NOVEMBER 1987
USA $2.95
CANADA $3.95

MUSIC TECHNOLOGY

KITARO
Towards the West

DANIEL LANOIS
Art from Chaos

MADE IN JAPAN
A Western View

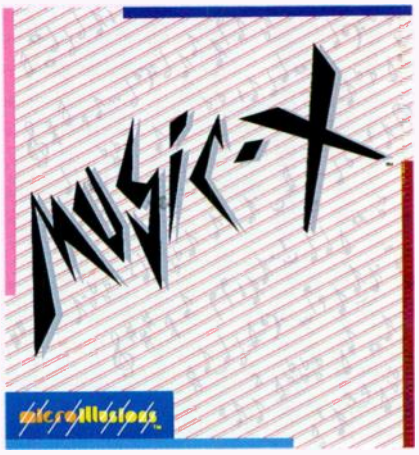
ON STAGE
Spyro Gyra

FAIRLIGHT SERIES III
A Primer

REVIEWS
Roland VP70
Yamaha TX802
Keytek CTS2000
Cakewalk Sequencer
RSF Sampling Drum Machine
Unisynth Electronic Guitars

0 74470 19993 9 11

MUSIC-X™



MUSIC X SOFTWARE, *like a fine instrument is crafted from the heart. It is more than an excellent tool, it is also a work of art.*

COMMITMENT: *We have committed ourselves to pushing ahead state of the art in professional music software, enabling you to open new worlds of creativity at a cost, both in hardware and software, that is well within the budget of any serious musician.*

NO COMPROMISES *or shortcuts have been tolerated as we designed this product. The master clock is accurate to 1 millisecond with a resolution of 192 clocks per quarter note. Sequences and library data can be any length, limited only by available memory — if you want, you can dump a 100K or larger sample into a library entry!*

KEYBOARD MAPPING *features allow almost any function of the sequencer to be controlled from a MIDI keyboard, footpedal, or other MIDI device. This includes starting/stopping the sequencer, initiating sequences, and even changing the key map itself!*

REAL TIME: *The system supports real-time recording of systems exclusive data, as well as full graphic-oriented and event-oriented editing of sequences. You can even record while in edit mode and watch notes appear on your edit display as you play them!*

LIBRARIAN: *A configurable librarian is included with the program. You can teach the librarian how to communicate with any MIDI instrument which outputs system exclusive data.*

EDITING: *An impressive battery of editing features will be supported. In fact, new editing features are being added daily as we interact with our network of working, professional musicians whose input has greatly contributed to the quality of this program.*

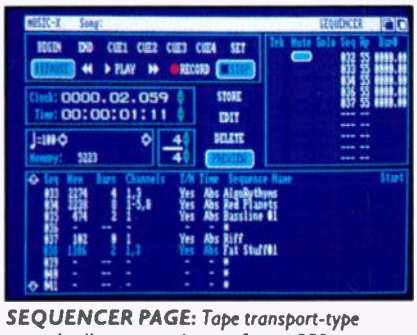
COMMITMENT: *Our commitment to music production does not stop here. A future product, Patch Editor Construction Kit, will allow you to create graphical patch editors for virtually any synthesizer you may own. Some technical knowledge will be required, but since patch editors, once created, can be traded between users, you should have no problem getting an editor for your needs.*

THE POWER: *Part of the power of Music-X comes from the computer it was created for: The Amiga, one of the most powerful and inexpensive personal computers available. At last you can run these many powerful applications in an environment that is a pleasure rather than a chore to use!*

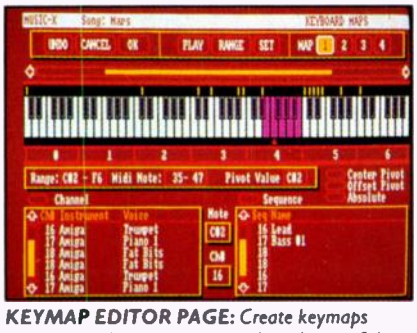
MICRO MIDI: *Although Music-X will work with any of the many MIDI interfaces for the Amiga, we offer our own MIDI interface which we feel is a cut above. It features six outputs (each output switchable as OUT, THRU or OFF), two switch-selectable inputs, a channel loading indicator, and an external clock output (sync/start stop) for synchronizing older, non-MIDI drum machines, and a serial pass-thru!*

MICRO SMPTE: *This complete SMPTE Reader will allow Music-X to synchronize with video or audio tape decks. It connects to the Amiga parallel interface and includes a pass-thru so as not to interfere with printer operation. Our Micro SMPTE is compatible with all Amiga models (A500/A1000/A2000).*

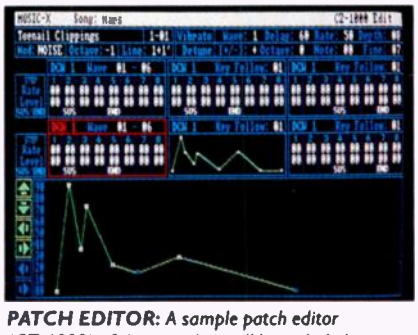
PHOTON VIDEO: *Photon Video is a complete, integrated video animation system. It includes facilities for both 2-D and 3-D animation, as well as automatic tape transport control and real time playback of rendered images. Our 3-D rendering module supports variable light sources, shadows, transparency, and reflections in a 3-D environment. Other modules include Cel Animator, Object Editor and Transport Controller with SMPTE support.*



SEQUENCER PAGE: *Tape transport-type controls allow manipulation of up to 250 sequences; each contain 16 MIDI channels worth of data.*



KEYMAP EDITOR PAGE: *Create keymaps by dragging the mouse over a selected area of the keyboard. The highlighted region can then be redefined in terms of real-time behavior.*



PATCH EDITOR: *A sample patch editor (CZ-1000) of the type that will be included with the product.*

COMMODORE AMIGA™

microillusions™

17408 Chatsworth St., Granada Hills, CA 91344
Inside CA 818/360-3715 •
Outside CA 800/522-2041
FAX 818/360-1464

One good thing deserves another...

Sequencer Plus Mk III

$495

and another...

Sequencer Plus Mark II

$295

and another.

Sequencer Plus Mark I

$99

PC MAGAZINE EDITOR'S CHOICE

After three years of refinements, **Sequencer Plus** has become the most **powerful**, most **professional** and most **versatile** MIDI sequencing software available in the world today.

So how could we improve it? By making it both **more powerful** and **more accessible.**

Introducing, the **Sequencer Plus Series: Three new versions**, each with differing levels of sophistication offering a system that will *grow with you* instead of being outgrown.

Sequencer Plus Mark III (Sp3) is a *"No Compromise" powerhouse!* 64 music tracks with independent offsets, tempo track, song position pointer, chase mode, extensive and selectable MIDI Filtering, 17 transform functions, block editing of multiple tracks, 11 memory buffers and a list of additional features that reads like a computer musician's wish list!

Sp3 also transmits patch files created with **Patch Master**, our unique *MIDI network organizer/universal librarian* program that supports virtually every major synthesizer.

Sequencer Plus Mark II (Sp2) is the perfect production tool for just about all of your most demanding musical creations. 32 music tracks, tempo track, 6 memory buffers, programmable play range, Song position pointer, intricate MIDI data editor, individual track load/save, and more!

Sequencer Plus Mark I (Sp1), at $99.00, is the *least expensive PC sequencer on the market* and is the perfect way to ease into computer music. 16 tracks, each with cut and paste operations, merging, looping, quantizing, transposition, independent MIDI channel, MIDI program and solo/muting! Complete freedom of time signatures and polyrhythms, 3 memory buffers, extensive punch-in capabilities and more! *Enough music-making power to satisfy even a seasoned pro!*

The **new Sequencer Plus series** is the choice of demanding musicians, composers, producers, and recording studios around the world- backed by a company with over a decade of experience in producing quality electronic music products. Best of all, it doesn't demand that you be a computer programmer, just a musician...and isn't *that* a relief!

For a free catalog and the name of your nearest Voyetra dealer, please call or write.

VOYETRA *Computer Music Products for the IBM PC/AT and Compatibles*

Voyetra Technologies, Dept E, 426 Mt. Pleasant Ave, Mamaroneck, NY 10543 (914)-698-3377

IBM PC/AT is a registered trademark of International Business Machines Corporation.

VOLUME 2, NUMBER 4
NOVEMBER 1987

THINGS TO COME

WELL, HERE WE are again. This month's roundup of reviewed instruments covers the range: MIDI converters and controllers, sequencers and samplers, big and small. Where will we find the time to use all these devices, anyway? Perhaps some of us *won't* find the time, and that – if we want – is OK. As most of us find out at some point or another, it can be unhealthy to let equipment distract us from our goals. First-hand experience is usually convincing enough, but there are enough examples of the adverse effects of excess technology around to save us the trouble.

It helps no end to have our intentions clear in our own minds, of course, but it can still be hard to figure out whether new technology represents more ways for us to do what we set out to do in the first place, or whether it represents a detour. If we come across a detour, we have to decide whether or not we like where it leads.

