines and keyboards. This point was left untouched (or perhaps taken for granted) in the last installment, where the discussion of percussion interface devices assumed that you would *of course* be interested in adopting one as your percussion-programming axe. An underlying assumption (of this entire series of articles, actually) is that you are interested in learning the techniques available for elevating electronic percussion programming to a level of rhythmic flexibility approaching that which *real* drums have always been able to achieve easily – and that you are interested in using these techniques to create percussion tracks which sound just like the real thing (at least on tape). Essential ingredients here are cohesiveness and spontaneity – the focused energy of a unified rhythmic motive, enhanced by the improvisations and minute variations that are the hallmarks of a live percussion track.

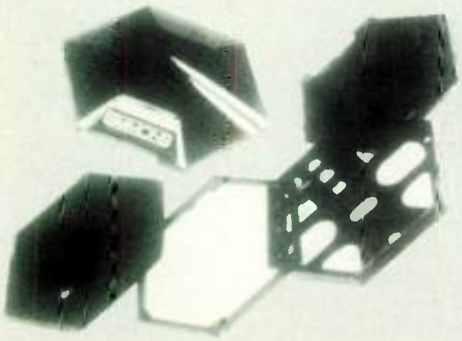
Well, I'm here to tell ya – pushing buttons on a little box is just not gonna cut it. Some of you may have read about pads and interfaces and thought, "Gee, that's fine and all, but it

doesn't help me much. I don't know the first thing about playing drums, and anyway, I can program any rhythm I can think of into my sequencer if I single-step it . . ." Sure you can – I won't argue. But those two little words – "think of" – may just end up weaving your best-laid plans into a tangled web before you get very far. Hey, life's pretty short, isn't it? Why put yourself through that kind of torture – hour upon hour of tedious, brain-taxing accountant work, just to create a five-minute track that sounds – well, *programmed*? When instead, you could be playing music?

Using a sequencer, nearly anyone can create complicated keyboard pieces even though they may have no keyboard skills. But just the smallest amount of keyboard chops will make it easier to record those sequences. The same applies to drum sequencing, now that the hardware is available to allow drumming skills to be used directly.

Before going on about pads, let's look at some other options. If you've been using one of the new crop of drum boxes with velocity-sensitive drum switches, hats off to ya. This is probably the greatest single improvement on the basic drum-box format that was possible, and at the least additional cost, although the full-tilt *samplers* which have such switches are at the top of the sampler price range. A variation on this approach, the velocity-sensitive keyboard (standard equipment on most keyboard samplers), allows similar dynamic input with the added advantage, since there are more keys than sounds, of being able to have multiple tunings of a given sound, as well as more sounds, simultaneously accessible to the player. For that matter, it is not often you will see someone playing a set of *sixty-one* pads! This is a huge step up from having to use a pitch knob, or worse, having to enter new pitches on a keypad – things which tend to kill spontaneity and make everything take five times as long. Incidentally, many drum machines which do not have velocity-sensing switches *are* able to respond to (and record) the velocity output of an external MIDI keyboard, and some of the ones with programmable pitches can do some fancy tuning tricks in conjunction with an external MIDI keyboard.

Nevertheless, these devices leave a bit to be desired when it comes to percussion playing because, quite simply, they are the wrong tool for the job. This business of playing drums with your fingertips on little keys and switches is enough to make a drummer curl up in a fetal pose. Drumming is about bashing big things with sticks, fer chrissakes!



The non-drummers among you may not be familiar with the magical rebound dance of drumsticks ricocheting off of elastic surfaces. If you have this under control (and it's not as hard as you might imagine), it lets you do things you couldn't dream of doing on a keyboard, and the very feeling of swinging sticks against something can work wonders for opening up your sense of rhythm.

You who *are* drummers know all about this, just as you know how frustrating, even futile, it is to try to translate your two-handed, two-footed rhythm grooves into the two-finger format.

The bottom line: percussion playing is a very intuitive, visceral process, and anything which constricts the physical aspects, or forces you into a lot of analytical thought to get the job done, is bound to detract from the result. You want to just *play* – and that is exactly what pads and interfaces allow you to do.

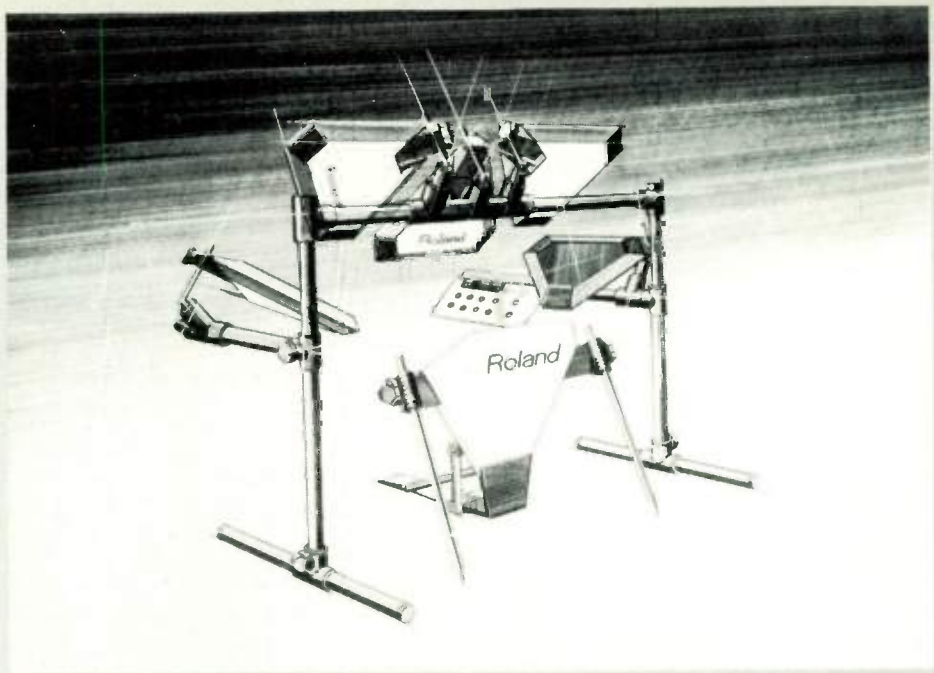
What's In A Pad?

MOST PADS CONSIST of a hard flat surface with a piezoelectric transducer – essentially a contact microphone – attached to its underside. The top side is generally covered with a sheet of rubbery material to maximize rebound and decrease the noise and hand trauma of stick impact, as well as, in some cases, acting as suspension and shock isolation for the hard surface (or batter, as it would be called on a drum).

On top of this, there's usually some type of interconnect and stand-mounting hardware. All of the above is then sealed into a rugged plastic housing in a style hopefully distinctive enough to avoid a trademark-infringement lawsuit.

All that aside, the differences in performance between the currently available brands of pads are subtle, and could easily take a back seat to such weightier considerations as price, size or the style of the mounting or inter-connect system. This is not to say that there are no differences – as with any other type of equipment, the "try before you buy" rule is a worthwhile one to follow, especially since there's no learning overhead involved with pads (they all work the same). For example, you can be fairly certain that Simmons pads will work well with a Simmons interface, Dynacord pads with a Dynacord brain, and so on. However, you may find (we did) that the Dynacord pads are a bit shy in the trigger output level department when hooked up to other converters such as the Simmons – as just one example.

Also, we found some variation in the ability of pads to reject vibration transmitted through the stand from other pads on the same stand. It *is* sort of amusing to have your crash cymbal sample trigger on snare pad accents, but the joke wears off pretty quick. Pads with floating batters, such as Yamaha and Dynacord, fared the best here. A pair of Roland PD20s worked side-by-side without a glitch, but both were triggered quite easily by strokes on a cowbell mounted above them. This has little to do with the trigger interface unit – with a single signal line from each pad, the trigger unit really has no way to distinguish soft hits from stand▶



▶ crosstalk, and at most can only be set to reject *all* input below a given threshold level – which limits the available dynamic range of the system. (A notable exception is the Roland Octapad with its integral crosstalk-elimination system, in which extra pickups are used to detect vibrations being transmitted through the frame and cancel out this component of the signal coming from each pad. However, while the internal pads do not crosstalk to one another, the Octapad is in the same boat as all the other interface units when its external trigger inputs are used.)

On a separate note, some pads had fairly even playing response at all points on the batter, while others were markedly more sensitive to dead-center hits.

Other differences are more subjective. I tend to prefer pads which give the highest amount of stick rebound and make the least amount of noise of their own. Good rebound is helpful for playing technique, while low noise makes it easier to focus on the sounds being triggered by the pads, rather than the sound of the pads themselves, which by their acoustic immediacy can trick you into a perception of heightened dynamics (there's that crude mechanical glory again) if you're not using closed-back headphones or *very* loud monitors.

The Pearl cymbal pad is an oddity which rated somewhat poorly in both rebound and noise categories. No doubt inspired by the observation that crash cymbals are played by bringing the side of the stick down on the edge of the cymbal, as opposed to using the tip of the stick, it is set up to allow the same technique, offering a down-turned semi-flexible plastic lip as the playing area. This strikes me (oops, sorry) as somewhat silly, since in contrast to real cymbals, the sound is gonna come out the same no matter *how* you hit the pad. And since it is indeed a pad which you are hitting, you might as well make the most of it – pads have much better rebound than crash cymbals and even most drums, especially those big deep ones with loose floppy heads – "floor toms", I think they're called. Besides which, it looks like a duck and *seemed* like it would break if I hit it hard enough (it didn't, though). On the other hand, it is smaller than any of the "normal" pads and better suited to being wedged into tight setups at odd angles, since you can come at it from almost any direction.

The Yamaha pad mount mechanism scores points for both ingenuity and ease of use. With one hand, you tilt and swivel the pad into almost any position, and then tighten a single knob with the other hand to hold it in place. (You are, after all, entitled to a little something extra in exchange for the higher price.) The more conventional mounting systems used by almost all of the other contenders require the individual adjustment of at least one piece of hardware for each available degree of freedom. Adjustable indexing keys ease the task of later reassembly to the exact same setup – or make it harder to change it around again.

The Roland PD20 pads are a bit smaller than the rest, which is helpful for those (like myself) who think that eight pads is just getting started, 18 is in the right ballpark, 28 is just about there – well, there *are* 32 sounds in my sampler.

There is an alternative to using drum pads altogether – namely, using actual drums outfitted with contact transducers or internal microphones. This is the sort of thing that dyed-in-the-wool *drummers* are inclined to do, especially if they go onstage a lot, and even more so if the electronic sounds are being used mainly to supplement the sounds of the acoustic set – or if they just can't stand playing on pads and don't mind the additional noise and bulk. But this poses some problems which are not encountered with pads, the most obvious one being a major tendency towards acoustical crosstalk (less of a problem for the drums themselves, although the absence of buzzing snares on floor tom hits is a dead giveaway that you are not listening to an entirely live recording).

Another less obvious problem is that the audio signal picked up from a drum is often ill-suited for deriving triggers.

A well-designed pad is a heavily-damped vibrational system, in which the piezoelectric pickup acts as a sort of mechanical high-pass filter which responds most strongly to the initial impact of the stick. The result is that the trigger signal has a very clearly-defined single peak and dies away rapidly, so that even at closed-roll playing speeds, the signal spikes created by the individual stick hits are easily distinguishable from one another. Most interface units are designed to take advantage of this – each of their inputs can be retriggered virtually as fast as MIDI can carry the note messages away.

On the other hand, drumheads tend to generate "dirty" attack transients whose real peaks may be hard to discern, especially at higher playing speeds, and the signal from a single hit may take a second or more to die away completely. The Simmons MTM devotes a hefty portion of its programmability to dealing with such inputs, using parameters such as "dynamic hold-off" and "percent above previous threshold" to bring things under control. With other interface units that are optimized for piezoelectric triggers, audio signal triggering can be expected to yield sluggish response, irregular velocity sensing, and multiple triggering.

And As If That Were Not Enough

AT LAST YOU sit with sticks in hand, gleefully swinging away at a veritable sea of pads before you – what are your feet doing? Holding up your legs? What a waste! Especially if you're used to including them in the beatmaking. Your basic kick/snare/hi-hat rhythm is just a touch beyond what can conveniently be played with two hands. Laying this rhythm down into a sequence in two passes (eg. kick and snare on one pass, hi-hat on the next) means having to do one of the following: 1) stick to a rigidly structured rhythm so that the different parts will mesh, 2) play what you feel on the first pass, then carefully memorize and analyze the results in order to come up with a second pass that fits well with it, or 3) just wing it on both passes, and settle for rhythms in which the individual parts may not work together in quite the way you had hoped (although the unpredictability *can* produce some interesting accidents, and a dose of chaos works wonders for breaking out of rhythmic ruts). It also means taking at least twice as long to record the same sequence.

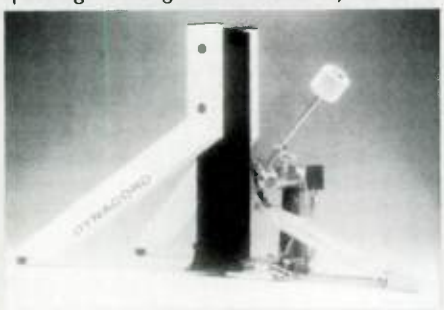
But – especially if you have already mastered the hand/foot co-ordination – how much nicer it is to be able to play the entire rhythm at once, in the organic way, with freedom to improvise, and without having to really think about it all that much. (Here's that American jazz-rock mentality cropping up pretty strongly now.)

Kick-drum pads – the obvious extension to the basic pad idea – have been around nearly as long as drum pads themselves, and are generally just much larger versions of the same thing. This is because apparently most drummers feel naked and exposed without something large and bulky to obscure their legs ▶

from view. (All right, that was unfair. The bottom-heavy look of the traditional drum set *is* quite visually appropriate to the sound it makes.) Only in the last couple of years or so has it dawned upon some manufacturers that a more compact item might be saleable as well.

The most extreme example of this is the Drum Workshop unit, which reverses the customary method of attaching a kick pedal to something else, and instead has a transducer mounted to the base of the kick pedal itself. The beater is re-oriented to swing downward towards the transducer, adding a bizarre visual touch. A low-profile metal and rubber plate provides solid support for the pedal and keeps it from sliding around, and that's it – you can't get much more minimal (although it's said that Terry Bozzio used to perform with a plain old footswitch as his kick pedal – so much for live dynamics).

Just one size up, Dynacord sells a very space-age looking unit to which any standard



kick pedal can be attached – this is more the ticket if you happen not to like the Drum Workshop pedal, or if you already have a kick pedal you like and don't want to fork out for a new one. It's just big enough to keep your foot out of sight, but weighs the better fraction of a kick drum – it feels nice and solid and you *know* it's not going anywhere. Uniquely, it also boasts an output-level selector switch for matching its output to the input characteristics of the interface unit being used.

Along similar lines, but at lower cost, the Techtonics Company has announced a product (the Techtonics 2000 electronic bass

drum) which closely resembles the Drum Workshop unit minus the pedal, allowing you to use the pedal of your choice – as long as you can set up the beater to swing downward.

Many of the available kick-drum trigger units and pads have dual connectors to make it easy to run two or more such devices into the same input on an interface unit – a simple path to the double kick drum setup.

To those who may have wondered whether a full-blown kick drum pedal is really necessary when no actual kick drum is involved, let me first say "No, it's not *really* necessary." I'll also suggest, however, that a good-quality kick pedal is to the drummer's foot what a rebounding drumstick is to the hand – namely, a free-swinging lever extension of the body which, under proper control, allows the development of greater speed, power and precision, with less work. It becomes part of a physical feedback network which can help to stimulate further rhythmic expression. Which is more than can be said of most footswitches.