Everyone develops their own method of weeding through the options that technology provides, but since most people aren't *born* with multiple MIDI synths, effects, and recorders, we end up figuring it out as we go along. Each new bit of information gives us more paths to follow, but whether we choose to go in a new direction is often unpredictable – *fortunately*. This is what keeps the process of making music the challenge that it is. The day that I honestly believe I've seen *every* production technique will be the day I give up trying.

Of course, this isn't to say that it's in everyone's cards to try everything simply because they *can*. Something called atmosphere has to pull all the separate components – the people, the place, the instruments – together. After all, you want your experiments to pay off at some point. This month, our interviews with Kitaro and Daniel Lanois show two artists who emphasize this point. Setting up the right atmosphere is essential in order to channel creativity and capture it in recordings. High tech or low tech – it doesn't matter which way you choose to go if it feels right. The technology doesn't *have* to get in the way, but, unless you strike the right balance, you *can* have too much technology.

Too much technology is *not*, however, the reason why my name will not appear at the bottom of the editorial in next month's Music Technology. It has been a great deal of hard work and fun to edit this magazine for the last year and a half, but the time has come for a change and a challenge, and it's one I look forward to – as Editor of Music Maker's latest magazine, Home & Studio Recording. It's certainly not as though I've lost all interest in the equipment and musicians appearing in *these* pages; it's just that I'll be concentrating on the recording side of life. I may even get around to making a few cassette dubs of my own tunes in the process as well, who knows.

As far as the direction of MT is concerned, you can look forward to hearing even more from Bob O'Donnell in coming months; he'll be taking over as Editor of MT starting next issue. It has been great working with Bob in the months past, and I'm sure you'll be pleased with the direction he'll be providing MT in the future.

And now, on with the show . . . ■

Rick Davies

MUSIC TECHNOLOGY *is published monthly, 12 times per year, for $35.40 per year, at:* **Music Maker Publications, Inc., 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 and at additional mailing offices. Postmaster: Send address changes to: Music Technology, 7361 Topanga Canyon Blvd, Canoga Park, CA 91303.

EUROPEAN OFFICE: *Music Maker Publications Ltd., Alexander House, 1 Milton Rd, Cambridge, CB4 1UY, England. Tel: 011 (44) 223-313722. PAN: Musicmaker. (Ad Manager: Graham Butterworth)*

Yes. There is a Santa Claus.

AND, ALESIS IS PROUD TO ANNOUNCE
THE BEST
DRUM AND SEQUENCER COMBINATION EVER PRODUCED.
SEE YOUR ALESIS DEALER.

HR 16 HIGH SAMPLE RATE 16 BIT DIGITAL DRUM MACHINE

48 sounds. 16 channels.
Touch sensitive.
100 patterns. 100 songs. Full MIDI.
Exquisite sounding.

MMT-8 MULTI TRACK MIDI RECORDER

8 Tracks. Full MIDI.
100 parts.
100 songs.
Super musical.

Merry Christmas and Happy Musical Holidays from...