Waiting For The Other Shoe

FOR WHATEVER REASON, relatively little attention has been directed towards giving the "other" foot – the hi-hat foot – its rightful position in the electronic percussion order. Perhaps it is because hi-hats, and cymbals in general, are such hugely expressive instruments whose sounds are more complex and difficult to reproduce than anything else in the percussion world (crude mechanical glory yet again). Even drummers who have completely abandoned drums in favor of pads still use real cymbals and hi-hats in their sets.

As the fidelity, bandwidth and recording time of samplers continues to increase, however, the canned versions of these sounds are becoming quite useable, if not actually respectable. Consequently, there's a great annoyance on the part of drummers who are unable to apply their accustomed playing techniques to hi-hat sounds. For example, open and closed hi-hat sounds are typically accessed by assigning two different pads to them – a very unnatural situation for those who are used to doing it all in one place.

Recently, two products have appeared which feature similar solutions to this problem – ironically, at opposite extremes of the price spectrum. The Sequential Studio 440 and the Casio DZ1 pad-to-MIDI interface unit both allow users to designate a pad whose sound routing can be temporarily changed by means of a momentary footswitch – independently of other pads, and without resorting to program changes. They also include this all-important extra detail – closing the footswitch immediately triggers the sound which is active on the pad while the switch is closed (although at a fixed user-settable velocity, since the switch is not velocity-sensitive). This means (among other possibilities) that a player can go from open hi-hat to closed hi-hat on one pad at the touch of a footswitch, and that closing the footswitch will immediately cut off the open hi-hat sound if it is ringing – a rough but workable approximation of the way real hi-hats work.

The appearance of this feature on the Casio interface unit is really the bigger news, since it can be applied to absolutely *any* drum machine with a MIDI input. (See Part 2 of this series, MT May '87, for a discussion of the features a sampler needs in order to exploit this trick.)

Meanwhile, in the Simmons camp, greater things are afoot (sorry again). The open hi-hat sound in the SDS7 drum brain is capable of variable decay time by means of a VCA whose decay rate is subject to continuous live control (ie. it can be changed while the sound is playing). What this allows you to do is have one sound cover the range from fully-open to fully-closed hi-hats and a large number of steps in between. In addition, this sound can be suddenly choked off by jamming the control value all the way up. This control can be actuated by

an assignable MIDI continuous controller, in addition to direct DC voltage control via an input on the SDS7 back panel – said DC voltage provided by the Simmons SDS7 hi-hat pedal. Trigger input #8 on the Simmons MTM interface can be programmed to accept the DC voltage input from this pedal and to convert it into appropriate MIDI controller messages – the advantage of this indirect approach being, of course, that these messages can also be recorded and played back by a MIDI sequencer.

Unfortunately, neither the SDS7 nor the hi-hat pedal were available to me – I must confess that here, I am relaying information from a Simmons manual which I have not had the chance to confirm directly (sigh). However, having long ago seen the same trick on the original SDS5, complete with its funky analog hi-hat sounds and a continuous-control pedal which was also capable of generating velocity-sensitive closed hi-hat triggers when stomped upon, I have every reason to expect that it works just fine – and this is another *big* step closer to the way real hi-hats work. There are as yet very few drum brains or samplers which provide dynamic control of envelope times via MIDI, but this is bound to change as more manufacturers see the light.

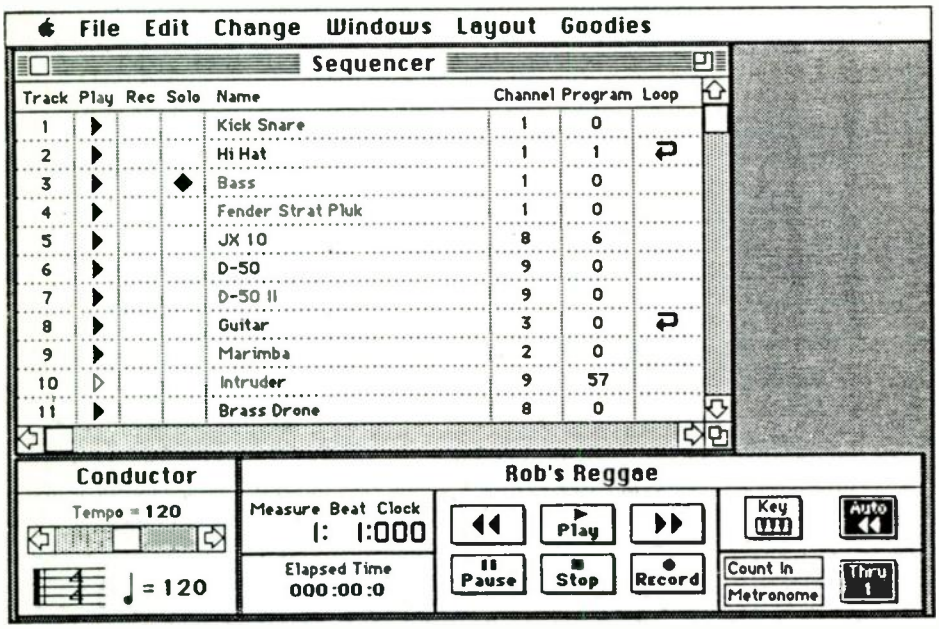
And speaking of change – a large pile of change, to be exact – Simmons has announced yet another major innovation in the player interface department, which they have dubbed "Zone Intelligence." Due to appear almost any time now on their new SDX system, this name refers to a hardware/software ensemble including special force-sensing resistor (FSR) pads, which can divine not only which pad you've hit and how hard, but also *where* the pad was hit. Specs for the finished system call for up to 16 separate zones on each pad, ideally in a concentric arrangement – which would again bring us a big step closer to the way real drums work. Among other possibilities, each zone might have a different sample mapped onto it. These could be samples of the same snare drum hit at various distances from the center of the batter (or at various distances from the microphone, for that matter). The zone system is said to extend to the SDX hi-hat pedal as well, allowing dynamic crossfading between multiple samples as the pedal pressure is changed. Tres killer, no? The big question remaining to be answered is – will the concentric zones be circular or hexagonal? Rumor has it (as always) that Roland is at work on a similar system.

With these latest innovations, we have more or less advanced beyond the point where it is really necessary to speak in terms of "programming" drum rhythms. Now, with only a moderate level of chops, you can just play them. So – sit yourself down, set your sequencer into record mode (999 bars of 64/4) and go. Wind back to the middle and listen to the stuff you were doing after a couple of minutes, when it started coming together. Pull out the good stuff, throw the rest away, and then go back and play some more. Beats working . . .

In our next installment, we'll be taking a look at percussion devices which address the needs of non-drum set playing. ■

Master Tracks Pro
Software for Apple Macintosh

REVIEW



This new sequencing program – Passport's first for the Mac – boasts some impressive new features, including a built-in System Exclusive data recorder. But does the world need another Mac sequencer? *Review by Jim Burgess.*

FEW COMPANIES HAVE been in the music software business for as long as Passport. This Californian company was one of the very first to market music software commercially.

Master Tracks Pro is their first Macintosh software product. It's a powerful sequencer with a great many unique features. And although there are several other programs available from Passport for other computers under the name Master Tracks, they bear little resemblance to this program.

Master Tracks Pro is a tape recorder-style sequencer that offers up to 64 independent tracks. It features a very respectable 240 clock-per-quarter note timing resolution, better than most of the sequencers on the market. The user interface is based around six main windows that can be moved about the screen and re-sized at will.

The Transport window provides access to all transport controls. The Sequencer window is where you interact with Master Track's 64 tracks. Tempo and meter functions are controlled from the Conductor window. The Song Editor window is used to assemble your song by moving blocks of MIDI data where you want them. All editing of individual notes within a track is carried out in the Step Editor window. Finally, the MIDI Data windows provide unique visual editing of all other types of MIDI data such as aftertouch and continuous controllers.

Recording & Playback

LAYING DOWN TRACKS is a breeze: simply select the track you want, record-enable it by clicking on the record button adjacent to the track and you're set. Like most sequencers, you can only record on one track at a time.

You'll find a complete Record Filter utility that you can use to tell the program to ignore certain types of MIDI data on input (Channel/Key pressure being favorite candidates). Those who like to quantize as they record will be happy to find a "quantize-on-input" facility here as well. MIDI Thru lets you route incoming MIDI data to echo out of the Mac on a specific MIDI channel, regardless of the channel it's being transmitted on. Anyone using a master MIDI controller or a large number of slave MIDI sound sources will choose their weapon at this point.

There are three different ways to use the transport: click the mouse on the appropriate transport control, use command keys on the Mac's keyboard, or map each of the transport functions to specific MIDI notes on your master keyboard. This way, you can utilize seldom-used notes (such as the bottom octave of an 88-key master keyboard) to run Master Track's transport controls from the comfort of your favorite playing position. Why don't all sequencers have this capability?

Choose a count-in or use the Key feature to start the transport rolling only when you play the first note. If you blew it and have to do the take again, don't worry about rewinding – Auto-Rewind zips the transport back to where it was at the beginning of the last take!

You can start playback or record from any position in the sequence simply by clicking on the measure indicator and entering the bar you want to go to. Unfortunately, there's no way to do this from the Mac keyboard itself; no big deal, but it would make zipping around your sequence even faster.

Another simple-but-brilliant feature: the Elapsed Time Counter. Almost every composer needs to know running length of their music, right? So why not tell them! Master Tracks constantly displays running elapsed time in minutes, seconds and tenths of a second.

Taming Tracks

THE SEQUENCER WINDOW is where you control the parameters that affect individual tracks. This is where you toggle on and off the playback of individual tracks and record-enable the one you're working on. The solo function lets you solo a single track; handy, but not very useful when you need to solo a group of tracks (Passport hopes to have multiple track soloing by the time you read this). You can name the tracks by clicking on them and typing away into the dialog box that pops up.

The Sequencer window is also used to specify a specific playback channel for each track. Master Tracks supports multiple-channel tracks: each individual track can hold up to 16 separate sub-tracks of MIDI data, one per channel. If you want to use this capability, set the track's playback channel to zero and it will retain each separate sub-track on the channel it was transmitted on. Otherwise, changing the playback channel to 1-16 will output all data on that track on that one specific channel.

Format *"Master Tracks supports multiple-channel tracks: each individual track can hold up to 16 separate sub-tracks of MIDI data, one per channel."*

Unfortunately, there is no provision for sending a single track to a specific combination of MIDI channels, a feature that's handy when you're layering two different instruments in one section of the song but need them to play separately in another.

You can set a specific program change to be sent at the beginning of your sequence on every track. Of course, Master Tracks supports multiple program changes during the course

of a sequence, but this first one is displayed clearly in the Sequencer window so you can see it at a glance.

You can loop tracks independently if you like. The loop point is always at the end of the last measure that contains MIDI data on that track, so looping is always "rounded off" to an even measure. The looping function won't work if you start playback from a spot that's later than the length of the track you're looping; in other words, a four-bar looped track won't play if you start from bar five.

Operation *"You can use the Edit Note function to provide access to the exact start time, duration, velocity and channel assignment of each note event."*

Editing the Globe

HERE'S WHERE THE real power of Master Tracks becomes obvious: visual editing. Of everything.

Master Tracks offers a comprehensive range of powerful global editing commands. Naturally you'll find the ones you expect to find under the Edit menu: Cut, Copy, Paste and Clear. Mix Data is just like Paste, except that rather than replacing data, Mix merges new data with the existing data that's already on that track. Master Tracks also lets you Insert empty measures anywhere within your sequence.

But that's only the beginning; the Change menu contains ten highly useful global editing commands which can affect any specific part of your sequence data.

You can change the Channel assignment of any selected region in one move. Note durations may be set to a specific value or adjusted by a percentage to make all the notes in the selected region shorter or longer proportionately.

Velocities (both on and off) may be set to a specific value or changed to a percentage of the current value. You can add to all velocity values by a specific amount or change smoothly from one value to another within the selected region. Continuous data such as controllers, pitch-bend and channel pressure can be changed in the same manner. It's also possible to re-map one type of controller change to another (for example, change breath controller data into modulation wheel data for instruments that don't recognize breath controllers).

The Change Conductor command makes it easy to incorporate tempo and meter changes throughout your composition. You can generate smooth tempo interpolations for perfect ritards.

"Strip data" is used to remove unwanted MIDI data from a selected region: just what you need to get rid of that aftertouch you didn't really want to record anyway. You can also strip out specific note ranges, a handy feature for separating drum parts into individual tracks and the like.

The Quantize command lets you select a note value to quantize to (including triplets). There is no provision for quantizing release times. The quantization grid can be offset by a specific number of clocks to put the notes ahead or behind the beat. Besides being great for changing "the feel," this feature provides a very quick way of compensating for MIDI delay in your synths/samplers.

You can even "humanize" your track by randomizing start times, durations and velocities by a specific amount.

And perhaps the best for last: Fit Time. It's a bonus to anyone who composes music for video or film. This simple feature lets you "squeeze" or "expand" a given section of music to a specified length, accurate to a tenth of a second. Just enter the desired duration for the selected section and the program does the rest.

The Selection

ALL OF THESE wonderful features may be used to affect only a specific region of MIDI data; that is, only on certain tracks and only in certain measures. Master Tracks uses three different types of windows to let you select exactly what parts of your composition are to be edited.

The Song Editor window is a general window that's used to build up the arrangement of your composition by moving complete blocks of MIDI data around.

Rows of tiny rectangles for each of the 64 tracks are displayed in this window, each block representing a measure of its respective track. ▶

REVIEW

► A solid block indicates that the track contains MIDI data in that measure, an empty block showing no data present. The track numbers appear on the left vertical axis of the window, while measure numbers are displayed by the measure ruler running along the top of the window horizontally.

Markers are provided to "map out" the various segments of your composition and make it easy to tell where you are. Setting up markers is almost identical to setting tabs on MacWrite. You can name the markers and move them around at will. Autolocating to any marker point is a quick process accomplished by using the Tab key on the Mac keyboard.

The Song Editor can also scroll along as your music plays, so you can see where you're at as the composition progresses.

The main purpose of the Song Editor is for building or changing the arrangement of your composition. Select a group of measures on a single track or multiple tracks simply by dragging over the desired area. Entire groups of tracks may be selected by dragging down the very left-hand side of the window, over the track numbers themselves. Likewise, a group of measures may be selected on all tracks by dragging across the border at the top of the window. Or, you can select specific tracks over specific measures if they all fall within the rectangle the mouse puts around them.

There's the catch, though. At present, there's no way to select non-adjacent tracks within a specific group of measures. For example, to copy tracks 1, 5, and 13 of an eight-bar chorus section you'll need to copy every track and go back and delete the ones you don't want. It still gets you to the same place, but it's a shame to have to go through two steps when one could have done the trick. Perhaps the next version . . . ?

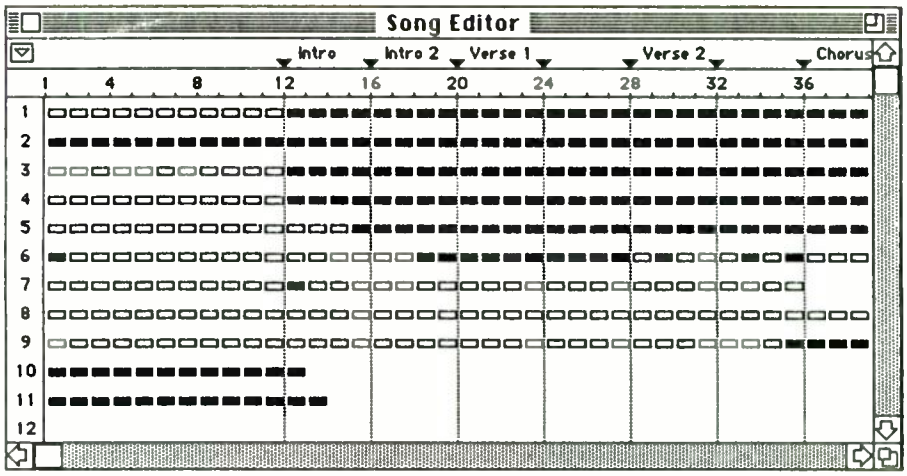
Any edits performed in the Song Editor window affect all types of MIDI data contained within the selected area; in other words, you can't copy just the notes but not the pitch-bend of a specific track.

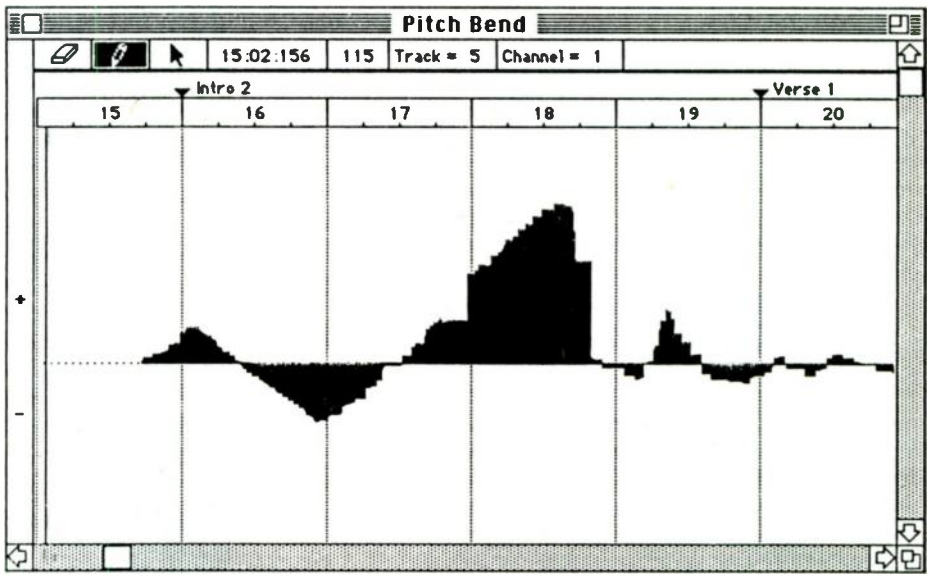
To put your music under the microscope, use the Step Editor or the MIDI Data windows.

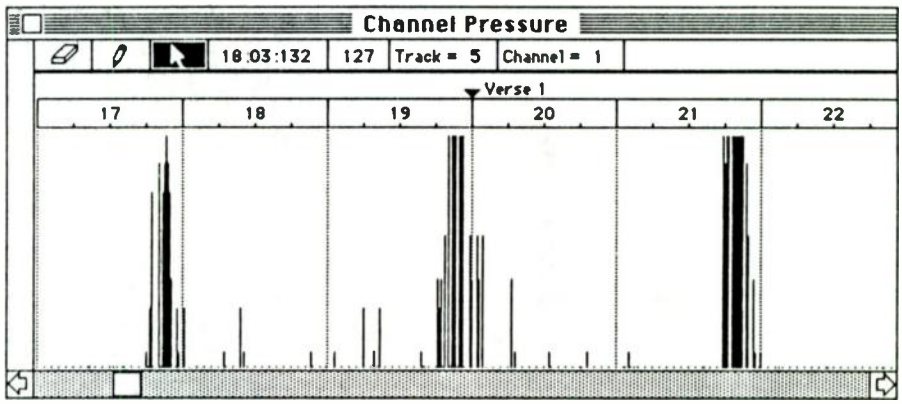
The Step Editor is basically a variation of the same sort of editing grid that many existing sequencers use: pitch on the vertical axis, time on the horizontal one. It displays the note data of a single track at a time. Individual note events appear as rectangular boxes whose length relate to the duration of the note.

To make it easy to tell where you are, a vertical keyboard is displayed on the left side of the window. You can tell where you're at by aligning with the vertical keyboard or watching a pitch indicator box which follows the position of the cursor.

Measure numbers and markers are displayed at the top of the window. To offer the most flexibility, Master Tracks lets you zoom in or out to six different degrees of magnification. The zoom setting also determines the accuracy of any editing performed in the Step Editor: at minimum zoom level each pixel represents 24 clocks (a dotted 64th note), at maximum zoom level each pixel represents a single clock (240 clocks per quarter note).







Once you're zoomed to the correct degree, you can edit notes individually or by region. Selecting a region in the Step Editor is similar to the Song Editor: simply drag the mouse over the area you want to select. The only difference is that you can't select areas from the vertical (pitch) axis, so all pitches will be included in any selected area. Now go ahead and choose your edit command.

You can also edit notes individually. Insert new notes either by clicking on a note icon of the appropriate duration at the top of the window, using a "pencil" tool to draw the notes in on the screen, or defining the desired pitch

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from your MIDI keyboard. Notes can be moved or copied freely with the mouse, and an eraser tool is available for deleting notes.

If you still need more control, you can use the Edit Note function to provide access to the exact start time, duration, velocity and channel assignment of each note event.

Features *"Master Tracks has a built-in System Exclusive storage capability that you can use to save patch data for your song along with the sequence file."*

Perhaps some of the most remarkable features of Master Tracks are its six MIDI Data windows. They provide visual editing of most other types of MIDI data other than the notes themselves. Continuous data is displayed on a graph with the data values on the left vertical axis and time (depicted by the measure numbers) on the top of the window. Pitch Bend, Channel Pressure, Key Pressure, Modulation, Controllers and Program Changes all have their own windows.

Like the Step Editor Window, these MIDI Data windows affect only one track at a time. You can zoom in and out and use markers to locate to specific points within the track.

You can select a region to be affected by the global editing commands in the Change menu in a manner similar to the previous two windows.

Somehow, though, it seems so much more pleasing to draw in controller changes with the mouse. Insert, change or delete data graphically: there's no doubt that this is one of Master Track's most powerful features. Just imagine being able to use the mouse to get a pitch-bend just right or to draw in the perfect fade at the end of your tune.

SysEx Too!

TO TOP IT all off, Master Tracks has a built-in System Exclusive storage capability that you can use to save the patch data for your song along with the sequence file. That's a life-saver for those of us who have a hard time keeping track of what sounds we used where.

This particular SysEx utility is a cut above most. It's capable of receiving multiple System Exclusive dumps from different synths one after the other and storing it as a single file. This means that at the end of the session you tell Master Tracks to receive and walk around the room to each MIDI device, making them spill their guts one after the other. Master Tracks keeps it all straight by means of the manufacturer's SysEx ID code, and will even count the number of incoming dumps and display the manufacturer's name!

When you work on the piece again, all you have to do is set the devices to receive SysEx data and send one single file out to all of them. Each MIDI device will tune in only to what it's supposed to hear and your composition will come back sounding exactly as it did when you last worked on it.

Conclusions

EVEN IN A world dominated by sequencers, this one stands out from the rest. Many will recognize certain similarities that this program has to another very popular Mac sequencer; then again, there's a lot of unique features here that haven't come together in any one program until now.

According to Passport, the latest revision of the program will be available by the time you read this. Amongst the various enhancements will be the capability to send and receive MIDI Files, the new standard file format that allows sequencers to talk to each other directly.

If you're already using a Mac for creating your music, this program is worth checking out. It may suit your needs better than what you're using now, and with the forthcoming MIDI File capability, you'll be able to use it alongside your present sequencer and take advantage of Master Track's unique features, without totally abandoning the sequencer you're already comfortable with.

If you're just getting in to using the Macintosh as a musical tool and you're looking to choose a sequencer, Master Tracks definitely qualifies amongst the very best of what's currently available.

And if you're not using a Mac for music at all, it's time you were! ■

PRICE $299.95.
MORE FROM Passport Designs, 625 Miramontes St., Half Moon Bay, CA 94019. ✆ (415) 726-0280

REVIEW

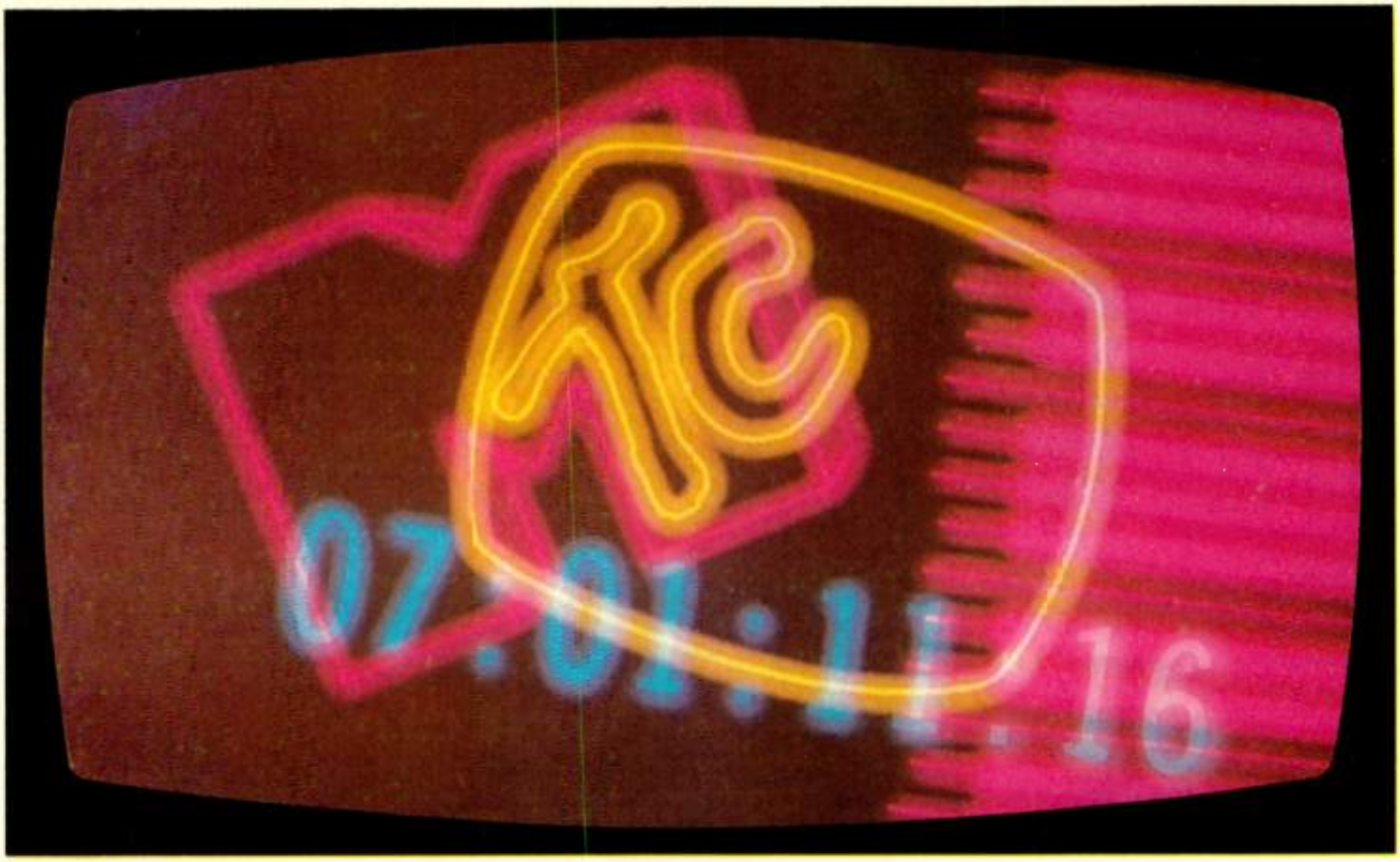


Illustration Stuart Catterson

M·I·D·I time code

MTC is here, and its applications hit closer to home than you might expect. What makes it tick? *Text by Chris Meyer.*

WE FIRST MENTIONED MIDI Time Code (formerly called "MSMPTE") in our inaugural issue back in June '86. This extension to the MIDI 1.0 specification was actually adopted by the MMA (MIDI Manufacturers' Association) and the JMSC (Japanese MIDI Standards Committee) at the summer NAMM show in Chicago that very month. It was discussed for almost an entire hour, and then Gerry Lester of Adams-Smith (maker of audio and video tape machine synchronizers) proposed a righteously better way of doing half the specification, and the arduous approval process started all over again.

The reworked specification was finally reapproved in February, 1987. This year's summer NAMM show, however, is where we will start seeing the first applications of MIDI Time Code. With this in mind, now is the best time possible to discuss exactly what this new area of the MIDI spec is all about, and what some of its potential applications are.

The Nuts & Bolts

AS HAS BECOME obvious by now, it is not necessary to know all of the bits and bytes of MIDI to be able to use it successfully. However, there are those who are insatiably curious about the details anyway, and it is those people we'll quickly oblige. (For those who don't need to know, we'll try to make this as painless as possible.)

As the full-length title implies, there are two parts to the MIDI Time Code and Cueing Specification – sending timecode over MIDI, and sending a "cue list" via "Set-Up" messages over MIDI.

SMPTE Time Code is the standard language of synchronization in any video studio, many film studios, in audio studios where synchronization between tape decks or console automation is taking place, and in any cross-breeding between these situations. Given such a weighty task, SMPTE Time Code itself is very simple – all it conveys is what time it is in hours, minutes, seconds, and some fraction of seconds (from 24ths to 30ths – referred to as "frames"), along with eight additional characters for whatever information a user prefers to include (typically, the session date or tape reel number). There is a SMPTE Time Code message for every frame, consisting of the above information plus a couple of notes concerning color frame, parity, and special conditions. There are also 16 bits to signify the end of a frame, making the overall total (in the most common case) 80 bits. SMPTE devices all work off of this common "time of day" to stay in sync with each other, and then do whatever local conversions are needed to get their job done.

In contrast to its simplicity of content, the actual reception and decoding of SMPTE Time Code is not easy. It is either imbedded in the video signal (no mean feat to decode), or comes as a series of binary 1s and 0s occurring as often as every 417 microseconds. On top of all of this, Time Code readers have to be both deadly accurate and tolerant of faults (such as tape dropout) to work properly. This (along with the fact that the demand has surged only recently) is why, until now, SMPTE Time Code readers for music and semi-pro applications have tended to cost around $1000 (and

readers for professional applications still cost a few thousand). Needless to say, adding reliable compatibility with SMPTE Time Code to normal musical instruments has *not*, in the past, been one of your cheaper propositions.

Conversely, MIDI is already implemented on the majority of musical instruments (and many semi-pro effects devices). With the hardware cost and the knowledge of how to decode MIDI already in place, it would be nice if receiving SMPTE Time Code became just a software proposition.

MIDI Time Code

THE "TIME CODE" portion of the MIDI Time Code spec makes this possible. It consists of three types of messages: one that transmits just the User Bits (those eight characters of additional information mentioned above – they usually stay the same throughout a reel of tape, or at least throughout a song or video take, and therefore only need to be transmitted once in a while); a Long Form message (a Universal System Exclusive code which, in one message, brings devices online up to the current time where the system is parked); and a series of eight two-byte messages that convey the time while the system is running. Because the first byte of each two-byte message is always a hex F1 (a previously unused System Common message in the MIDI 1.0 spec), these eight messages are usually referred to as the "F1 messages."

These eight F1 messages each carry a piece of the current time – two messages for frames,

two for seconds, two for minutes, and two for hours (with one of the hour messages also telling how many frames there are in a second). Four of these messages are sent per frame, equally spaced throughout the frame. Thus, it takes two whole frames to tell what time it is. (Of equal importance as the information inside the messages is *when* the messages arrive; they mark quarter-frame boundaries.) A happy consequence of this implementation is that very little of MIDI's potential bandwidth (that is, its total capacity for data) is taken up (at the fastest frame rate, 7.7%), while time is updated very often (four times per film frame or twice per video field – roughly every 8-10msec). Once a receiver reads eight messages and has assembled a valid time, it can keep counting internally and just use the new messages to track fluctuations in the tape speed and the like. If need be, the receiver can also interpolate finer times between the messages to accurately keep up and place such things as tempo clocks.