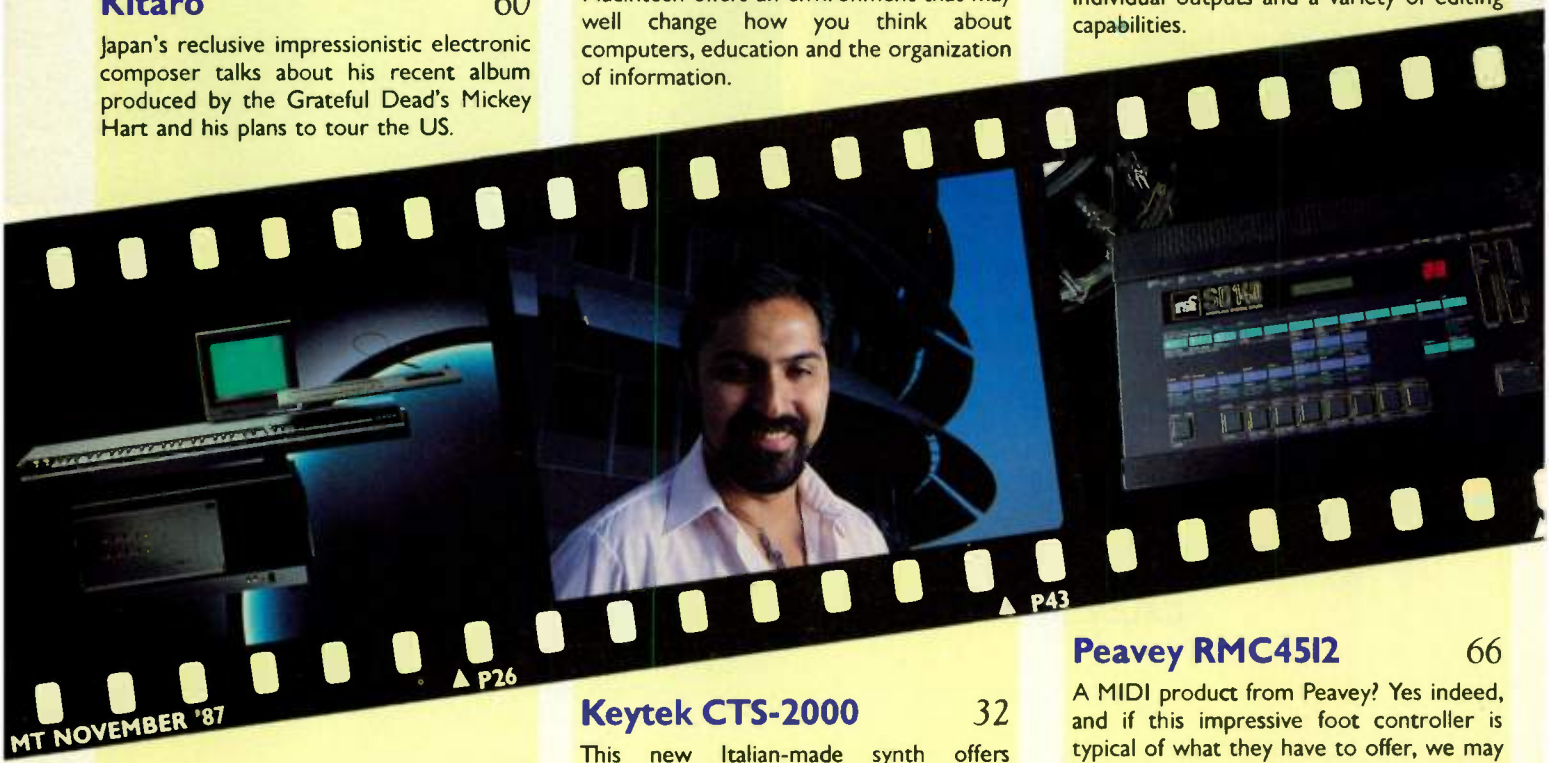
ALESIS

IN THIS ISSUE

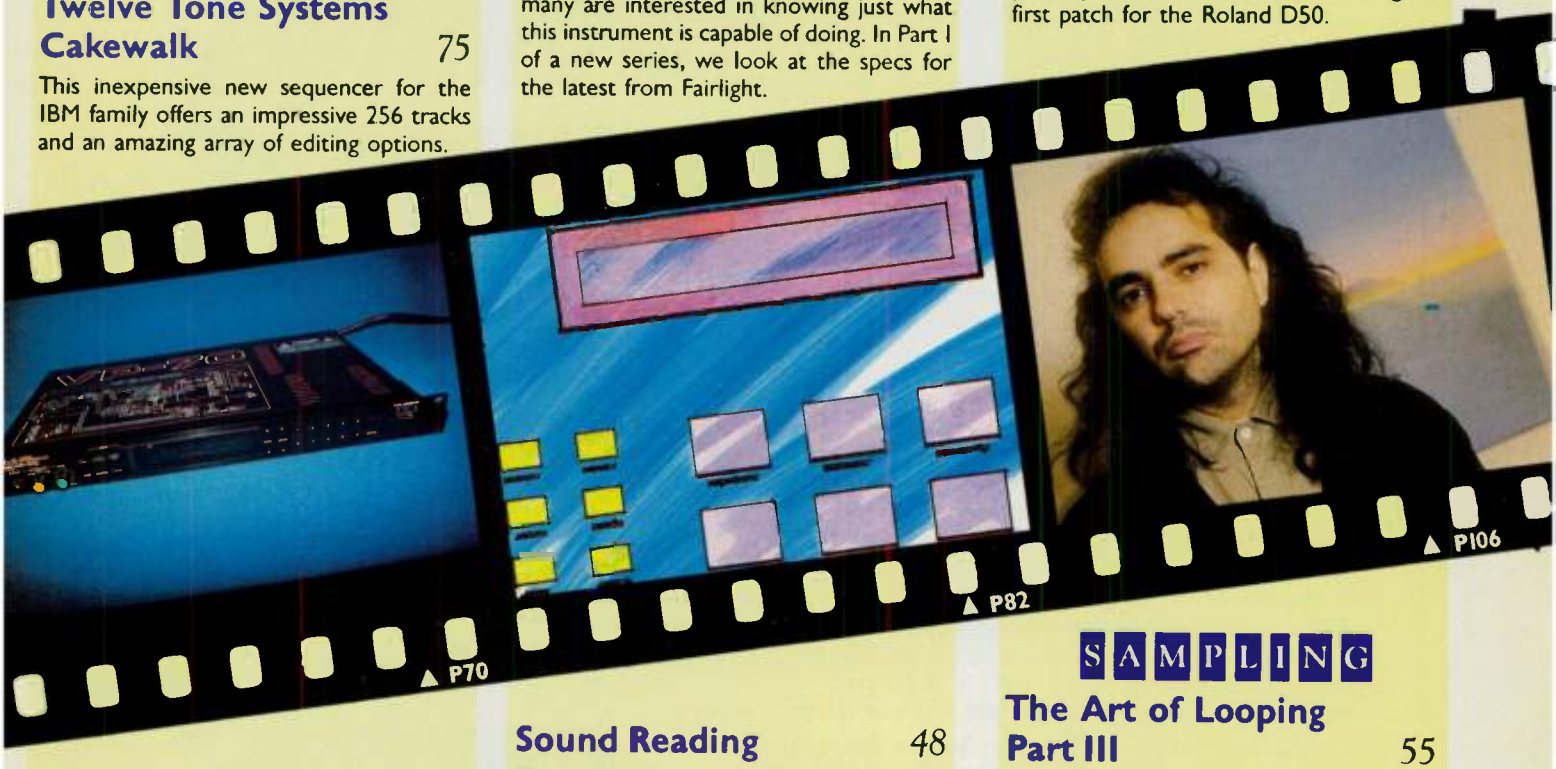
CONTENTS

MUSIC

REVIEWS



CONTENT



The music industry chose the Ensoniq ESQ-1 as the Most Innovative Keyboard of 1987 ...over the Yamaha DX7II, the Roland D-50 and Korg DSS-1

Each year, members of the music industry are polled for their choices of the most innovative instruments in a number of categories*. For the last 2 years the award has gone to Ensoniq. This year, the Ensoniq ESQ-1 was chosen for its great sound and versatility over some pretty significant competition.

There are good reasons why the ESQ-1 is at the head of the keyboard class. First there's sound—always the top criteria in evaluating a musical instrument. The ESQ's 3 digital oscillators per voice and choice of 32 different waveforms give you an unmatched pallette of tone colors.

If you'd rather just plug it in and play, there are thousands of great sounds available on cartridge, cassette and disk from Ensoniq and a number of other sound developers.

The ESQ-1 is also the only synth in its class to feature an on-board 8 track MIDI sequencer with functions that rival many stand-alone units. And because the ESQ-1 is multi-timbral with dynamic voice allocation, it can play a completely different 8-voice sound on each sequencer track.

In fact, the more you're into MIDI, the better the ESQ performs. With velocity sensitivity and full use of all the MIDI modes, it's one of the most popular central instruments in MIDI studios around the world.

Great sound and versatility. The award-winning Ensoniq ESQ-1 and the new ESQ-M Synth Module. Only at your authorized Ensoniq dealer.

For more information and the name of your nearest dealer call: **1-800-553-5151**

*The award we're referring to is the 1987 Music & Sound Award for Most Innovative Keyboard/Synthesizer.

ensoniq®

ENSONIQ Corp, Dept. T, 155 Great Valley Parkway, Malvern, PA 19355 ☐ Canada: 6969 Trans Canada Hwy., Suite 123, St. Laurent, Que. H4T 1V8 ☐ ENSONIQ Europe BV, Domplein 1, 3512 JC Utrecht, Holland ☐ Australia: Electric Factory, 188 Plenty Rd., Preston, Vic. 3072 ☐ ENSONIQ Japan: Ochanomizu KS Bldg., 2nd Floor #201, 3-3-1 Hongou Bunkyo-Ku, Tokyo

NEWS DESK

YAMAHA SAMPLER!

The inevitable has finally occurred; Yamaha has jumped into the sampling fray feet first. As we were going to press we received some preliminary (ie. it *might* change) information on their new rack-mount, 12-bit linear, stereo sampler and couldn't resist passing on the information.

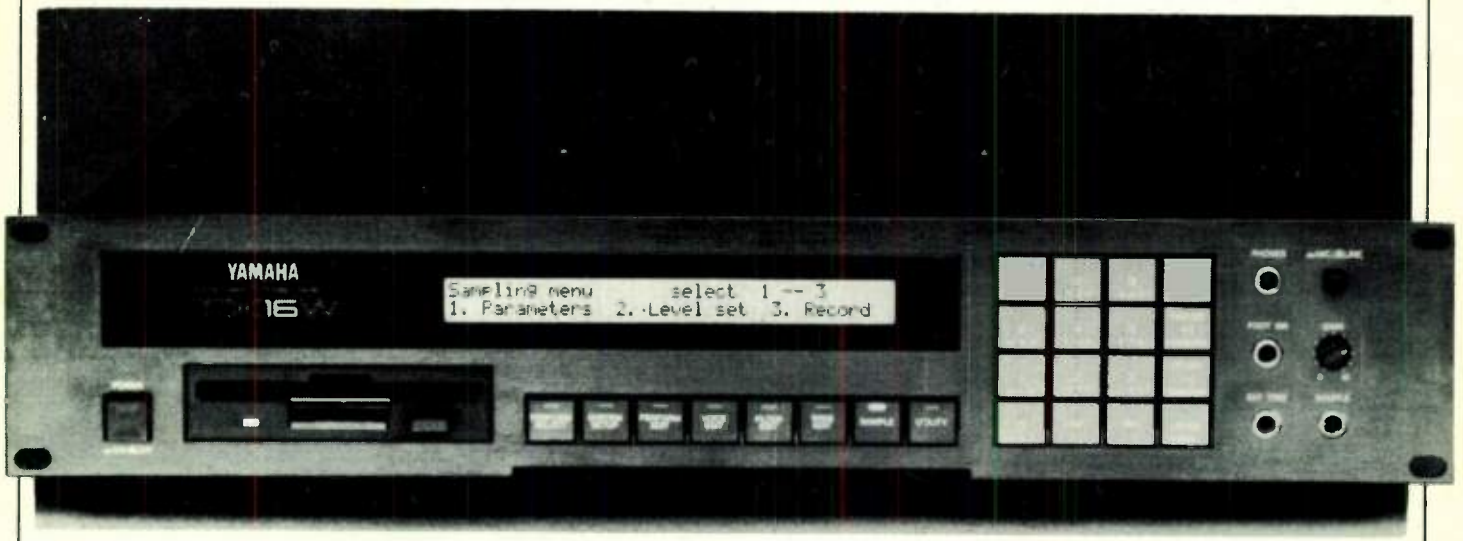
The TX16W Digital Wave Filtering Sampler – its official name – features 1.5Mbytes of RAM standard with the possibility of adding three optional 1.5Meg EMI memory expansion boards. The quoted sampling rates and times for a standard instrument are as follows: up to 16.3 seconds of mono sampling at the 16.7kHz rate, 7.9 seconds of mono or stereo sampling at 33.3kHz and 5.2 seconds of mono sampling at the maximum 50kHz rate. Now these times seem a bit odd (the same amount of time for mono and stereo sampling?), but they are the quoted *preliminary* specifications which Yamaha sent out. The sample input is a single stereo phone jack on the front panel.

The TX16W is 16-voice polyphonic, can play up to 16 different Voices at once and features eight individual outputs and a pair of stereo outs. The unit also has an RS422 port for external computer editing capability and data transmission to hard disk or other storage media.

The instrument's voice structure can be divided into 64 Performance Memories, 32 Voice Memories, 64 Timbre Memories and 32 Filter Memories. Speaking of which, one of the TX16W's unique features is its Dynamic Digital Filter, which offers real-time digital control over the sounds. The unit also features Key and Velocity Crossfade (up to 16 per key) and up to 32 split points per voice. All voice and function storage is done on 3.5" floppy disks and the 80-character (40X2) back-lit LCD provides graphic displays of some of the more important parameters.

While a definite retail price has not yet been set, the TX16W will list for under $3000.

MORE FROM Yamaha Music Corporation, Digital Musical Instruments Division, PO Box 6600, Buena Park, CA 90620. Tel: (714) 522-9011.



DR. T'S IN A NUTSHELL

Dr. T's Software has announced an October release for an impressive upgrade to their KCS sequencer which will provide a multi-program environment for other Dr. T's programs. Until now, using different programs meant quitting one before accessing another; with the MPE feature, however, ST owners will be able to select the sequencer's "external" option and load four programs at once. Among other things, this will allow the sequencer to record real-time patch edits carried out from an editor program.

The MPE will be incorporated into the KCS 1.5I which should be available in early November for $249. Registered owners of version 1.0 or 1.5 should contact Dr. T's for upgrade information. Plans are in the works to port the program over to the Macintosh and Amiga computers as well.