M I D I

"What MIDI Time Code and Cueing will probably become best-known for is consumating the impending marriage of the 'MI' and semi-pro audio markets with the high-end pro audio and video fields."

Set-Up Messages

THE SET-UP MESSAGES are a series of Universal System Exclusive messages that define when (in SMPTE or MIDI) a particular event is supposed to occur. These "events" might include when an action (such as a sound effect) is supposed to start or stop, when to punch in or out on a particular track on a sequencer or tape deck, or when an event (such as a reverb program-change) is supposed to occur. They also provide the ability to remotely edit this information in a device by deleting events and so on. Each Set-Up message may have additional information encoded within it in the form of a System Exclusive message for the specific device.

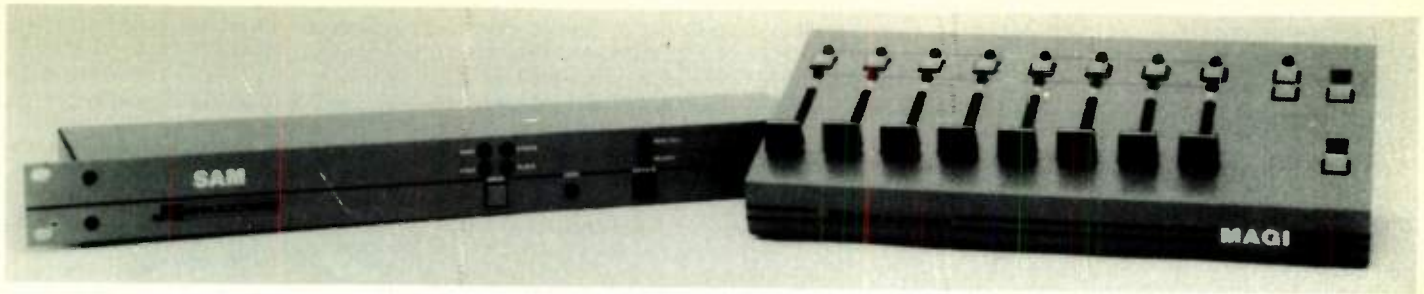
Special Set-Up messages include notes to request a cue list from a device, disable performance of the cue list, offset the time that a device is supposed to react (two tape decks that are supposed to synchronize may have different times actually printed on their tapes), and to tell when the whole system is supposed to stop whatever it is doing. Up to 127 devices may be individually addressed (or "spoken to") and, therefore, set up. Up to 16,000 different events (tape tracks, sound effects, and so on) may be addressed per device.

The idea behind the Set-Up messages is to be able to create a cue list which consists of actions that are supposed to occur at specific times. This list is then broken down into a series of Set-Up messages (one per cue, or event) which are transmitted ahead of time to the receiving devices. Then, when these devices see the MIDI Time Code or SMPTE time that matches the time of a cue, it performs that event. (There is still room in the specification for over 100 more "general" and 16,000 "special" types of Set-Up messages, so there is quite a bit of room for expandability in the future.)

Time Code vs. Clocking

AT THIS EARLY stage, MIDI Time Code is already being confused with music sequencing via MIDI and MIDI clocks. This is a tempting comparison to make; both use MIDI, and both are about "events" happening in a specific order at specific times provided over MIDI. However, carrying this comparison much beyond this point only obscures what the applications of MTC are, and therefore causes confusion.

"Normal" MIDI concerns itself mostly with musical events. A typical MIDI event tends to translate directly to a musical gesture – such as a note-on. A MIDI Time Code event has a much broader meaning – it can mean starting a whole sequence, telling a tape recorder to punch in a certain track at a certain time, changing a digital reverb's delay parameter, or whatever. Whereas musical events are timed according to what bar, beat, and clock they fall on, MTC events are usually referenced to SMPTE Time Code – at what frame of a picture a sound effect is supposed to fire, at what timecode a tape deck punches in and out, and



so on. It is best to think of MIDI Time Code as "sequencing" all the other motions that go on in a studio aside from the music itself. Keeping this in mind will hopefully cut down on the confusion.

Typically, there are far fewer MTC events than MIDI musical events occurring in a piece of music or scene of film (one rarely changes reverb programs as often as one strikes a snare drum, for example). Despite this, it is good to keep their datastreams separate in a system so as to keep them from competing for precious MIDI bandwidth. This is not too much of a hassle, since the two classes of information tend to be aimed at two different sets of equipment (eg. drum machines vs. digital reverbs – though there are crossovers). Perhaps the best aspect of MIDI Time Code (and its closest analogy to MIDI music sequencing) lies in its ability to bring the co-ordination of many different devices in a studio up on one computer screen.

But we're starting to get ahead of ourselves – let's start talking about specific applications.

Current Applications

THANKS PRIMARILY TO Fostex, more and more home studios are using SMPTE Time Code. As mentioned above, getting our MIDI'd sequencers, drum machines, and so on to sync up with SMPTE has meant spending about a grand on a SMPTE-to-MIDI clock converter such as the Friend Chip SRC, Garfield MasterBeat, Roland SBX80, or Fostex 4050. Recently, interface boxes for computers such as the Atari ST (such as Hybrid Arts' SyncBox and Steinberg Research's SMP24) and the Apple Macintosh (Southworth Music Systems' Jam Box) have started to include SMPTE readers at about a half to a third of this cost.

Amazingly, SMPTE-to-MIDI Time Code converters have already appeared for as little as $200 (the JL Cooper PPSI). I expect we'll see sequencers (hardware and software), drum machines, and the like starting to incorporate the ability to receive MIDI Time Code directly. This will give them SMPTE capabilities without any additional hardware cost (just a new PROM or system disk). Systems built around these products will only require a single SMPTE-to-MIDI Time Code converter. Considering that the aforementioned SMPTE-compatible computer interfaces are also converting SMPTE to MIDI Time Code for retransmission, there will be no shortage of such converters.

One of the new hot topics these days is semi-pro automated mixing. Currently available devices use MIDI messages to represent fader levels and other such information, and then send it out to be stored along with normal performance data in your system's MIDI sequencer. One such unit – JL Cooper's MAGI – has a built-in automation sequencer that sends and receives MIDI Time Code. There is also software on the horizon (Digidesign's Q-Sheet) that will sequence and provide moving fader representations synchronized to MIDI Time Code for mixers such as the Yamaha DMP7 and the Twister.

Stepping out of the strictly "musical" realm ▶

▶ for a moment, audio-for-video is perhaps the field most eager for MIDI Time Code. Sound effects for film and video have typically all been performed live against the visual image, or triggered from a library of sound effects on cart machines (a variation of the old eight-track players). The creation of these effects is often referred to as "Foley," with the performers being called "Foley walkers" (one of the most common needs is to recreate the sound of somebody walking).

M I D I

But nowadays, more and more studios are storing their sound effects on disks for sampling keyboards (the now-venerable E-mu EII being the most common one) and are using those where the subtleties of a live performance are not needed.

This new "electronic Foley" person lays in sound effects either by playing the keyboard in real time against the video image, or by recording them as events in a sequencer which is then synced via SMPTE to the video. This last method has been a tedious process in the past, since a lot of calculations are needed to convert tempo, bars, and beats accurately to the video or film frame numbers. More often than not, a sequence is created for each sound effect that has to be triggered, with the sound being on the downbeat and the sequencer's start time being set to that of the desired video frame.

Now, "MIDI Time Code Sequencers" can be created which allow events to be logged to occur at certain MIDI/SMPTE times instead of bars and beats. The appropriate MIDI events (such as a note-on) are then sent when the appropriate times are received, allowing whole strings of sound effects to be laid in against film or video quickly and with great precision. Some cutting-edge studios are already re-editing dialog on samplers. Someday, they will also be using these tools to re-sync words to picture.

Future Applications

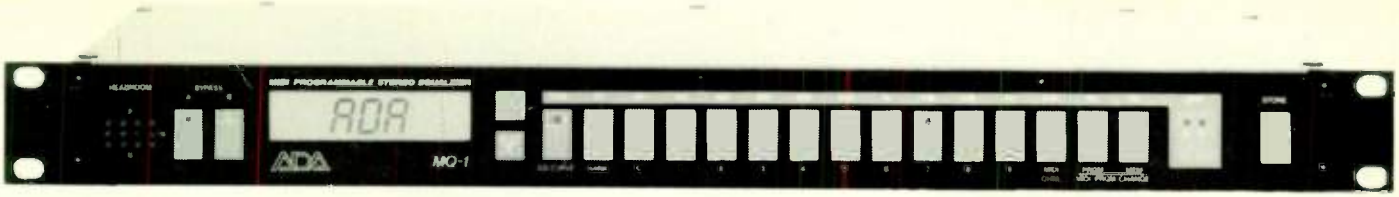
ALL OF THE above applications are being achieved with just the timecode portion of the MIDI Time Code specification. Life in the MIDI world will move on to a new level when the Set-Up messages start to be applied.

There are a couple of reasons why Set-Up messages are more attractive than the normal, existing MIDI messages for cueing and triggering events. First, some of the applications mentioned, such as automated mixdowns, require a great deal of normal MIDI information to be performed. Those who have complained that MIDI is too slow will really start to scream once they try to move 16 faders at once via MIDI. Having the information of the moves sent to or stored in the target device ahead of time, and then merely sending "FI" messages during real time, suddenly makes such feats much more feasible.

Second, more details about an event can be encoded into a Set-Up message than can be conveyed using normal MIDI performance messages. For example, Sequential's Studio 440 is capable of firing a sample in response to a MIDI note-on message, as are most samplers. However, the 440 also has the ability to receive and execute a cue list that contains such information as sound number, pitch, panning, velocity, and overall level per event, as opposed to merely saying "play a middle C at this velocity – whatever that happens to mean to you." Effects devices can have programs downloaded to them along with the time they are supposed to change, as opposed to just responding to MIDI program changes. In general, Set-Up messages (and devices that implement them – both MIDI Time Code sequencers and the targets themselves) will allow for even more precision and more portability between different studios than normal "dynamic" MIDI.

Lastly, and most importantly, Set-Up messages help to distribute intelligence around a system. A particular device best knows how it is going to react to a given action, and therefore is in the best position to prepare for it. A sampler (or a CD player, or even a cart machine) that knows it has a certain amount of delay could internally offset SMPTE times backwards so that it actually fires the sound on time. Samplers with hard disks may even be able to download the appropriate files ahead of time, once given this list of the samples they will be using. A tape deck could internally offset current ramp-up and ramp-down times for its record bias, to make punch ins and outs tighter. A MIDI sequencer could set up its record buffers ahead of time if it knows that a certain track is going to be entering record.

In general, knowing the future makes



planning for it a lot easier, and receiving a cue list of Set-Up messages ahead of time gives machines this power.

So, where will this all lead? With MIDI Time Code, MIDI seems to be perched on the verge of becoming a studio's auxiliary automation bus. For those who can't afford an SSL Total Recall system (or own devices that don't respond to it), it will become possible to automate, at the very least, effects devices (ADA already has a programmable graphic equalizer that responds to MIDI Time Code and Set-Up messages, with promises of more devices to come) and MIDI sequencers, with the possibility of continuing up through tape decks and recording consoles.

All of this automation power can come from one "black box" that displays, edits, and manages cue lists for each device in a system and tells them all when to do their thing. The hardware cost of adding these capabilities is minimal, and the software is nowhere near outrageous. And given the level of sophistication that MIDI music sequencers have already achieved, equally friendly and powerful MIDI Time Code sequencers are certainly not a dream.

But So What?

"FINE" YOU SAY. "But I've been surviving this long without MIDI Time Code; therefore, I obviously don't *need* MIDI Time Code to live." Which is true enough. Yet it does offer some amazing possibilities . . .

Just as MIDI Time Code is considered an "extension" to the MIDI 1.0 spec, so it also presents extensions to the possibilities of an automated set-up. No one ever *needed* digital Foley, but it certainly has made laying in sound effects tightly against film or video a lot easier, and made some new tricks possible as well. No one ever *needed* automated mixdown (let alone total automation of tape decks, effects devices, and sequencers to boot), but again, it sure has made some tricks a lot more precise, a lot easier, and in some cases, just plain possible.

Your "need" for MIDI Time Code capabilities should be viewed in the same light – if you intend to start operating on this new, higher level, then you should start looking for manufacturers who are going to implement and provide this power.

What MIDI Time Code and Cueing will probably become best known for is consumating the impending marriage of the "MI" ("Musical Instrument") and semi-pro audio markets with the high-end pro audio and video fields. Semi-pros will have the power which, until recently, was reserved for those who could afford to see a couple of extra zeros on the end of the price tag, while pros will get to integrate some of the neat toys from the MI and semi-pro market without banging their heads against a wall.

Democracy at work. ■

(Although Chris Meyer is known as the father of MIDI Time Code, he would like to extend credit to those who also contributed significantly to the specification and its adoption: Evan Brooks of Digidesign, Jim Cooper of JL Cooper Electronics, Anne Graham of After Science, and Gerry Lester of Adams-Smith. For a copy of the MIDI Time Code and Cueing Specification itself, contact the International MIDI Association, 12543 Hortense St., Studio City, CA 91604. The spec may also be downloaded, along with the text of a conference on the subject, from the Synthesizer and MIDI forum on PAN.)

Photography Tim Goodyer

ROLAND D50

Linear Arithmetic Synthesizer

Precisely one month after we compiled our first report on Roland's new synth, the review model has done still more to convince us of its worthiness. The details are even better than the first impressions. *Review by Simon Trask.*

IF YOU READ the first part of our D50 review in last month's MT, you'll be aware that Roland's latest synth offers a new angle on synthesis. Not only does it employ a new method of sound-creation – known as Linear Arithmetic synthesis – it also allows you to combine synthesized sounds with sampled ones. But unlike Korg's DSS1 (which at first glance is the obvious comparison point), the D50 doesn't involve itself in sampling. And while Korg's instrument uses samples as the sonic basis for synthesis, on the D50 the samples are attack segments of sounds which can be *combined* with synthesized sounds whose attack depends on the relevant envelope setting.

This is a sound principle (if you see what I mean), because the attack portion of a sound conveys a lot of information about what the instrument is – as the D50's 16-bit samples well illustrate.

The best way to understand this is to listen to a sound such as Jete Strings (internal patch 54) without and then with the samples tacked on the front end. Now, you may think that combining two different ways of (re)creating sounds in the way that the D50 does may not

produce a very integrated result. But while this is true for some of the D50's factory sounds, they are the exception rather than the rule. Most of the sample/synth combinations which come with the instrument are well integrated, "whole" sounds. There again, there's no reason not to experiment with odd combinations.

As we said last month, the D50 has 100 samples onboard. That's a high number, even when they are only attack segments (ie. short, and therefore using relatively little memory). And bear in mind that unlike sampling instruments, the D50 stores its samples in internal memory at all times – you don't need to go through the laborious process of loading them in each time you switch the machine on. Still, it does seem a pity that there's no way of loading in further samples (from a ROM card library, for instance).

Programming

BUT WHAT ABOUT this new type of synthesis? When Yamaha introduced FM synthesis to the world, they also introduced a new programming system (and let's not go into that). But while Roland's LA synthesis may be new, that newness has to do with what goes on *inside* the instrument, and hence with the quality of its sounds. When it comes to the way in which LA synthesis is presented as a programming system, there's much that is familiar.