MORE FROM Dr. T's Music Software, Inc., 220 Boylston Street, Suite 306, Chestnut Hill, MA 02167. Tel: (617) 244-6954

MUSIC PUBLISHING FOR ATARI

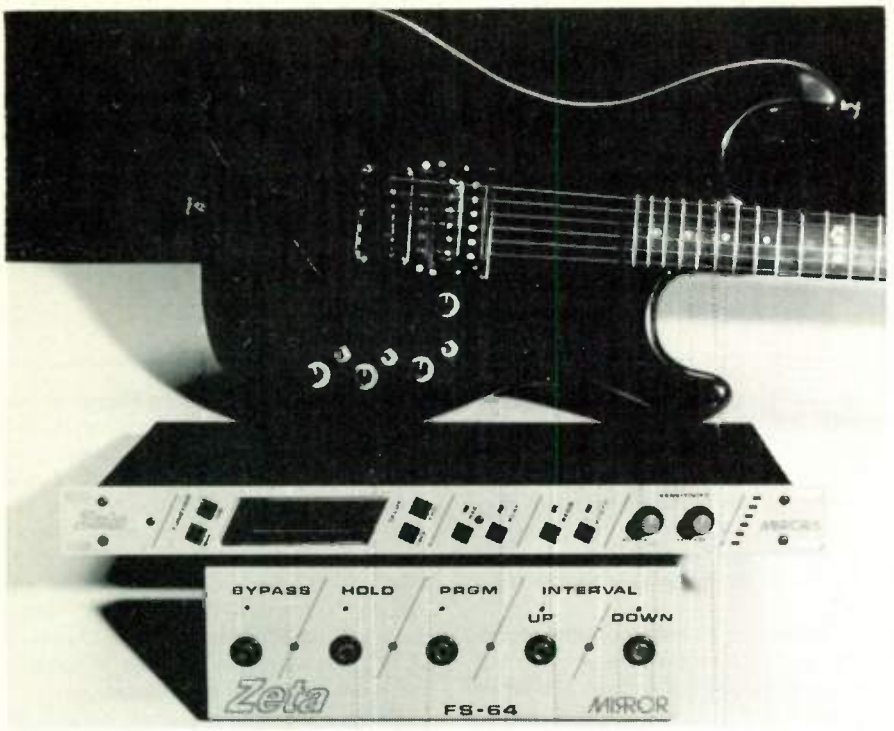
Hybrid Arts is offering EZ-Score Plus, a desktop music publishing program for the Atari ST. Features include an Auto-Score function which allows you to convert song files from sequencer files; score creation through the mouse, computer keyboard or MIDI information; variable stave spacing; and the ability to add lyrics. All standard editing features are included.

Files are upwardly compatible to MIDI-Score, the Hybrid Arts scoring system. The suggested retail price of the program is $99.95.

MORE FROM Hybrid Arts, Inc., 11920 W. Olympic Boulevard, Los Angeles, CA 90064. Tel: (213) 826-3777

HEAVEN CAN'T WAIT

The New York Yamaha Users have announced the opening of Y.C.A.M.P. (Yamaha Computer Assisted Musician's ▶



ZETA GUITAR CONTROLLER

Zeta Music Systems have announced the probable availability of their Mirror 6 MIDI Guitar Controller in November. The Mirror 6 will incorporate fret position scanning plus pitch extraction, allowing it to be used as a MIDI controller, electric guitar, or both simultaneously. Zeta Music claims that commonly-used guitar techniques such as pitch-bending, hammer-ons and pull-offs can be successfully transferred to the synth being controlled.

The Mirror 6 consists of an interface rack unit, footswitch and MIDI-equipped guitar. Zeta is reportedly working on arrangements with popular guitar manufacturers to produce Zeta-equipped versions of favorite guitars. While firm pricing information is not yet available, it is projected that the systems will run between $3500 and $4000.

MORE FROM Zeta Music Systems, Inc., 2823 Ninth Street, Berkeley, CA 94710. Tel: (415) 849-9648 or (outside California) 1-800-622-MIDI

► Paradise), an on-line service for musicians. Y.C.A.M.P. supports all computer types (ST, Amiga, IBM, CX5M, etc) and products and is available via modem (300/1200 baud) at (516) 295-0823.

MORE FROM Y.C.A.M.P., Suite 22-B, 551 Central Avenue, Cedarhurst, NY 11516. Tel: (516) 295-1427

BLANK BLOOMING

Blank Software has announced upgrades to two of their existing programs, Sound Lab for the Amiga and Drum File Version 1.1.

Sound Lab allows all MASOS (Mirage Advanced Sampler Operating System) functions to be controlled from the Amiga and provides a graphic presentation of looping, keyboard assignment, memory allocation and voice program editing. Audio processing functions are also available, such as compression and interpolation, which are not available on the Mirage alone. It is available for $299.95.

Also announced is Version 1.1 of Drum File for use with the E-mu SP12 and Macintosh II and SE. Drum File provides MIDI transfers of sound and song information between the Mac and SP12, song file management and Sound Designer compatibility. New functions include auto-sort library windows, storable MIDI configuration, and printer output of all libraries, drum documents and internal mixes. It's available as a free upgrade to owners of version 1.0.

MORE FROM Blank Software, 1034 Natoma Street, San Francisco, CA 94103. Tel: (415) 863-9224

PATCH COMMANDER UPDATE

Meico Electronics' foot-controlled patch changer, the Patch Commander, has been updated with a MIDI Merge function which allows MIDI data to be merged with the patch changes generated by the unit. The MIDI In data and Patch Change data appear on the MIDI Out connector as a single signal. The unit contains a 256-byte data buffer to prevent any loss of incoming MIDI data when a Patch Button is depressed.

The suggested retail price for the new Patch Commander is $299.

MORE FROM MEICO Electronics, 35 South Dishmill Road, PO Box 251, Higganum, CT 06441. Tel: (203) 345-3253

SONICFLIGHT SOFTWARE SOUNDS OFF

MIDImouse Music has announced the release of two programs for the Atari ST: the Matrix 12/Xpander Librarian and Fast Tracks ST.

The Matrix 12/Xpander Patch Librarian allows patch data to be loaded or sent to the instrument, saved on disk, examined, and transferred from file to file. The structure allows an entire data dump (all 100 single and multi-patches) file in the computer at a time. Once in the computer, single patches may be transferred between

DYNACORD DRUMS TO MIDI

A new electronic drum converter, the P20, has been introduced by Dynacord. In addition to drum-to-MIDI conversion, the P20 features on-board digital drum and percussion sounds. The memory section of the unit holds 100 user programs that are programmable for individual channel sound source selection and volume level, as well as MIDI mode selection, channel and note assignment, note stacking, variable dynamics-to-pitch relationships and inverted dynamic triggering. Also programmable are six digital delays, each with independently adjustable echo rates and decays.

The P20's 24 internal sounds include bass drum, snare drum and tom-tom samples along with handclap, cross-stick, hi-hat and cowbell. External sockets allow any of the Dynacord Digital Sound Modules to be used, and the MIDI implementation provides access to other MIDI sound sources.

The suggested retail price is $2449.

MORE FROM Drum Workshop, 2697 Lavery Ct. #16, Newbury Park, CA 91320. Tel: (805) 499-6863

files and the synth, and entire files may be dumped.

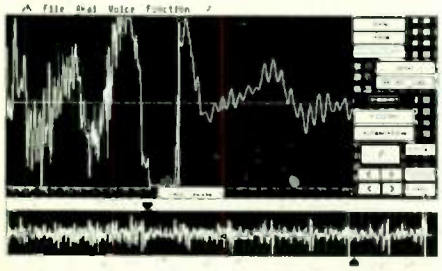
Included on the disk is a set of 100 performance sounds that were created on the Matrix 12. The retail price is $49.95 plus shipping.

MORE FROM MIDImouse Music, Box 272, Rhododendron, OR 97049. Tel: (503) 622-4034

DRUMWARE SOFTWARE FOR S900

Atari ST owners who need a program to sculpt and save their S900 samples may be pleased to learn of the Soundfiler S900, a new editor/librarian from Drumware.

The Wave Editor includes visual looping,



waveform display, waveform drawing and clipboard "cut and paste" style editing with phase inversion, waveform reversal, mixing and merging.

The Digital Signal Processing elements include crossfade looping, digital enveloping, gain normalization and state variable digital equalization.

The retail price is $299.

MORE FROM Drumware, 12077 Wilshire Boulevard, Suite 515, Los Angeles, CA 90025. Tel: (213) 478-3956

THE QUEST FOR MIDI

Music Quest, Inc, has announced the MIDI Starter System, designed for the novice musician and the computer-music hobbyist. The system includes a MIDI interface card, sequencer, and editor/librarians for the Casio CZ and Yamaha DX21/27/100 synths for use with IBM PCs or compatibles. The interface card is compatible with the Roland MPU401 and Voyetra OP4000/4001 MIDI interfaces and supports other popular software packages.

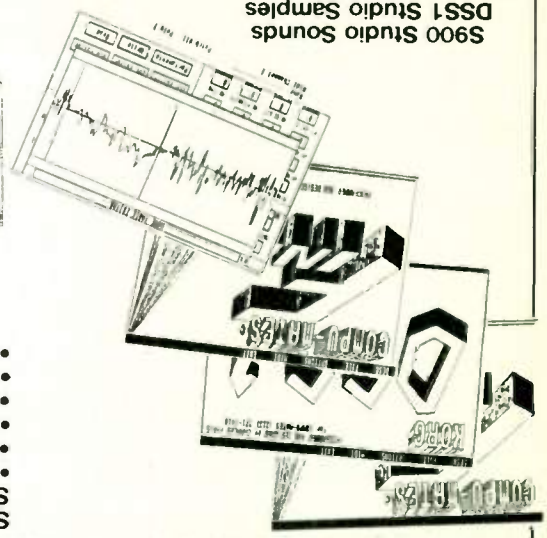
Shipping is expected in early November. The introductory price for this package is $199.

MORE FROM Music Quest, Inc., 1700 Alma Drive, Suite 260, Plano, TX 75075. Tel: (214) 881-7408

AMIGA SUPPORT FROM SOUNDQUEST

Sound Quest has entered the music software field, exclusively supporting the Commodore Amiga. Their initial product line includes the DX Master Editor, The DXII Master Editor/Librarian, and the Master Librarian; their D50 Master Editor/Librarian will soon be released. All programs will be distributed by Dr. T's Software.

The DX Master Editor stores and edits DX7, DX9 and TX data files, offering▶



BBE SINGLED OUT

Barcus-Berry Electronics has announced availability of the BBE 401, a single-channel version of the 802 and 402 harmonic phase compensators. It is designed to add brightness without equalizaton and to reduce mid-range smear and associated masking effects for vocalists and acoustic instruments using microphones.

The BBE 401 acts as an active direct box so that high-impedance microphones, guitars, basses or synthesizers can be plugged directly into the mixer's low impedance input. Conversely, the BBE 401 matches a low impedance microphone to a high impedance input and provides the +24 volt phantom-power needed by condenser-type microphones. It also acts as a mic pre-amp for amplifying microphones to drive −10dBu line levels.

The unit can also be used for live performance, inserted between the house mic and the house mixer, in order to improve the clarity of vocals without disrupting the wiring installation. It can be used with any microphone and any sound system.

The suggested price is $229.

MORE FROM Barcus-Berry Electronics, Inc., 5500 Bolsa Avenue, Suite 245, Huntington Beach, CA 92649. Tel: (714) 897-6766

▶ mouse or keyboard editing facilities and a controllable random voice generator. The DXII Master Editor has the additional capability of storage and editing of 13 types of DXII Sys Ex data and adds screens to control fractional key scaling, micro-tuning, performance memories and system setup. The Master Librarian allows storage, retrieval and editing of Sys Ex data from most MIDI equipment, and provides a space for notes and keywords for disk searches.

MORE FROM Dr. T's Music Software, Inc., 220 Boylston Street, Suite 306, Chestnut Hill, MA 02167. Tel: (617) 244-6954

MIDIWORLD SAVES T8

A new librarian program for the Commodore 64 with a Sequential 242 cartridge and a Prophet T8 is available from Stanley Jungleib's MIDIWORLD. Though long out of production, many keyboard players still appreciate the sounds they can get from the T8. Although Sequential released the Model 900 dump utility for the Commodore 64 and 242 cartridge which stored complete sets of T8 programs, there has not been a librarian to allow naming, saving and loading individual programs – until now. The T8 program also includes a random patch generating utility, which comes in handy for overcoming "programmer's block."

The price for the T8 package is $35, which includes the disk, two sample programs, instructions, handling, first class shipping anywhere and guaranteed satisfaction. California residents add 7% sales tax.

MORE FROM Stanley Jungleib's MIDIWORLD, 3051 North First Street, San Jose, CA 95134. PAN: STANLEYJ

INTERNAL REVERBERATION

It's unusual these days to see a "first of its kind," but HME has come up with one – the first reverb mic with reverb circuitry built into the mic itself. The RM77 has an adjustable control that permits varying the amount of reverb and a three-position switch providing mute, echo, or normal capabilities. A built-in "pop" filter is designed to minimize external interference

commonly heard in condenser mics. The RM77 comes standard with a three-pin XLR cable, mic clamp, vinyl bag and gift box.

The suggested retail price is $144.

MORE FROM HM Electronics, Inc., 6675 Mesa Ridge, San Diego, CA 92121. Tel: (619) 535-6092

"It's like a dream machine."

PETER CHAIKIN,
ENGINEER/PRODUCER

Producers and engineers are calling it a dream machine.

We call it the DMP7 Digital Mixing Processor. It's a new kind of creative tool and there's never been anything like it.

For the first time, you can command true digital mixing and dramatic digital effects. And have them programmed into memory for instant, flawless recall.

All from one incredible, affordable, simple-to-use 8x2 console.

All MIDI-controllable. Remote-controllable. And syncable via MIDI to SMPTE.

It makes unforgettable music.

The DMP7 takes a "snapshot" of all of your settings. Instead of developing writer's cramp after each session, you can digitally store your production notes into memory. Up to 97 snapshots (with RAM4 cartridge), each consisting of more than 200 parameters, from fader positions to effect settings to 3-band parametric EQ settings.

Then recall them instantly at the touch of a key, or with an external MIDI controller or sequencer.

Yamaha introduces i
event programmer, a
All at

All the colors are built in.

The DMP7 has three on-board digital signal processors that deliver spectacular effects like stereo chorus and echo. Panning. Four kinds of reverb, and reverb plus gate. And up to three effects can be mixed simultaneously.

There are 18 different effect types in all, with up to nine parameters for each. So you can create precisely the sounds you want, while saying goodbye to outboard gear and their multiple data conversions. The DMP7's mixer has eight input channels, and its digital cascading feature lets you connect additional DMP7s to add more inputs (up to 32), as you require.

And build yourself the ultimate digital console.

There's a lot more you should know about the DMP7, and your Yamaha Professional Audio dealer can tell you the whole story. See him this week.

And believe in dreams.

s newest digital mixer, nd digital processor. once.

Here's the setup.

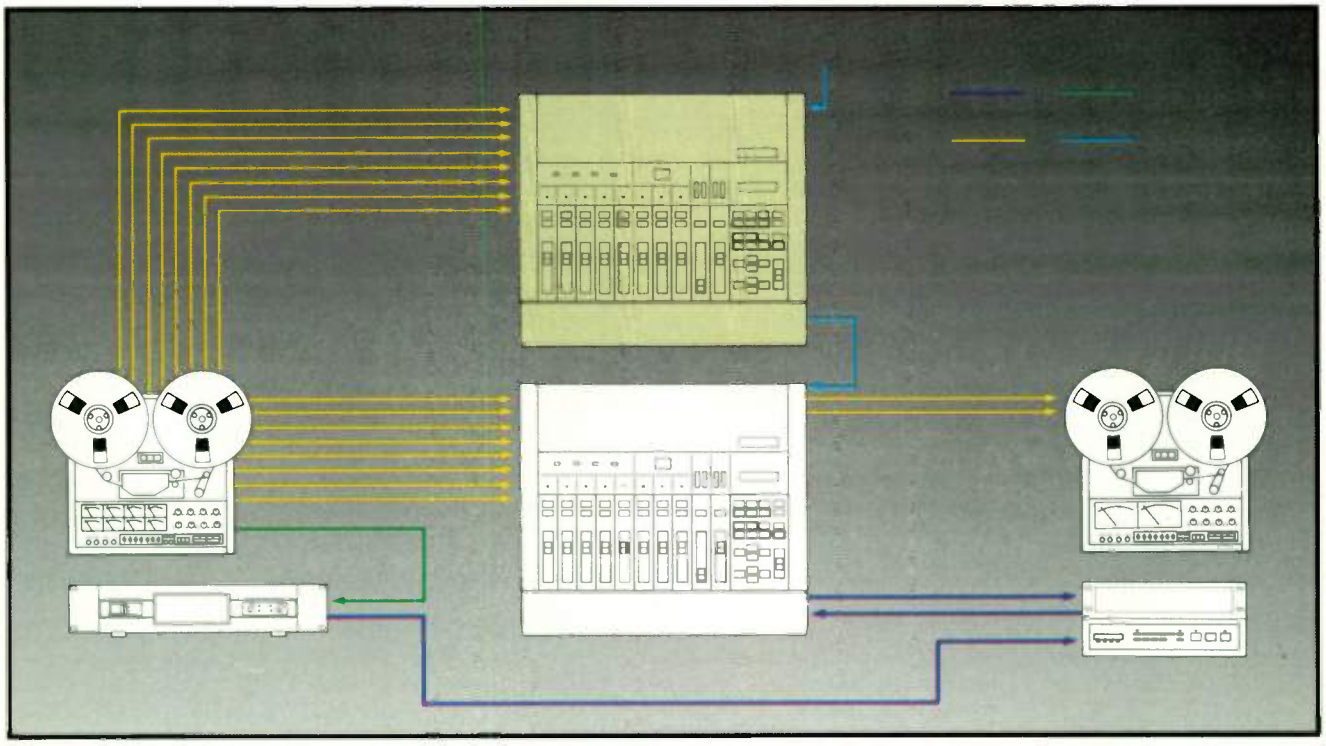
The simplicity of the diagram below belies the many capabilities of the DMP7 at work in an automated multi-track mixdown/processing system.

Dry tracks from the multi-track recorder are fed into the DMP7. A SMPTE track is fed into the SMPTE/MIDI converter. The MIDI timing track controls the MIDI sequencer/recorder.

The MIDI control information for the DMP7 can now be recorded in sync with the music. The DMP7 then automatically mixes everything down into a two-track master or demo. Until now, this level of mixed automation was only possible with more expensive consoles.