Essentially the D50 follows the time-honored waveform, filter, amplifier format, with three LFOs for each tone that can be applied to those components. The WG (waveform generator) section allows you to select a square or sawtooth waveform, the former additionally having a pulse width setting. Filter and amplitude envelopes are both five-stage. The filter envelope allows you to vary the filter cutoff dynamically, while you can also bring in resonance (ie. boost around the cutoff point).

The LFOs offer a choice of triangle, sawtooth, square or random waveforms as the modulation source. You can set mod rate and delay time together with sync on/off. Pulse width, cutoff frequency and level can each be modulated by any one of the three LFO's while pitch can be modulated by LFO1. The depth of modulation can be set, and can also be controlled dynamically from the keyboard. This aspect of the D50 should not be underestimated; it adds greatly to the flexibility of the instrument.

There are some points to bear in mind when it comes to using the samples. For a start, not all combinations of samples and synth sounds will be in tune – so you have to indulge in a spot of fine-tuning (an easy process on the D50) to remedy the situation.

What's more, the D50's sampled sounds aren't automatically multi-sampled across the keyboard. Now, while you probably won't miss a multi-sampled triangle waveform, there's no doubt that other D50 samples do lend themselves to this process. Roland has provided samples of some instruments (for instance piano, flute and organ) at different pitches – which allows you to set up multi-samples for yourself. But how can you prevent a sample from spreading all the way down the ▶

▶keyboard? And how can you cover up the fact that a low-pitched, one-shot sample becomes very "plinky," or perhaps even disappears, in the upper region of the keyboard?

Luckily, Roland has thought of this: you can set an amplitude bias point and bias range for each partial within a patch. Essentially, this allows you to "fade out" a partial at any point on the keyboard, either above or below the bias point, while the range can be anything from two or three notes to several octaves. This can be very useful for cross-fading one sound into another, and needn't be applied only to samples.

Many of the D50's string patches offer examples of this technique. Jete Strings has a very woody double-bass bowed attack in the lower range of the keyboard which gradually gives way to a violin scrape as you move up. This sounds odd in isolation, but is splendidly effective when combined with the synthesized string sounds.

An interesting aspect of the D50's angle on synthesis and sampling is the way a sampled attack can affect the way in which you hear the sustained (ie. synthesized) portion of a sound. A good example of this is the flute sound of the flute/piano duo (patch 23). This has a flute chiff sample on the front end which immediately tells you what the sound is. However, take this off and the sustained synth sound begins to sound more like a muted horn (or trumpet when you get up into the top keyboard octave). Interesting.

Preset Sounds

THE 64 FACTORY patches which come with the instrument are of a uniformly high standard, and certainly do a good job of showing the instrument off to its best advantage. I've only one thing to say to Roland's programmers: it was worth the effort.

If you want to be mightily impressed – and at the same time get an idea of the range of sounds the D50 makes available – try wandering into your local music store and playing the following: Jazz Guitar Duo (13), Arco Strings (14), Digital Native Dance (21), Bass Marimba (22), Breathy Chiffer (31), Pipe Solo (36), Soundtrack (37), Cathedral Organ (38), Vibraphone (42), Glass Voices (51), Slap Bass and Brass (73) and Intruder FX (81).

You'll hear bright and clear percussive sounds which owe a lot to the D50's samples, together with synthesized sounds which can be sparkling and bright or warm and full. And you'll hear samples and synthesized sounds working in fresh and inspiring harmony.

You'll want to keep the D50's factory sounds, so it's fortunate that Roland has stored them all on a ROM card which comes with the instrument.

The D50's onboard processing certainly plays its part in creating the overall *character* of its sound. But don't assume the reverb and chorus are being used to beef up a weak-sounding instrument. You can go through the factory patches removing reverb and chorus (which, like everything else on the D50, is a very speedy process once you're familiar with the instrument's organization), and the sounds still stand up as being impressive.

Chorusing has, of course, long been an integral part of Roland's synths, and the new digitally-implemented version holds its own against previous analog incarnations, being capable of introducing great movement and fullness to a sound.

As mentioned last month, the D50's reverb section limits you to specifying *for each patch* one of 32 "rooms" and the balance of dry and reverb signal; you don't get the multitude of variable parameters you might typically find on a dedicated reverb unit. In practice, though, the D50's reverb works remarkably well for a wide variety of the instrument's sounds.

Now, there are bound to be differing opinions on the usefulness of onboard reverb. But I reckon it's a perfectly valid addition to an instrument, and will be welcomed by many musicians.

Editing on the D50 is based around a series of displays (the synth has a 2X40-character backlit LED window, below which are five buttons variously used for display and parameter selection). These are organized on levels which correspond to the Patch/Tone/Partial organization of the instrument, and on each level you can step sideways in either direction through the displays.

Once this organization becomes familiar, it's possible to find your way around the instrument remarkably quickly. And the provision of an all-important "Exit" button ensures that you can always return to the play level with the greatest of ease – this is particularly reassuring when you first start using the D50.

Other useful functions to be found on dedicated front-panel buttons are Compare (which allows you to compare your editing attempts with the sound you started from), Undo (which recalls the initial value of the last parameter edited), and Copy (which allows you to copy parameters from one partial or one tone to any other within the current Patch).

Performance

AS MENTIONED LAST month, you can balance the volume of tones and of partials within a tone using the onboard joystick. A valuable feature, but the bad news is that making adjustments while playing notes introduces glitching (in the form of clicks) into the audio

signal. And the same thing happens if you are adjusting the chorus or reverb balance while playing sounds. The subjective loudness of this glitching varies depending on the sound(s) being played, but though it never dominates proceedings, it is nonetheless an irritation which shouldn't exist. To be fair to Roland, however, the quality control people have apparently cleaned up the signal on the D50s which are actually hitting the stores – so with luck, the clicks will click no more.

Better news is that the D50 allows you to balance partials (ie. sounds) dynamically from the keyboard. As you might expect, the D50's keyboard is sensitive to both attack velocity and channel aftertouch.

It's about time more people were introduced to the delights of release velocity and polyphonic aftertouch – and the D50's sounds would certainly respond very well to such features (release velocity on the strings would do just for starters). OK, I know that means lots more money and the D50 is well priced as it stands. But the machine won't respond to such information over MIDI either (in common with most other synths, it should be said), so there's no hope of purloining a more sophisticated keyboard to do full justice to your D50. Guess I'll just go on hoping.

The D50 allows you to set both velocity (+/−50) and aftertouch (+/−7) sensitivity for each partial within a patch. The minus values "reverse" the sensitivity, so that a hard strike or heavy pressure results in a quieter sound. This arrangement allows you to tailor sensitivity to your own playing, and to create switching and cross-fading effects. For instance, you could bring in a sampled double bass slap using a strong attack, while gentler playing would bring in a synthesized sound ▶

REVIEW

▶with a softer attack and a warmer sound.

Velocity and aftertouch can also be used to control such features as pitch-bend, filter cutoff and depth of LFO effect, while velocity can be used to alter attack time.

Further control is provided by the ability to mute individual partials when in Edit mode. This can be done very quickly using four of the front-panel patch selector buttons, and really is invaluable both for isolating particular aspects of a composite sound when editing, and for switching sound components in and out during performance.

The PG1000 programmer (which can be bought as an add-on to the D50) allows you to switch partials in and out at any time (ie. not just in Edit mode), which makes life a lot easier.

In fact, making life easier is what the PG1000 is all about. With 56 sliders controlling twice as many functions (two functions are selectable per slider), you can make adjustments to parameters much more rapidly than the D50 onboard editing system (though itself an easy arrangement) allows you to. But if you can make impressive changes to a sound more quickly this way, then you can also make a complete mess of editing more quickly too. No matter how elegant the packaging, there's just no substitute for having a clear understanding of what you're doing when you're moving all those sliders. The same goes for Manual mode on the PG, which transmits all current slider settings to the D50. Be warned.

The D50 eases itself into Exotic Tunings Territory with the inclusion of a "key follow" feature which can be applied to filter cutoff and to pitch. It's the latter application which results in some distinctly unusual sounds. Minus values reverse the pitch scaling of the keyboard; "−1" simply reverses the normal semitonal scaling so that the highest pitch is at the bottom end of the keyboard.

semitonal scaling so that the highest pitch is at the bottom end of the keyboard.

But it's the fractional scalings which hold the most interest: along with eighth- and quarter-tones, such fractions as 7/8, 5/8 and 3/8 (representing how many octaves change over 12 keys) produce some intriguing tunings which can be put to good use either singly or in combination for special effects, and for exotic sounds such as gamelan bells.

Interfacing

THE REAR PANEL of the D50 reveals the usual MIDI In, Out and Thru sockets together with left (or mono) and right stereo outputs, a headphone output and four foot controller inputs. Two of the latter have dedicated functions (volume and sustain respectively), while the other two are programmable both for their effect on the D50's sounds, and for their MIDI controller function. Internally, the programmable footswitch can be set to step upwards through patches (within the current bank), turn portamento on and off, or turn Chase mode on and off; alternatively, its internal effect can be turned off altogether.

The programmable footpedal can be set to control the aftertouch effect, the modulation effect, or the volume balance between upper and lower tones; and like the footswitch, its internal effect can be switched off. It's worth noting, though, that these settings operate globally (ie. for the instrument as a whole) rather than for individual patches.

The D50's MIDI implementation allows you to set reception and transmission on or off for each of eight MIDI message types: aftertouch, bender, modulation, volume, hold, portamento, program change and System Exclusive. Each of the assignable foot controllers can be set to any MIDI controller function from 0-95 that is within their own type (ie. switch or continuous).

Roland has included MIDI Mode 4 (Mono mode) reception on the D50. As with other Roland synths, this seems to have been implemented primarily with MIDI guitar players in mind. Voices are allocated monophonically across eight MIDI channels (there aren't many eight-string guitars around, but there *are* several guitar-to-MIDI converters that will transmit on more than twelve channels, if you so desire) with each voice independently controllable by pitch-bend. But you can't allocate a different sound to each channel, since the D50's "texture" is defined by the current key mode (whole, split, or dual).

More unusually, Roland has allowed you to set a MIDI transmission channel for *each* D50 patch – the idea being, presumably, that during performance you can layer different instruments for each patch simply by switching to the channel the relevant instrument is set to.

Another useful feature is Separate mode,

which I alluded to last month. Essentially, this allows upper and lower tones to be played from separate MIDI channels – allowing independent control (from a sequencer, say) of each D50 tone across its full range. In this mode, the D50's keyboard can only play the upper tone – though if you want to use the D50 as a master keyboard while having it play back from a sequencer, you can set the synth to "Local Off" mode from the front panel.

Conclusions

THERE'S NO QUESTION that the D50 will very quickly carve out a niche for itself among modern hi-tech keyboards. It's an original instrument with a sound all its own – retaining all the warmth and fullness of Roland's best analog synths, but adding a brightness and sparkle which comes partly from the sampled sounds, but also from the new LA synthesis system.

It's also a very accessible instrument. For a start, the ease with which you can combine sounds (synthesized and sampled) means that you can create new sound textures without necessarily having to indulge in too much programming – and there are plenty of combinations in there.

Ultimately, though, to get the most out of the instrument you have to get down to programming. Fortunately, the D50's familiar programming system and sensible layout mean that you're unlikely to find this too unpleasant or frustrating a task.

The sampled aspect of the D50 shouldn't lead you to expect everything you would expect from a sampler when it comes to accurately reproducing acoustic (or any other) sounds. The D50 is a *synthesizer* first and foremost, and don't let anyone dupe you into thinking otherwise.

On the one hand, the D50's samples act as a means of lending a greater feeling of accuracy and precision to synth sounds, while on the other, the synth side acts as a means of expanding and enhancing the current vocabulary of sampling.

But, and this is a crucial point, you don't *have* to combine synthesized and sampled sounds. Part of the flexibility of this instrument is that you can use sampled sounds by themselves, synthesized sounds by themselves, or a combination of the two.

Play rapid or staccato notes, and chances are the sample is all you'll hear anyway. Then again, the D50 is capable of producing some *big* synthesized sounds, and if you start layering four synth partials, those sounds get mighty impressive.

And while we're at it, let's not forget the onboard digital reverb and chorusing, which make a significant (and wholely welcome) contribution to the character of the machine.

Whichever way you look at it, the D50 is one hell of a good instrument, and one that has already booked itself a place in the synthesizer's hall of fame, long before most musicians have even set eyes on one. ■

PRICE D50 $1895, PG1000 $395
MORE FROM RolandCorp US, 7200 Dominion Circle, Los Angeles, CA 90040. ✆ (213) 685-5141

PATCH WORK

This is the page where MT's editorial team invite you, the readers, to demonstrate your own synthesizer programs. Send us your favorite sounds on a copy of an owner's manual chart accompanied, if possible, by a short demo-tape. Include a description of each sound and write your full name and address on each chart. If we publish your patch, you'll be rewarded with a ***complimentary one year's subscription*** *to MUSIC TECHNOLOGY. Interested? Then get twiddling and get scribbling!*

The address to send sounds to: ***Patchwork, MUSIC TECHNOLOGY, 7361 Topanga Canyon Blvd., Canoga Park, CA 91303.*** ■

OBERHEIM MATRIX 6R

O'Hammer

John O'Hara, Summit, PA

John created his 'O'Hammer' patch with his pressure-sensitive Yamaha KX76 controller keyboard in mind; applying pressure produces a sort of guitar harmonic feedback-type sound (as it raises the frequency of one of the hard-synced oscillators). He reckons that many Matrix 6R users "wouldn't have come across these specific settings of the inter-related DCO1 Frequency and Matrix Mod #0's amount. The pitch and mod wheels or levers can make it come alive, and less key pressure produces a growling sound, especially on lower notes. By holding Pedal 2 down and a high note, and quickly and repeatedly hitting a low note a sort of whammy bar effect occurs that's enhanced by varying the pressure on the held note." ■

Matrix Modulation

	Source	Amount	Destination
0	PRES	+62	DCO1F
1	PED2	+55	PORT
2	KEYB	+45	VCFFQ
3	VEL	+30	E1DEC

	0	1	2	3	4	5	6	7	8	9
00 DCO1	Freq 10	Fr/Lf1 0	Sync 3	Pw 31	PW/Lf2 0	Wave 63	Wsel BOTH	Levers BOTH	Keybd PORTA	Click OFF
10 DCO2	Freq 0	Fr/Lf1 +33	Detune +2	Pw 31	PW/Lf2 0	Wave 63	Wsel BOTH	Levers BOTH	Keybd PORTA	Click OFF
20 VCF/VCA	Mix 63	Freq 0	Fr/E1 +63	Fr/Prs 0	Res 0	Levers OFF	Keybd KEYBD	VA1 40	VA/VI +50	VA/E2 +50
30 FM/TRCK	FM 0	FM/E3 0	FM/Prs 0	Trckln KEYBD	Track1 0	Track2 15	Track3 31	Track4 47	Track5 63	
40 RMP/PRT	R1Spd 0	Trig STRIG	R2 Spd 0	Trig STRIG	Port 0	Spd/VI 0	Mode LIN	Legato OFF	Keymd UNIS	
50 ENV1	Delay 0	Attack 0	Decay 23	Sustn 0	Rel 39	Amp 55	Amp/VI +63	Trig MTRIG	Mode NORM	Lf1Trig NORM
60 ENV2	Delay 0	Attack 0	Decay 10	Sustn 50	Rel 0	Amp 55	Amp/VI +63	Trig MTRIG	Mode NORM	Lf1Trig NORM
70 ENV3	Delay 0	Attack 0	Decay 20	Sustn 0	Rel 20	Amp 40	Amp/VI +63	Trig STRIG	Mode NORM	Lf1Trig NORM
80 LFO1	Speed 58	Sp/Prs 0	Wave TRI	Retrig 0	Amp 0	Ap/R1 0	Trig OFF	Lag OFF	Smpl KEYBD	
90 LFO2	Speed 30	Sp/Kbd 0	Wave TRI	Retrig 0	Amp 0	Ap/R2 +63	Trig OFF	Lag OFF	Smpl KEYBD	

SCI PROPHET 600

Six To Go

Ron Runyon, Everett, WA

	1	2	3	4	5	6
Poly-Mode:						
FIL ENV	3	4	0	0	0	0
OSC B	0	4+	0	6	3+	0
Destination:						
FREQ A	On	On	Off	On	Off	Off
FILTER	Off	On	Off	Off	On	Off
LFO:						
FREQ	8	10	6	6	3+	7
SHAPE	T	T	T	T	T	T
INIT AMP	0	2	0	3	6+	3
Destination:						
FREQ	On	On	On	On	Off	Off
PW	Off	On	Off	Off	On	On
FILTER	On	Off	Off	Off	Off	Off

Unison:	Off	Off	On	Off	Off	Off
Osc A:						
FREQ	2	4	5	3	3	4
SYNC	On	On	Off	Off	Off	On
SHAPE	S	SP	P	P	SP	P
PW	0	5	5	5	6	7
Osc B:						
FREQ	3	4	3	6	3	3
FINE	2	0	2	0	3	0
SHAPE	S	S	TP	S	P	TP
PW	0	0	6	0	6+	7
Mix:						
A	0	–	–	1	0	–
B	0	1	3	–	0	1
Glide:	1	0	3+	0	0	1

Filter:						
CUTOFF	5+	6	4	6	3	3+
RES	2+	2	3	2	1+	4
ENV AMT	5	4+	7+	2	5+	6+
KEYBOARD	H	H	H	F	F	H
ATTACK	0	0	0	5	0	0
DECAY	2	3	0	3	2+	4
SUSTAIN	6	4	7	6	7	4
RELEASE	3	3	3	6	7	4
Amp:						
ATTACK	0	0	0	3	1	0
DECAY	5	7	6	4	2	5+
SUSTAIN	5	6	5	6+	10	0
RELEASE	1	3	3	4	7	4

A selection of hot sounds here for the Prophet 600, which their creator describes thus:

1. 'Synth Trumpet' – useful for both lead and synth sounds, and especially on accents like a brass section.