The DMP7 doesn't just do the final mixing, but the final processing as well, to each individual track. With reverbs, flanging, delay, and stereo panning. Or whatever sweetening you need. And again, in sync with the music.

What's more, if you have more than eight tracks, you can cascade in the digital domain to another DMP7 for 16 tracks. Or another for 24. And yet another for 32 tracks.



YAMAHA®
Engineering Imagination™

Yamaha Music Corporation, Professional Audio Division, P.O. Box 6600, Buena Park, CA 90622.
In Canada, Yamaha Canada Music Ltd., 135 Milner Avenue, Scarborough, Ontario M1S 3R1.

READERS' LETTERS

Send any question or comments that you may have to: Reader's Letters, Music Technology, 7361 Topanga Canyon Blvd., Canoga Park, CA 91303.

Dear Music Technology,

Sequential greatly appreciates your preview of the Prophet 3000 stereo sampling system. Unfortunately, due to some errors on our part, we would like to ask you to print the following three corrections.

At the 48kHz sampling rate, the standard memory (two Mbytes) provides 10.6 seconds of stereo sampling, not 49 seconds. On the other extreme, at 16kHz in mono, the maximum available time would be 63.6 seconds. One expander is the maximum and a Prophet 3000 will overflow up to eight voices to its expander, for a total of 16 voice polyphony. If desired, however, multiple expanders can be tied together to form separate 16-voice systems, and receive sample data from the same hard disk unit. This allows you to keep all of your presets in one place.

There are output jacks for each of the eight voices. Voices cannot be assigned to different output jacks but presets can be assigned to different voices.

We apologize for any inconvenience to you or your readers.

Stanley Jungleib
Sequential

Dear Music Technology,

I would like to comment on a few things in the August '87 issue. On page 50 in the NAMM article, Dan Goldstein said that the term "CD quality" was "mildly redundant." The dictionary definition for redundant is as follows: 1) Using more words than necessary, ie. superfluous. 2) Something similar or repetitious or a duplicate, which means that he was saying that "CD" and "quality" mean the same thing. But then he contradicts himself by saying that some CD players sound great and others sound terrible and that 16-bit doesn't guarantee great sound quality. It seems to me that what Dan was trying to say is that the term "CD quality" is either ambiguous or it is presumptuous. Am I right?

Next, a great deal was said in both the NAMM Report and in the article on page 75 about the Kawai K5 giving real-time additive synthesis power to musicians, but nothing was said about another instrument from a Canadian company called Lyre who demonstrated real-time additive synthesis hardware and software at NAMM. This instrument (the FDSS) has separate envelopes for pitch and amplitude for each of its oscillators. It is available for the IBM PC and soon will be available for the Mac. Each harmonic can be modulated in real time from MIDI through velocity, pressure, mod wheel, etc., giving it real-time expressiveness like the DX7.

Because it is a true real-time synthesizer, it also means that envelopes are uniform all the way up and down the keyboard, unless programmed differently. The additive synthesis capabilities in SoftSynth, on the other hand, are nothing more than software taking additive formulas and producing a wavetable for a sampler. As a result, it suffers from the same problems that samples do when notes too far away from the root are played. A true real-time additive synthesizer, however, does not have this problem.

Now if I sound like a sales rep or an advertisement, it is because I heard it for myself and also have a demo tape. What that synthesizer could do with only 16 harmonics was amazing. Imagine synthesis with the advantages of real-time control and the realism of a sampler; that is what the Lyre synthesizer sounded like.

Personally, this is where I feel synthesis is heading. Or should be. In fact, because of the Lyre, I am not too excited about the abbreviated capabilities of the Kawai K5. Would you please do a thorough report on this amazing package from Lyre?

William H. Roberts
Indianapolis, IN

As you may have seen, last month's Newsdesk featured some preliminary information on the Lyre products and you can certainly expect to see a full review of them within these pages in the near future.

Dear Music Technology,

Robert Rich's article on Just Intonation was just amazing! I have been enlightened. I read the article the day after I listened to "The Beauty in the Beast" by Wendy Carlos, and it all came together.

In the further reading section, where can I purchase these books? Also, does Mr. Rich mean Revision 3 when mentioning the Prophet 5 (what year would that be?)

A million thanks to Music Technology for being light years ahead of all music/technology magazines.

Harold Terrell
Bookler, CO

Robert Rich responds: "Thanks for the feedback. The books I listed in the further reading section should be available in most libraries with a decent music section (but note the correction listed below). The Prophet 5's I referred to are indeed Rev. 3.0 through 3.3, which start at serial number 1300 around 1980.

I would also like to correct some mistakes that entered into my article during the editing process. First of all, Harry Partch's* Genesis of a Music *was published in 1949, not 1979, which is the date of the paperback edition. Partch died in 1974.

Also, in the third column, the fifth is termed "the fifth note in the scale." This gives the misconception that a scale is a fixed entity. Even if the fifth did happen to be the fifth note in a scale, there are any number of other scales where it is not!

Another minor point is that Bach's "Well-Tempered Clavier" was written for well-temperament, which is slightly different from the scale we now call equal temperament. Hopefully those are the only errors!

For anyone interested in corresponding, my address is: Robert Rich, Soundscape Productions, PO Box 8891, Stanford, CA 94305.

Dear Music Technology,

I was madder than someone who just paid full retail for an Emulator II+ HD, only to find out about the Emulator III, when I read your article/interview with Laurie Anderson. I'm a member of D.A.M.M.A.D. (Drunks Against Mad Mothers Against Drunks), and as such, I like to drive around, down a case or so of Schmidt 16oz

beer, and listen to music. Well, I made the mistake of listening to Anderson's "United States" while I was drinking and driving one Sunday morning, and I almost drove into the Whapeton, ND high school girl's track team and precision baton troupe bus. That music is dangerous! I think Laurie Anderson would be better off if she MIDI'd an accordion to a Doberman and played "In Heaven There Is No Beer."

And just where does someone who composes pieces such as "America" and "Home of the Brave" come off berating true American music – the military band marches. On page 66 of MT August '87, Ms. Anderson states, "The real-life 'Home of the Brave' is modern-day America, where the national budget for military marching bands is larger than that for the arts . . . It's incredible. That's a lot of band uniforms and sheet music."

Yes, Ms. Anderson, that is a lot of band uniforms and sheet music, but there are a lot more people in the Army Marching Band, than there are in yours! Do you have any idea how much the cost of braid-work has escalated in the past year? Well, it's gone up a lot! And how about those hats? Do you think you can just trot on down to the thrift store and pick one up next to the day-old bread? If you think so, you're as wrong as Sousa playing Hank Williams Junior.

I must also take offense with some of Ms. Anderson's actions. She took a "little cheap drum machine apart because it was broken." She then sewed the working triggers, and triggers from other drum machines into a suit which she could play. Well, does anyone out there remember that song, "Look for the union label?" First, there are qualified repair technicians out there who could have fixed the drum machine, and even if she wanted it sewn into a suit, there are union tailors and seamstresses whose entire livelihood is sewing. Well Laurie, remember Norma Rae? Think she's happy about this? I doubt it. And I'm certainly not going to talk about the auto-erotic implications of playing a suit, but if someone doesn't stop pretty soon, someone is going to go blind.

Ernie Existential
Billings, MT

PS: I'm going to send a copy to Optical Media. Maybe they can help Laurie if/when she does go blind.

Ernie, we find your concern and compassion heartwarming. Thanks for speaking out against this disturbing trend.

Dear Music Technology,

Your review of the Bacchus TX81Z Graphic Editing System in the September '87 issue was a little confusing. Michael Stone repeatedly compares the program's graphical interface to that of the Apple Macintosh. He calls it a "mouse-and-icon based graphical interface familiar to Macintosh and ST users" and says that if you look only at the mouse and screen "you'll swear you're running on a Mac." He concludes the article by referring to the program's "wonderful Mac-clone user interface." However, he also mentions Bacchus' claim that the interface is "not a Mac clone" but uses "the best features of the Apple, Sun, and Xerox PARC user interfaces."

It seems to me that there are two issues here. First, how the program looked on the screen, and second, how the program worked from the user's point of view. As for the look of the program, I saw it at the June '87 NAMM Show running on both a Hercules monochrome display and on a Wyse vertical display. Only on the Wyse display, which costs over $1000, did the quality of the screen graphics approach the sharp resolution of the Macintosh screen.

As for the feel of the program, it seemed to me, a Mac user of over three years, only superficially like the Macintosh's user interface. It does have windows and a mouse, but most Mac users, especially novices who have just gotten used to the Mac, would find this program confusing. There is no menu bar at the top of the screen, but instead an "icon bar" at the bottom. Rather than give a textual overview of the program as a menu bar does, the "icon bar" requires the user to puzzle out exactly what each picture means before grasping the overall structure. The most important difference is that the Mac interface guidelines provide for an Undo function in every program. (To be sure, it is not always implemented, but most of the best programs have it.) Finally, the multi-button mouse means that the user must at first remember which button does what. This means it will take much longer for the point-and-click operation to become automatic for the user.

It is easy to call a graphical interface "Mac-like" or even "a Mac clone" without evaluating which differences will go unnoticed by the user and which differences will confuse him or her. My sense is that the folks at Apple spent a lot of time and effort refining a user interface that is easy to use and consistent across most Macintosh applications. It is always dangerous for a company to create a hybrid interface, because without sufficient testing, it may end up only being comfortable for its authors.

Besides this sticky point, I found the article very interesting and informative, as is your entire magazine.

David S. Bogartz, Editor
Soundwaves
The Boston Computer Society's music and computers newsletter

Dear Music Technology,

I just got my first issue from a subscription and already I'm writing!

In reference to the "Sounds Natural" series, I want to know how I could obtain the book you were covering – "A Synthesist's Guide to Acoustic Instruments."

About the magazine, I've been enjoying the up-to-date coverage on products and hints on applications. I expect I'll be enjoying this subscription.

Scott Foxx
Garner, NC

Thanks Scott. "A Synthesist's Guide to Acoustic Instruments" is available in many music stores all over the country or, if you can't find it, direct from the publisher: Music Sales Corp., 24 E. 22nd St., New York, NY 10010. Tel: (212) 254-2100. ■

RACK ATTACK

Buy any two Alesis Micro Series Signal Processors and get a free Micro Rack Adapter. Ask your Alesis Dealer for details.

FOR THE ABSOLUTE BEGINNER TO THE ABSOLUTE PROFESSIONAL ...THE ABSOLUTE BEST

In the few months since the Alesis Microverb was first released it has established itself as the new **digital reverb standard** in personal and professional music studios around the world.

Bold statements need factual support. How about these facts:

The design of Microverb is based on years of exhaustive research into the phenomena of spatial acoustics so that the sound of Microverb rivals even the most expensive digital reverbs. And Microverb has been nominated for the 1987 Technical Achievement and Creativity Awards of MIX Publications...other products nominated cost as much as 40 times more than Microverb! What's our secret? The heart of Microverb is an Alesis designed custom VLSI signal processing microchip using 16 bit linear PCM encoding with 90dB dynamic range. In other words, Microverb is **technically superb.**

Alesis digital reverb programs are legendary for their uncolored accuracy and warmth. 6 small reverb programs, 7 large, 2 gated and 1 reverse. Plates, rooms, halls, chambers, caverns and deep space. All these sounds are available to help you create deep, rich drums, lush strings and keyboards, super funky and tight rhythm tracks, or a brilliant sheen on lead instruments and vocals. Serious users tell us our reverb programs are the acoustic foundation on which they build their mixes. In other words, Microverb is **superbly musical.**

Front panel rotary knobs control input, mix and output levels. A continuous rotary knob selects the programs. Stereo input as well as output are provided on ¼" jacks. Plus, there's a defeat jack for convenience. Microverb accepts any level from the weakest guitar or mic signal to professional +4dbv console output levels. The dry signal is conveniently maintained in stereo allowing series connection of several Microverbs when you're not using a mixer. So an ambient program can be utilized to create body for drums and instruments through the first Microverb, and a plate, hall or chamber can be used on the second Microverb for a spacious reverb with smooth decay. In other words, Microverb is **instantly and effortlessly accessible...and flexible.**

The one third rack space extruded aluminum case is 19" rack mountable with the Microverb Rack Adapter. You can rack mount one, two or three or simply set them free standing on your amp, mixer or keyboard in only 5¾" of space. In other words, **you can take this digital reverb anywhere.**

And you will. Because like every other Alesis product, Microverb is a breakthrough not only in design and performance but also in price. This means that you can afford to dedicate a Microverb to vocals, another one to drums, one for keyboards and guitars, and a spare for last minute creative urges. The sound of multiple digital reverbs (the sound you hear in every major record production done today) is now affordable to *anyone* who loves to make music. Your music deserves it.

Don't settle for anything less. You can afford the absolute best.

Microverb's suggested retail price is $249. Alesis provides free educational materials and brochures to help you get the most out of our products and music processors in general. We encourage you to take part. Ask your Alesis dealer or write to:

LOS ANGELES:
Alesis Corp. PO Box 3908
Los Angeles, CA 90078

LONDON:
6, Letchworth Business Center,
Avenue One, Letchworth,
Hertfordshire SG6 2HR

ALESIS

REVIEW

I·N B·R·I·E·F

Roland S220 Sampler

The company's latest rack-mount sampler incorporates many of the multi-timbral features found on the S50, including 16-voice polyphony, for about the price of an S10. *Review by Aaron Hallas.*

A TREND IS occurring in the music industry toward rack-mount versions of existing synthesizers and samplers that offer increased flexibility by way of added features. Since the review of the S10 sampler in MT December 1986, Roland has followed suit and, in the process, made significant headway with their line of 12-bit samplers. The sound library has grown considerably larger and the sounds, with very few exceptions, are very good to excellent. The rack-mount version of the S10, the MKS100, has been replaced with the upgraded S220. They have doubled the amount of voices available, added some new features such as four individual outputs, the ability to store two different sounds in each bank, and made it multi-timbral as well.

The S220 retains the basic sampling features of the S10 and the MKS100: 30 or 15kHz sampling rates with 4.4 seconds of sampling time at the higher rate and 8.8 seconds at the lower rate. The sample memory is divided into four banks (A,B,C,D) which can be played individually or grouped together in combinations called structures. The split structures (A/B, C/D, AB/CD or A/B/C/D) can be used for multi-sampled sounds or can contain different sounds for creating drum sets or ensembles. The banks can also be linked together (AB, CD or ABCD) for longer sample times.

Multi-timbral operation and multiple outputs alone set the new model heads above its predecessor. The ability to process sounds individually is one of the most desirable features in any multi-timbral synth or sampler and Roland has found a very flexible way to divvy up the S220's 16 voices between its four outputs in what they call the Multi Function mode. Multi Modes 1-5 allow voices to be assigned to the individual outputs as follows: as four four-voice instruments; as two four-voice and one eight-voice; as two six-voice and two two-voice; and as two eight-voice with split structures or with linked structures. The key range (highest and lowest key numbers) can be set for each bank and more than one bank can have the same MIDI channel so a split/layered effect can be had even if your master keyboard doesn't have this feature.

A Detune mode is available that combines a structure with a detuned version of itself, as is a Delay mode with programmable Delay time, level and key offset (transposition). You can also layer sounds in the Dual mode with the ability to mix or switch between the two sounds with varying key velocity. The Separate Function divides the output of the structures between two output jacks. The S220 becomes eight-note polyphonic in the Dual and Detune modes.

A new Address Velocity Switch mode has been added that allows two sampled sounds to be stored in each bank with separate start points, end points, loop lengths and loop tunings. The two different sounds are called Address Groups 1 and 2. The Address Velocity Switch sets the minimum velocity required to play one of the Address Groups. For example, closed and open hi-hats could be stored in the same bank and the closed hi-hat would be played up to a set velocity level at which point the open hi-hat would be played. This feature can effectively double the number of sounds available at any one time.

I was pleased to find a built-in arpeggiator with selectable tempo, mode (up, down, up/down or random), range (1-3 octaves), repeats (1-16 times) and decay (nice touch). The arpeggiator can be synchronized internally or to an external source.

The MIDI implementation is quite complete on the S220. Separate MIDI channels are assignable for each bank in the Multi Modes, and a channel offset feature permits the setting of non-contiguous channels – for example, Bank A can be set to channel 3, Bank B to channel 7, etc. MIDI Mono Mode is featured, however a maximum of only eight voices can be played at one time and the channel assignment in this case must be contiguous (1-8, 2-9, etc). A separate channel can be assigned for global control changes if you like.

The S220 can receive and transmit a wide range of MIDI messages such as Volume, Balance, Registered Parameter and System Exclusive messages, so some pretty powerful programming can be done right from the front seat of your sequencer. Program Change messages 1-28 allow access to the eleven structures as well as on/off of the Detune, Delay and Dual modes. Aftertouch is now recognized and can be assigned to control the output level or balance of the individual outputs and can also be assigned to control the amount of detune. The S220 can receive Note On/Off, Modulation, Pitch-bend, Hold, Volume and Aftertouch separately for each channel in Multi and Mono Modes.

Sampling couldn't be easier on the S220, separate mic/line inputs and a switchable limiter help keep the distortion problem to a minimum, and the 16-bit DAC at the output insures crystal clear playback of the samples. The auto-looping feature is a real blessing and if you don't like the computer's choice of loops there is always the manual-looping mode.

The only real problem I had with the instrument was with the quick disks which it uses to store information. They don't store a whole lot of data – though you can store performance information as well as sampled sounds – and they're not readily available outside of music stores. But that minor point aside, I'd say that Roland has a real winner in the S220. ■

PRICE $1395

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

Formerly Performance MIDI

Introducing...

Concepts: One

Beyond State of the Art!

For IBM PC/XTs, PC/ATs, '386 machines and most compatibles
and a MPU-401 compatible interface

Concepts: One represents the beginning of our new line of professional MIDI software and hardware products. The Concepts: One sequencing system is designed for performers, musicians, audio/film production, and studio applications, with features that you, the serious professional DEMAND from your MIDI sequencing software!