2. 'Footloose Organ' – sounds like a cheap electric organ, great for playing 'Footloose' or a lot of Peter Gabriel-type organ parts. Not so hot for leads, but it's a real good "cheesy" patch.

3. 'Feedback Lead' – a Minimoog-type lead with a twist; holding any note and moving the mixer slowly from Osc B to Osc A will introduce guitar feedback an octave above the note you're holding.

4. 'Voices Carry' – an imitation of the keyboard sound in 'Til Tuesday's Voices Carry'; it's a good solo sound but has a slow envelope (it also makes a good haunting patch).

5. 'Chord Pad' – a synth patch with plenty of timbric movement; using two mod sources and two different destinations gives it some interest.

6. 'Funky Bass' – what can you say, except it's a great funky bass! ■

Patch 1: Osc A – Tune a Perfect 4th below B; Osc B – 1 Oct

Patch 2: Osc A – 2 Oct; Osc B – 2 Oct

Patch 3: Osc A – 2 Oct + Perfect 5th & Tune PW to Square Wave; Osc B – 1 Oct

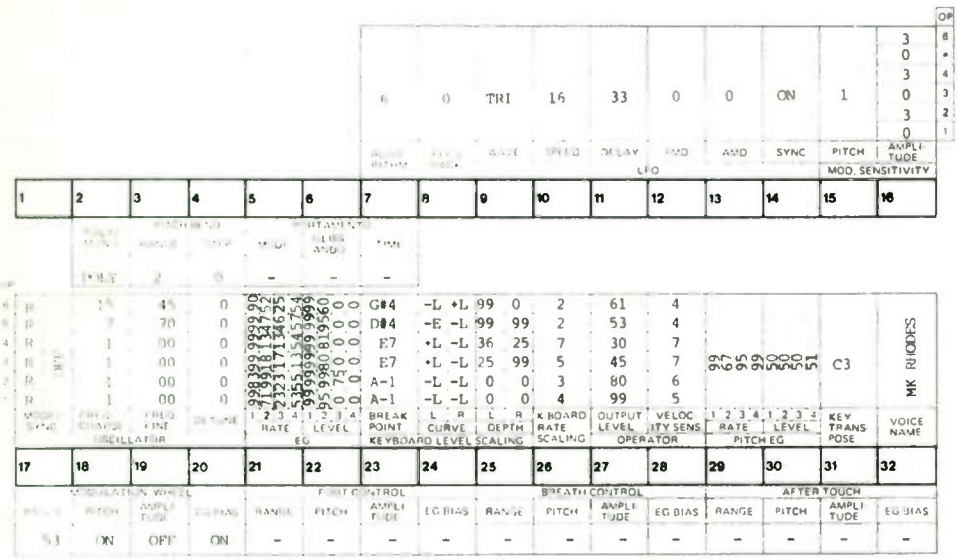
Patch 4: Osc A – 1 Oct; Osc B – 3 Oct

Patch 5: Osc A – 1 Oct; Osc B – 1 Oct

Patch 6: Osc A – 2 Oct; Osc B – 1 Oct

KEY: S = SAW; T = TRI; P = PULSE
H = HALF; F = FULL

YAMAHA DX7
MK Rhodes
Mark Knox, Philadelphia, PA



After hours of comparing this patch to his suitcase Rhodes 73, Mark finally came up with a patch that "really sounds like an old muddy, funky Rhodes, right down to the 'bouncing damper' effect and the pitch-bend of the tines. MIDI this sound to a couple more TXs with the same sound (and slightly detuned) for a truly awesome result! The only drawback to 'MK Rhodes' is that its output is not as hot as usual..." ■

Valhala Data Cassettes and ROM Cartridges
for Ensoniq ESQI
801A/B, 802A/B, 803A/B, 804A/B

IMAGINE THIS SCENE for a moment. Proudly clutching a wad of hard-earned cash, you dash into your local music store, returning several minutes later groaning and puffing under the weight of an Ensoniq ESQI. Arriving home, hoping your neighbors are watching enviously, you go inside, plug it in, and presto! You've only got 40 sounds to play with.

However, you remember reading that you could get "0-120 in 3.6 seconds." You now realize that you need the "optional" EEPROM cartridge which at around $69.95 is decidedly not a snip. Once you've bought said cartridge you crave even more sounds, so in march Valhala, who have just produced eight data tapes with 40 new sounds apiece. And as these only cost $19.95 each, you feel much happier already. (They have also managed to put these sounds on various ROM cartridges, if you prefer, but you do pay for the convenience.)

The tapes themselves contain a range of patches including 49 pianos, brass, strings, percussion, lead, bass, woodwind, sound effects and some more conventional analog-type sounds. The best of the pianos are the clean and lifelike ACGRDI on tape 801A, the deep and gutsy (especially in the lower regions) LOPNO on 801B, and (also on 801B) SUSPNO, which offers more in the sustain department but less in the way of weight. The Steinway impression on 802B (STEINI) is pleasant, but the FM pianos and FM Rhodes on 804A and

804B tend to sound thin, unspectacular, and all rather similar.

At this point you may have already noticed the main problem with the Valhala range; not all patches of a similar type are on the same cassette. This means you need to buy all eight tapes to make sure you've got all the variations you want, and once you've bought them, it's harder to keep track of which sound is where.

Basses 1-10 are spread out between 801A, 801B, 802A and 804A. Most impressive is BASS4 with its distinct dancefloor appeal, and BASS2, which is reminiscent of Yamaha's FlapBass on the DX100. Also on 804A is the not-very-good-at-all KORGBS, and the only slightly better ESQBAS.

Strings are even more diverse, with competent patches on every tape out of the eight, although string enthusiasts would be well advised to start with tape 804A, which has some of the best, notably HISTR, DXVIOL and CARLOS. MELSTR (803B) and BACH (801A) are also worthy, while OBX-A (803B) is brassier.

The two lead sounds which really shine, and which would rise above any mix, are WENDYC on 801A and FLTMAR on 803A. The former is powerful and atmospheric, while the latter uses prime, synth3 and sine waveforms on the Ensoniq's oscillators to good effect. Other good patches to watch out for include CHOIR2 (801B), PROFIT (803A) – a sort of heavy metal guitar if you keep your eyes shut – and RPTDRM (803B), a repeating kick-drum that changes in tempo rather than pitch as you move along the keyboard.

The prize for the worst name goes jointly to ISAO and TOMITA (803A and 804A respectively), surely a case of misplaced inspiration if ever there was one, though the former is actually rather pleasant.

As with any collection of synth sounds, it's useful to remember that what at first appears to be uninspiring can be made uniquely interesting by a bit of judicious tweaking of the parameters. And layering, say, a choir with a relatively long attack over a bass sound can prove unexpectedly rewarding.

Loading 40 sounds at a time from cassette into the ESQ1 doesn't take very long, and that's when the fun starts. At best, there are some superb and original sounds; at worst, there are building blocks and food for thought. ■ *David Bradwell*

PRICES $19.95 per tape; $59.95 for 80-voice ROM; $116.95 for 160-voice ROM; and $169.95 for 320-voice ROM

MORE FROM Valhala, PO Box 20157, Ferndale, MI 48220. Tel. (313) 548-9360

ROLAND DEP3

Despite its grand title, the DEP3 is first and foremost a digital reverb that can produce straight delay effects as an alternative. Is the compromise a good one? *Review by Paul White.*

THE PROBLEM WITH producing an exciting machine is that, if you want to follow it up with some cheaper but less sophisticated models, those models inevitably pale into insignificance alongside the device that started it all. Yamaha certainly found that to be the case with their DX synths, and now Roland must be wary of treading a similar path with their DEP series of effects units.

DEP stands for Digital Effects Processor, and so far there are just two models in the line. The first, the DEP5, was unveiled last year and proved a runaway success, since in addition to offering a wide array of high-quality reverb treatments and longer delay effects, it also offered the unique option of letting you use two of those effects simultaneously.

Now the DEP5 has a smaller brother, the DEP3, and comparisons are inevitable. The DEP3 is somewhat cheaper, and is designed specifically to create reverb (including gated and reverse effects) and unmodulated digital delays, with or without regeneration. You're also restricted to using one effect at a time, as per just about every other piece of outboard gear going.

But while that brief description may seem fairly unexciting on the face of it, the fact of the matter is that the DEP3 is probably the cheapest programmable reverb around – no small feat in itself.

Format

THIS IS A 16-bit, mono-in, stereo-out system, with a specification that indicates low noise and negligible distortion, along with a bandwidth (12kHz) that makes even the delay settings bright enough for demanding studio applications.

For a programmable reverb unit to be cheap, though, compromises have to be made. For a start, the DEP3's designers have left out all but the most important variable parameters: the reflection density is not variable, nor is the early reflection spacing or reflection pattern.

But you can still choose a basic reverb type, vary the decay time and pre-delay, and mess around with such things as high-frequency damping and a three-band equalizer – and all those parameter values can be incorporated into programs.

Luckily, none of those wretched, unfriendly, non-interactive up/down buttons are involved in setting up an effect, only good ol' fashioned knobs and dials. True, there are up/down buttons for selecting the program number, but in the light of those rotary controls, I think most musicians should be able to live with that.

All the connections are to be found on the back panel, where standard unbalanced ¼" jacks are used for the input, the left and right outputs and the inevitable remote socket, which accepts a conventional footswitch for bypassing the effect. There's a switch to cancel the dry or direct sound when the DEP3 is used with a mixer, and there's the very sensible inclusion of a dual level switch to enable the machine to work at +4dBm or −20dBm. For recording, −10dBm would have been preferable, but it seems almost all effects manufacturers are now catering for those perverse people who plug instruments straight into the things without going via a mixer. It all sounds quite unhygienic to me. Still, the levels are close enough to match up to budget recording gear with no problems. The only other feature on the back panel is a lone DIN socket to handle any MIDI In signals you might care to feed it – more on this later.

When you unpack your shiny new DEP3, you find that the first 20 programs (out of a total of 99) are inhabited by factory samples. These may be overwritten if you need the space, but can always be recalled if necessary by using a specially contrived power-up sequence that is unlikely to be repeated by accident.

Working from left to right, the first thing you come across is a six-section input level LED meter logically situated next to the Input level control. The meter reads from −14dB to +12dB, but in practice, the DEP3 is pretty tolerant of high input levels anyway.

The EQ section is a straightforward three-band affair offering 12dB of cut or boost at 100Hz, 1kHz and 10kHz. Without going wild with this, it's possible to change the character of the reverb treatment quite dramatically – but the extra range is there if you need it.

To set up the reverb sound itself, there are only four parameters to vary. First, there's an 11-way selector switch offering a choice of three room settings, three halls, two plates, one delay, and both gated and reversed reverb. To this pre-delay can be added – up to 120msecs, and the decay time can be varied up to a maximum of 99 seconds for the longest hall setting. A tweak on the high-frequency damping control, and you've got yourself a simulation of the effect caused by high frequencies being absorbed by soft furnishings and curtains.

There's only one thing to keep in mind when setting up or modifying a treatment, and that's that the parameters may bear no resemblance to the knob positions. This is because any program can be recalled, but the front panel knobs will still be indicating the last setting they were set to. To get a knob to work, it must be moved slightly, after which the internal computer relinquishes control of that particular parameter. If you want to set up a brand new effect from scratch, it's good practice to give all the knobs a twirl before

starting, including the EQ section and the selector switch.

Considering the ease of use of a system using control knobs rather than those unspeakable buttons, this minor inconvenience seems a small price to pay – and users of early programmable synths (like the Prophet 5 and Juno 60) should be quite familiar with it in any case.

Another thoughtful touch involves the Manual button, which lets you recall a program, alter the parameters, and then compare it to the original version before committing it to memory – again, synth programmers will already be familiar with this system. The button is mounted next to a two-digit LED numeric display which normally shows the program number currently being used. During programming, the display can also indicate which of 16 MIDI channels the DEP3 has been set to receive on.

Overall, the Roland's front panel is perfectly logical to use, and for those who don't like reading, the unit's entire operation can be worked out without the aid of the manual in just a few minutes.

On the MIDI side, the DEP3's 99 programs respond to program-change information in the same range, but unfortunately there's no way you can assign specific effects to specific program numbers; you have to arrange for your synth sounds to be in the memory locations that correspond to your desired reverb treatments – or vice versa.

As already indicated, you can control the DEP3 on any one of 16 MIDI channels, though there's also the option of using Omni mode so the unit will respond to patch-change information regardless of the channel it arrives on. For the more ambitious, the reverb parameters themselves can be accessed by means of MIDI System Exclusive codes, all of which are documented in the manual.

The Effects

THE SMALLEST ROOM setting (not *that* smallest room), is a clangy, metallic tunnel echo which is great for special effects such as robot voices, trashy dancefloor drums and so on. It doesn't sound like a natural reverb, but then again, it's not intended to.

The next two rooms sound more like *rooms* and are good for filling out a sound or just adding a touch of ambience. They have a certain amount of coloration and flutter, just like a real room would have, and, as such, are not unconvincing.

The Hall settings are less colored than the rooms and also less dense, so that they can recreate the longer delays between reflections which naturally occur in a large building. The HF roll-off helps enormously here to simulate boomy, cavernous effects, and a bit of pre-delay also heightens the impression of space.

Both Plate settings sound bright, with just the right degree of metallic edge to conjure up the old mechanical plate sound, still a popular drum treatment for producing an up-front, attacking sound despite its age. Plates are also used to add brightness and shimmer to vocals, and these simulations are equally adept when put to this use.

Delay is a simple echo effect with no modulation, so there's no chorus or flanging available. Here, the Pre-delay control sets the delay time (up to a maximum of 500msecs), and the Reverb Time control sets the feedback for repeat echoes. With the feedback at maximum, the sound cycles around for several minutes before dying away.

The first of the Non-linear selections is gated reverb, though the DEP3's version of this is far from conventional. The Pre-delay control may be used as normal, after which you set a reverb decay time which has nothing to do with the gate time, the latter, in turn, being set with the HF Damp control. Confused? Well, what happens is this. The reverb is initiated by a drum (or whatever) and, after the required pre-delay, starts to decay. However, if the decay is not complete before the gate time has elapsed, the reverb is chopped off in its prime, giving a classic gated reverb sound. The beauty of this system is that you can arrange to hear some decay in level before the gate cuts off by setting a fairly fast decay time, or you can have a level burst by setting a long decay time and then chopping that off before it's had a chance to decay.

The function of the DEP3's Reverse setting is also a little unusual. Here the reverb starts to build up as soon as the drum beat has occurred, again at a rate according to the setting of the Reverb Time control. Then, when it reaches a maximum level, it abruptly shuts off, giving an illusion of reversed reverb. That's pretty standard stuff, but what Roland have done is to use the pre-delay time as a sort of mute or gate to create a dead space between the sound and the reverb following it. This is *not* the same as normal pre-delay, since the reverse envelope is still doing its stuff during the pre-delay period – it's just that you can't hear it.

Conclusions

THIS IS A good compromise between a fully programmable reverb and a preset system. The parameters that make the most difference are all there to vary, and there's no shortage of memories to store the results in.

The use of user-friendly controls makes the machine a joy to work with, and the sound quality, though not as refined as a really expensive reverb might offer, is certainly well up to home recording standards, and wouldn't be out of place in a small professional studio.

The MIDI program-change facility makes the DEP3 a candidate for semi-automated use running from a MIDI sequencer – though the lack of patch assignment could put off any live players who rely heavily on MIDI to set up their configurations.

The delay effects are fine as far as they go, even if some kind of external modulation input would have been handy.

But put the virtues of this machine in one hand, its price in the other, and perform the usual balancing act, and I think I can honestly say that you get your money's worth. ■

PRICE $625

MORE FROM RolandCorp US, 7200 Dominion Circle, Los Angeles, CA 90040. ✆ (213) 685-5141

SOUND READING

The Complete Guide to MIDI Software

by Howard Massey and the staff of PASS

THE CURRENT STATE of music software being as it is – constantly under improvement, continually being expanded, and growing at a rate equal to the multiplying ability of gremlins tossed into a lake – it's difficult to believe that any one book can claim to be a "complete guide." (Especially when it seems that all it takes is one person with a computer, an application and some basic programming knowledge to create new software.) If you're making attempts to keep up with new programs, you'll realize how overwhelming the task can be.

Nevertheless, Howard Massey, et al, have made a valiant attempt to look at the most popular comput-

ers being used for MIDI applications (including IBMs, Macintoshes, Ataris and Commodores) and the software programs which are most readily available. The spec sheets, suggested retail prices, manufacturers' addresses and details of special features allow comparisons and analysis of specific needs right in your own home. There are 60 reviews and "guided tours" of the most popular MIDI software packages, which typically lead you through loading, operations, possible applications, and a brief opinion on whether the program does what it's supposed to do.

This book could be used in a variety of ways. It would be an excellent starting point, for example, for someone looking to design a personal MIDI studio, before any equipment has been purchased. You can compare computers (as far as memory capacity, necessary accessories, and possible limitations or strengths are concerned) with the reviewed programs, their ease of operation or complexity, and your individual needs. Or, if you already own a computer, this book presents at least four (for the Atari ST), and up to 24 (for the Commodore), possible musical software programs for use in the MIDI stu-

dio. Price comparisons would prove extremely helpful in budgeting your software dollars.

The reviews use an "informed but impartial" tone in describing the strengths and weaknesses of each software package. They often take into account practical considerations as well (such as the cost benefits of floppy vs. RAM storage), and use clear, concise prose for the analyses. Anyone who has tried to use a poorly written owner's manual will appreciate the straightforward style of these writers.

So, if you're planning on setting up a MIDI studio, or if you'd like to know more about the software available for your own computer, it would be worth your while to get a copy of this book. But when you see the title, be a trifle skeptical over the "complete" part. A book like this can only be really complete during the month it's written in. ■ *Deborah Parisi*

Touch Sensitivity: Building Keyboard Skills for Playing and Home Recording

by Jack Wheaton
with Peter Alexander

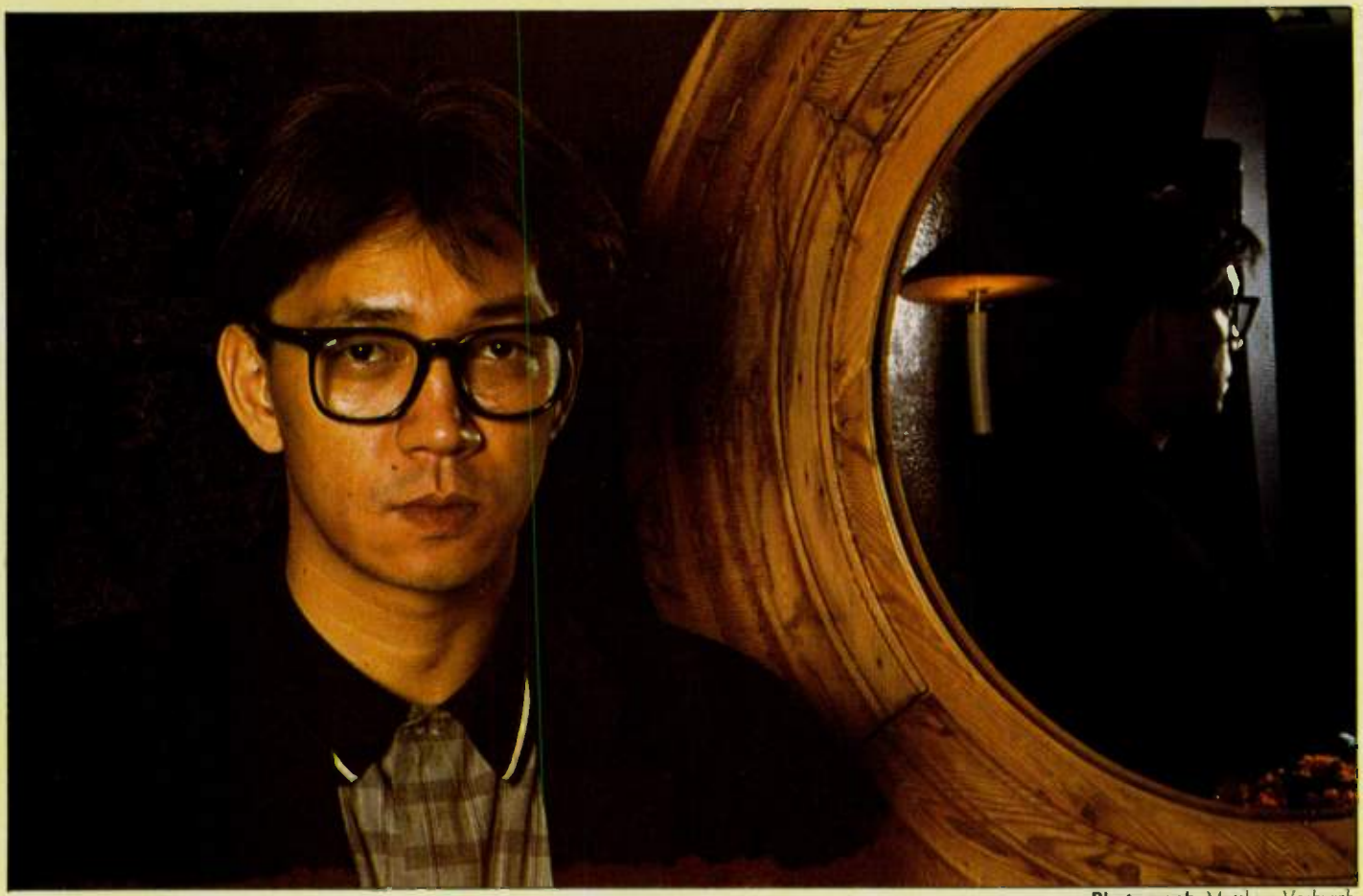
THIS AMBITIOUS BOOK is much more than a bland, pedagogical approach to training keyboard players. Instead of considering the piano or organ as the primary instrument, the writers concentrate on the unique challenges posed by the synthesizer to modern musicians. While a glance through the pages shows many of the same types of simple exercises found in most beginning piano courses, the writers breathe life into their text by stressing creativity, experimentation with the unique capabilities of electronic instruments, and just plain fun.

Concern over the total musician is evident in sections covering the least stressful sitting position, how to properly warm-up and tone the muscles of the hands, and how to get the most out of your practice times. "User-friendly" is an appropriate description, considering the relationship which is developed with the writers under their warm encouragement. The cassette tape which accompanies the book also helps the feel of familiarity, with Peter Alexander narrating and performing the exercises notated in the text.

Despite some weak diagrams and a few annoying typographical errors early in the book, most of the lessons are marked by clarity and a real understanding of the frustrations of keyboard players. Indicative of the ambitious undertaking of the writers is the attempt to reach all players – at all stages of experience. Unfortunately, although this *sounds* good, it is unlikely that an advanced player (defined as "someone who already plays piano or organ and is a music reader") would persist in practicing the extremely basic exercises, no matter what the promised reward. An advanced player's ego simply wouldn't permit it. The writers assume that you "have little to no music reading skills," but if you do, you'll find it difficult to follow through the book.

If, however, you've never played keyboards before, or if you're basically a "by-ear" player with the discipline and desire to learn to read music, this book is a must. The beginning player will especially benefit, learning to read music and jam simultaneously.

Bringing together strong technical training, the new technology, and a creative, good-natured attitude towards what music should be is quite an accomplishment. Hopefully, this will be continued (and will perhaps be more useful for advanced players) in Volume 2. ■ *Deborah Parisi*



Photograph *Matthew Vosburgh*

Stranger in a Strange World

At a time when musicians are finding new ways to put more feel into electronic music, Yellow Magic Orchestra founder-member Ryuichi Sakamoto's experience raises a new question: does electronic music really need to sound human?

Interview by Paul Tingen.

HALFWAY THROUGH MY conversation with Ryuichi Sakamoto, I suddenly make a near fatal mistake. We're talking about the equipment Sakamoto is using, and my Japanese host has just explained to me that a NEC computer and a DX7 synthesizer are the main things he's working with at the moment. So I ask whether the NEC computer is the "brains" of his music system – with "brains" actually meaning "nerve center," the device off of which he runs all his other keyboards and assorted machinery. Yet on hearing the word "brains," Sakamoto suddenly jerks and shouts: "No, no, no. Just tool, not brains," and pointing at the side of his head he continues: "That's the brains." It's the only time during the interview that he's fierce, almost angry, about what he probably saw as an all too common misunderstanding. But that example also serves to illustrate the kind of uphill struggle an interview with Sakamoto can be.

First of all, there's his English. His limited vocabulary and obvious problems with grammar hamper a lot of the communication between us – more, in fact, than I had expected from the short telephone conversation I'd previously had with him to arrange the interview. Second, there's his almost complete lack of interest in discussing the finer points of musical technology. For, in common with several other of today's technological innovators, he seems to regard his technology as no more than a tool. He doesn't have a studio of his own, and his "tools" are remarkably

simple and sparse for someone as highly regarded as the hi-tech innovator he is. At his home in Tokyo he works only with that NEC computer, running an eight-track sequencing package called Come On Music, with the DX7 as his only MIDI keyboard, a small mixing console, and a Yamaha SPX90 effects rack – the last moving him to one of his only spontaneous remarks about equipment: "It is very good, the SPX."

On top of all this, Sakamoto also seems to be one of those artists who doesn't reflect on what he is doing, but just follows his intuition. It might be a Japanese trait, or it might be a result of our language problems, but every time I ask Sakamoto *why* he has done something, he answers with "I don't know," or, if I'm lucky, a two-line answer which partly repeats the question. And when I put an observation of my own to him, he often seems genuinely surprised about the point I make, chews it over for a while, and then mutters something which boils down to: "It might be like that, but it might also not be like that."

In short, Mr Sakamoto is a bit of an interviewer's nightmare. Apart from that, he is extremely polite and does everything to ensure that I'm OK, pouring me tea and coffee and dashing over to the bathroom to get a towel when I spill some coffee on the table.

I'm talking to the Japanese composer, actor, producer and keyboard wizard in a room in the small yet stylish surroundings of the Blakes Hotel in West London, just after he's finished some string sessions for David Sylvian's forthcoming album. The hotel has a distinctly oriental feel, with its cane benches, bamboo ornamentation, plethora of mirrors and plants, and its gigantic wood and paper umbrella in the entrance hall. The homey atmosphere of the place obviously suits Sakamoto who, in real life, has nothing of the hard image which he portrays in a lot of photographs, and as Captain Yonio in the film *Merry Christmas, Mr. Lawrence*. He is small, wears dark glasses and a long black jacket, and has a shy, modest demeanor. He listens attentively to my questions, often answering with a chuckle, smoking almost continuously, and occasionally apologizing for his bad English.

But while "why" and "how" questions are met with stumbling blocks, "what" questions seem to go down a bit easier with Sakamoto, since he obviously enjoys elaborating about what he has been doing over the last year, during which relatively little has been heard from him here in the West.

It appears that he's spent a lot of time on projects in his native Japan . . . "At the beginning of last year I made a new album called *Futurist*, which was released only in Japan. The album title was inspired by the Futurist movement, prominent in Milan around 1909, in which I was very interested. Side A was very American pop-oriented, whereas Side B was more aggressive, featuring a mixture of hip-hop rhythms and Italian opera. The sound of the album was also influenced by the movie *Dune*, which has a lot of low sounds."

Following the release of the album, Sakamoto went on a lengthy Japanese tour, and then released – again only in Japan – a live album of that tour called *Mediabahn*.

Last August he went to Bath, England, to produce an album called *Hope in a Darkened Heart* by local artist Virginia Astley. Then it was back to the East. "In September I made a soundtrack for a Japanese animation movie, my third one. After that I went to China for the shooting of *The Last Emperor*, a movie directed by Bertolucci, in which I acted with John Long and Peter O'Toole." It was only the second time that Sakamoto has acted in a movie, and as with *Merry Christmas, Mr. Lawrence*, he was also commissioned to write the score for the film – a project which he still had to finish at the time of our interview.

After the shooting of *The Last Emperor*, Sakamoto finally began work on a venture of greater significance to the western music-lover.

"In December I made the demo tapes for a new album, which I recorded in New York. The album, called *Neo Geo*, was produced by Bill Laswell and me, and will be released worldwide by CBS somewhere during the summer."

This is good news, because Sakamoto's output to western record stores over the last few years has been modest, to say the least. His last western release was a compilation of some of his Japanese solo albums called *Illustrated Musical Encyclopedia*, which was available here as an import on 10 Records. The record was largely a solo venture, though it featured a collaboration with Thomas Dolby (the song 'Fieldwork') and some contributions by, for example, percussionist David Van Tieghem and Simon Jeffes of the Penguin Cafe Orchestra. By and large, though, the album was dominated by Sakamoto's approach to keyboard and computer music: fresh, delicately textured sounds, captured in strict, precise, even cold rhythms. And, of course, the usual potpourri of influences: Tibetan and Japanese sounding tunes, big band jazz, a rap in Thai, and "technopop" – the category Sakamoto helped create when he was playing with the now almost legendary Yellow Magic Orchestra.

Yet there's no doubt this artist has come a long way since YMO stunned the world with their hard-edged rhythms and inventive use of synthesizer timbre and melody. The classically trained Sakamoto (he studied composition at Tokyo's University of Arts), founded YMO with drummer Yukihiro Takahashi and producer, bass player and keyboard programmer Harumi Hosono in 1978.

Throughout the band's career, YMO remained an anomaly in Japan's conservative music scene. Musically they were way ahead of their time, taking the formal, all-electronic style instigated by Kraftwerk in Germany into new areas – yet like Kraftwerk, their ability to laugh at themselves gave them an acceptability that many electronic acts have failed to find, both before and since.

"I've called myself a world musician before, because I don't feel as if I belong to one specific culture. I feel a stranger everywhere, even in Japan".

YMO disbanded in late 1983, having pioneered the techniques of sampling and computer music programming to a degree which took musicians in the West quite some time to catch up with. Sakamoto started work on what turned out to be a prolific solo career. He released a number of solo albums, and began a collaboration with ex-Japan frontman David Sylvian which has lasted until today. He also produced several albums for his wife and fellow songwriter and pianist Akiko Yano.

But it was his leading role in *Merry Christmas, Mr. Lawrence*, which largely made his name in the West. His acting showed a stunning confidence and attention to detail, and the soundtrack he wrote for the film – featuring a vocal version of the main theme, 'Forbidden Colours,' sung by David Sylvian – won him a British Academy award.

NOW WE ARE about to hear *Neo Geo* (it means New Geography), an album that marks a musical turning-point in Sakamoto's career. As we're still in the "what" area, he's happy to elaborate.

"*Neo Geo* is a new way for me. Usually I use a lot of computerized sequences. But this time, the album has more acoustic instruments, meaning real drums, real guitar and real bass. For other people that kind of real music is normal, but for me it's a pretty new thing." The artist laughs. The new album features Tony Williams and Sly Dunbar on drums, Bootsy Collins (Prefunk, Funkadelic) and Bill Laswell on bass, Harry Kubota and Eddie Martinez on guitar, David Van Tieghem on percussion and, surprisingly, Iggy Pop, who sang and wrote the lyrics for one song. When I ask Sakamoto about other vocals on the album, he initially comes up with a very eastern way of expressing himself: "There are some more voices on it, but it's not the same." Then he continues: "The rest is sampled and there's some talking. As far as making the record goes, I prepared every tempo, every sequence and every part on my ▶

computer in Tokyo and then brought the tapes over to New York, where I overdubbed the other instruments, keeping part of the computer sequences."

Apart from the use of real guitars, bass and drums, there's another thing which makes *Neo Geo* stand out from Sakamoto's previous work . . .

"I used two traditional Okinawa songs. Okinawa is a Southern Japanese island. It's new for me to use traditional music directly like this." Very true: in the past, Sakamoto has used elements of traditional Far Eastern music, but has never quoted melodies or songs directly.

The combination of using a whole set of American musicians, who then have to play Japanese traditional tunes, raises an interesting point. But, sadly, this is a "why" question, and one that meets with a reply that is as confusing as it is enigmatic: "The music itself answers the question. For example, I used the traditional Okinawa songs with a go-go beat from Washington DC, using Bootsy Collins on bass. It's hard to tell what kind of music it is. But I think this is my new way."

It could also be argued, however, that this is just another chapter in Sakamoto's ongoing experimentation with Western and Eastern music styles, something that's taken place without the artist committing himself to either side – though he once remarked that he considers himself more Western in approach, because he uses melody, harmony and rhythm.

He comments: "I've called myself a world musician before, because I don't feel as if I belong to one specific culture. Somewhere this means that I feel a stranger everywhere, slightly remote from everything, even in Japan." To which he quickly adds, giggling: "But don't take that word 'stranger' too seriously. It sounds so pessimistic. It's just that I like mixing ethnic music with a black beat and electronic technopop. I take my influences from everywhere, and when I'm deciding on a treatment, I don't care whether a song is traditional Japanese or not."

And why (sorry) did Sakamoto decide to use real players for his new album? "The idea came from Bill Laswell. Yet even now I prefer an electronic beat." Sakamoto laughs. This sentiment seems to negate his earlier statement that using real instruments is his new direction. But the artist can explain this apparent contradiction. Because for Ryuichi Sakamoto, the electronic beat is part of a self-created heritage – and unlike a lot of Western musicians, he's not afraid of the coldness which immaculate computer performances so often invoke. He wouldn't dream of endeavoring, as so many modern drum and sequencer programmers do, to instill a false "human feel" into the electronic elements of his music.

"Basically," he says, "I like that coldness. It's also a kind of private history. In my opinion, YMO and Kraftwerk invented technopop and I will not move away from that direction. It's my roots."

ON A MORE practical level, we discuss the equipment which Sakamoto uses to shape his ideas. I've already mentioned the equipment he uses to demo his ideas at home. From there, it's on to nearby Lentil Studios to complete recording. It's there that he stores his Fairlight II, along with a PPG Wave 2.3, an Emulator II and a Prophet 5 – the last only for its ability to provide a guide click.

"I find it easier to compose for film than for my solo albums. When I'm doing a soundtrack there's an object already in front of me, and I just adapt to that."

"Poor old Prophet 5," Sakamoto reminisces. "I don't use it anymore for anything else, because I'm tired of the sound. I've used it too much, I think. Before, I didn't care which keyboard I used, because I felt that it was me who created the sound, not the keyboard. But I have to admit that my taste now gears towards the digitals and the sampling machines. I use the DX7 a lot, for which I have a programming software package which I can run on the NEC, and I usually bring in a whole lot of CDs, records and a Sony PCM F1 recorder to sample from and with."

Does he still play the acoustic piano? "Very little. There were two solo piano pieces of mine on a compilation album called *Piano Music*, which appeared on Peter Baumann's Private Music label, but now I have to admit that I play the DX7 too much. It's my main keyboard. I'm not a good piano player anymore, if I ever was one (laughs). On the *Mediabahn* tour I used Yamaha's new MIDI acoustic piano, because I need an acoustic piano sound, but I didn't like it. The touch is too heavy for me."

On an inspirational level, Sakamoto relates that "the chords and the melody usually come together. Sometimes I write it down on paper, sometimes I play it on the DX7 and store it in the NEC.

"There are a lot of ways in which music comes to me. Sometimes I just play and improvise. Sometimes, before playing, I might have been thinking about words and concepts or the image of a landscape. I'm usually seeing something visual, and I work that into a piece or into sound."

So how does he go about composing music for film?

"I find it easier to compose for film than for my solo albums. Making a solo album is a very conscious thing for me. When I'm doing a soundtrack there's an object already in front of me, and I just adapt to that."

Sakamoto agrees with the suggestion that his music has become mellower over the years.

"It's not a conscious thing, but it seems to be where my natural musical taste is leading me. With YMO there were a lot of very hard, rigid beats. On the other hand, my favorite composer is Debussy. I love that kind of delicate, ambient music. I also like Eno's music a lot, and earlier Steve Reich and Philip Glass. But I didn't want to bring my 'important' musical taste into YMO. I wouldn't, because YMO was just a pop band."

On another level, Sakamoto's reluctance to let too much of his musical taste influence YMO refers back to a personal philosophy about balance which lies at the root of his musical work.

"When I write music, I'm caring about the balance. I'm not sure what the balance means (laughs) . . . But it's a kind of concept. I try to find the balance between my personal thing and a more public thing, so I wouldn't impose my taste on YMO. Or I try to hold a balance between artistic and commercial. And also, it shouldn't become too emotional.

"Another example would be the balance between sound itself and the structure of music. Or between technology and human feeling, or between the old world and the new world. In Japan there are a lot of traditional things in society, and also modern things coming up very drastically. So I'm always caring about the balance of two opposite things. If you say that my music isn't always that emotional, then you're right. But it is very personal. This concept of balance is my personal thing."

Later, while Ryuichi Sakamoto guides me to the hotel exit, I become more and more impressed with the personal warmth which he displays, and his natural, unaffected behavior, something so rare in the glamor-soaked world of fame, money and music. Somehow, Sakamoto doesn't quite fit in.

I look at this small, unassuming character and wonder who he really is. A stranger? Really, a stranger in a strange world. ■

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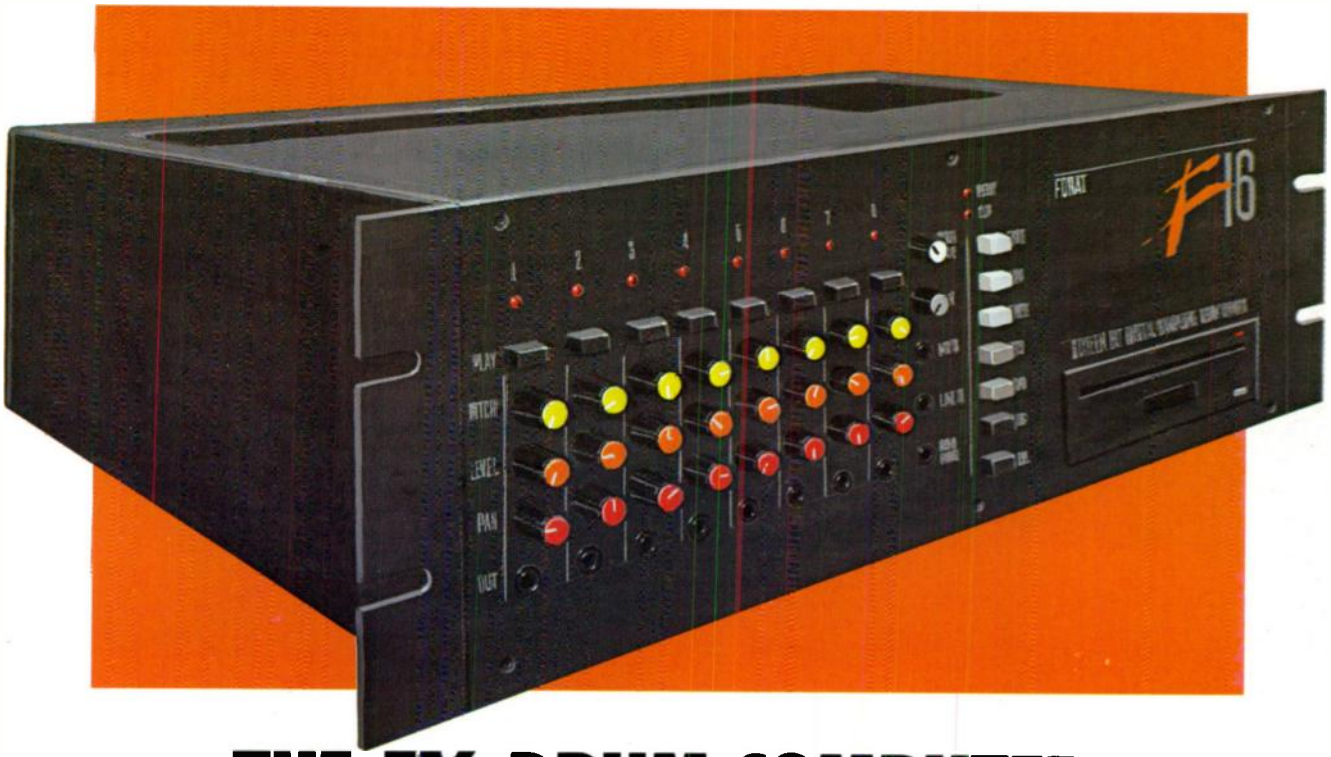

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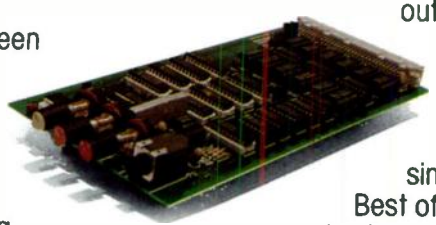

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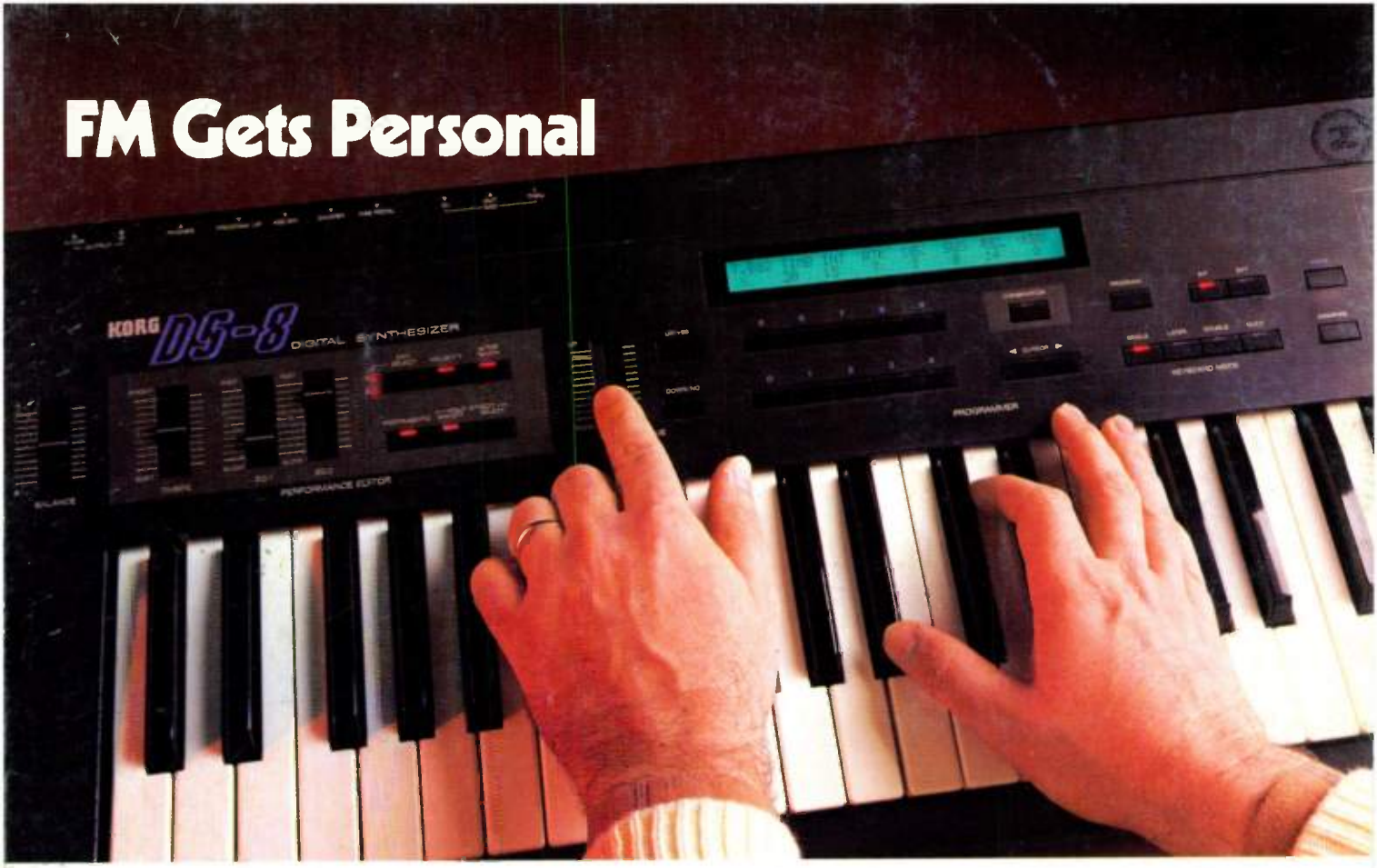

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