The Concepts: One (version 1.5) sequencing system is the result of more than two and a half years of intense research and development. We asked professionals what they needed in high performance MIDI software. Their answers led us to incorporate many features including:

- Support for up to 15 megabytes of extended memory on AT models
- The ability to record and play 16 MIDI channels simultaneously on each track
- Facilities for recording, playback, and editing of System Exclusive messages
- Real time muting, tempo, and All Notes Off commands
- Quantization available in three types: Note on, Note Off, or both. The quantize value may range from two clock ticks to a quarter note!
- Scrolling of note and controller displays in playback
- Automatic downloading of up to 16 sets of Bulk data (patches, etc.) before sequence playback
- Track lengths of up to 206,400 bytes each
- Full Song Position Pointer implementation
- Mix, Unmix, Split, Load, Save, and Copy individual tracks
- Cut, Paste, Copy, Remove, Replace, Delete, Transpose, Quantize, or change MIDI channels in a phrase of music
- Scaling, modification, or replacement of Note Velocity, controllers, Pitch bend, etc. within a phrase
- Loading of sequences automatically from a song list. Load any sequence within the list via MIDI Program changes from your MIDI master keyboard.
- Saving each Punch In to a different track while looping

In order to find out more about the unique features of Concepts: One, and order your **Free demo diskette,** call toll free **1-800-451-7973,** or in California **1-800-227-8218.**

See for yourself how Concepts: One is "Beyond State of the Art," the MIDI sequencing software professionals prefer.

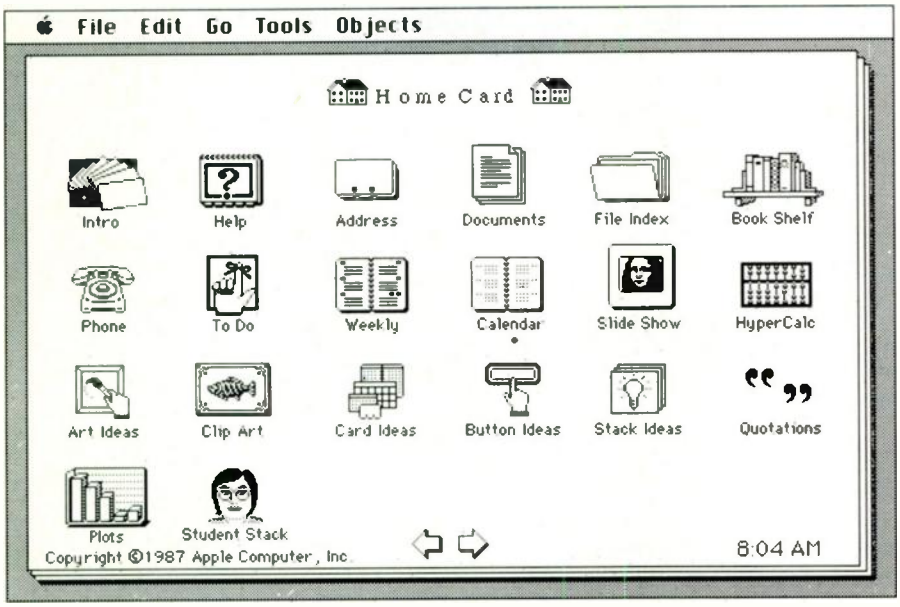
midi
CONCEPTS, INC.

7657 Winnetka Ave., Suite 327, Canoga Park, CA 91306
Pan ID: MIDICONCEPTS

Apple Hypercard

The new interactive "database-type" program from Apple brings a whole new perspective to the relationship between the Macintosh and education. *Review By Jim Burgess.*

REVIEW



HERE'S THE SCENE: You've just finished unpacking and hooking up your brand-new XX100 Triga-numeric Multiphonic synthesizer. Before turning it on, you take the owner's manual and light it in flames. As the smoke clears, you pull out a single disk and stick it into the mouth of your hungry Macintosh.

Soon a message appears on the Mac screen: "Tutorial: Getting the Most out of your XX100." A fanfare of trumpets emanates from the Mac speaker. A life-like digitized photo of the XX100 front panel appears on the screen.

First thing's first: you need to know how to turn Omni off and set the XX100's MIDI channel. One mouse click on the MIDI icon and the screen dissolves to show a menu of the XX100's MIDI parameters. You see the word "Omni" and click it on. The screen wipes to the side to make way for a step-by-step instruction that tells you exactly how to turn Omni off. At the bottom of the screen there's a question mark icon labeled "Set Channel?" As you click on it, the screen zooms open to reveal instructions for changing the MIDI receive channel.

Sound like a dream come true? Thanks to Apple, this scenario could become reality at your home studio. Hypercard, Apple's brand-new interactive, information medium program, is about to change the face of training, and in a broader sense, education itself.

Overview

HYPERCARD IS AN interactive medium designed to function as a unique information environment. It can be used to look for and store text, custom graphics and digitized photographs. Unlike a book, which is a linear medium (ie. you flip from one page to the next), Hypercard's interactive capabilities let you connect any one piece of information to any other. By emulating the way we think (by association), Hypercard lets you find out what you need to know – fast.

Hypercard's primary applications are education, training, organization and presentations. However, that's just the beginning; there are literally thousands of unique applications waiting to be discovered by creative Hypercard programmers over the next few months.

How it Works

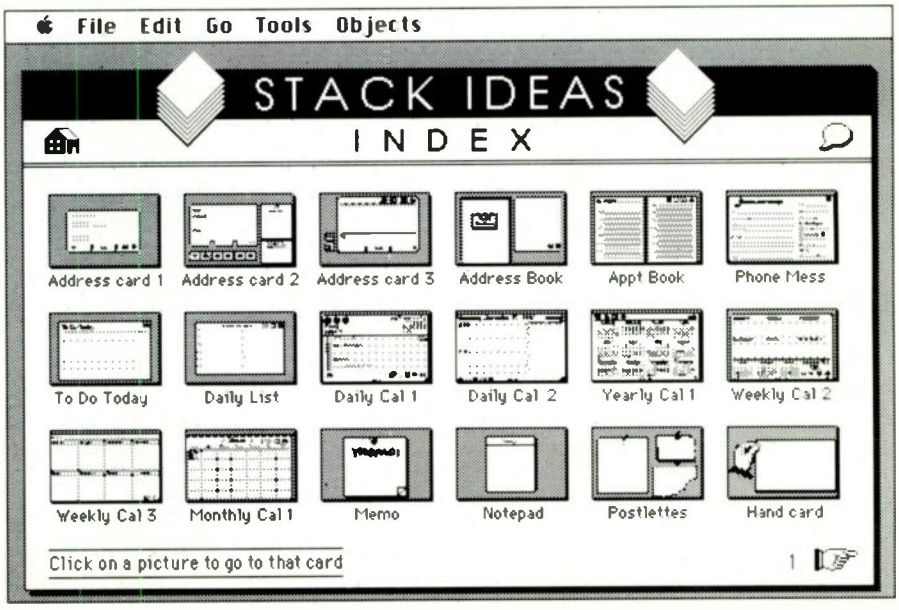
HYPERCARD STORES INFORMATION in the form of "cards." A card is a screen of data. Each card consists of custom graphics, a number of text fields and one more important thing: buttons. They're the link that lets you "connect" one card to another.

Individual cards are organized into "stacks." A stack usually contains a group of related cards – an address file of cards, for example. You open a stack the same way you open any other Mac file. Frequently-used stacks can be opened directly from the "Home" card just by clicking on a button icon that corresponds to the file you want. Think of the Home card as a kind of Finder within Hypercard.

Hypercard offers several "User Levels" to let you choose the level of interactivity that you require. Some may wish only to browse through other people's previously organized stacks, some might want to customize a stack for their own applications, while still others will want to create their own stacks from scratch. All you have to do is select the appropriate user level: for example, "Text" level permits the user to enter text onto cards, whereas "Scripting" level allows the user to modify graphics and edit script – the stuff that tells the buttons what to do.

Creating a Custom Stack

LET'S GO THROUGH the motions of creating a simple Hypercard stack. Suppose



you want to create a "Sound Stack" to keep track of every synthesizer or sampler sound you've got; a kind of sound database that tells you what sounds you've got, how good they are, and which disks to find them on.

Starting with Hypercard's built-in paint tools, you create a custom design for your card. You might prefer to simply modify an existing card to suit your purposes or create a new one from scratch. Either way, you'll find a complete selection of paint tools all nested together under a menu. If you're doing something that needs a lot of access to the paint tools, you can "tear off" the menu and use it as a window that sits anywhere on the screen.

Looks like MacPaint! The reason for that is because the person most responsible for Hypercard's creation is the one-and-the-same author of MacPaint, Apple's own Bill Atkinson. Not coincidentally, Hypercard's graphics capabilities resemble a much-improved MacPaint . . . .

By creating your custom card graphics in Background mode, you'll create a kind of a template that can be used to add new cards to the stack anytime you need to without having to draw them from scratch.

Once you're satisfied with the graphic design of the sound cards, it's time to decide just exactly what sort of information you want to keep track of. "Fields" are text blocks that you can place anywhere you need them. Using the Field tool, you simply click and drag a rectangular shape to create a text field of any size. The field can be re-sized anytime you need by dragging one of the corners to change the shape; if there's text already in the field, it will automatically re-justify itself to the new field size. Field parameters (ie. font styles, etc.) can be modified in a dialog box that will appear if you double-click on the field.

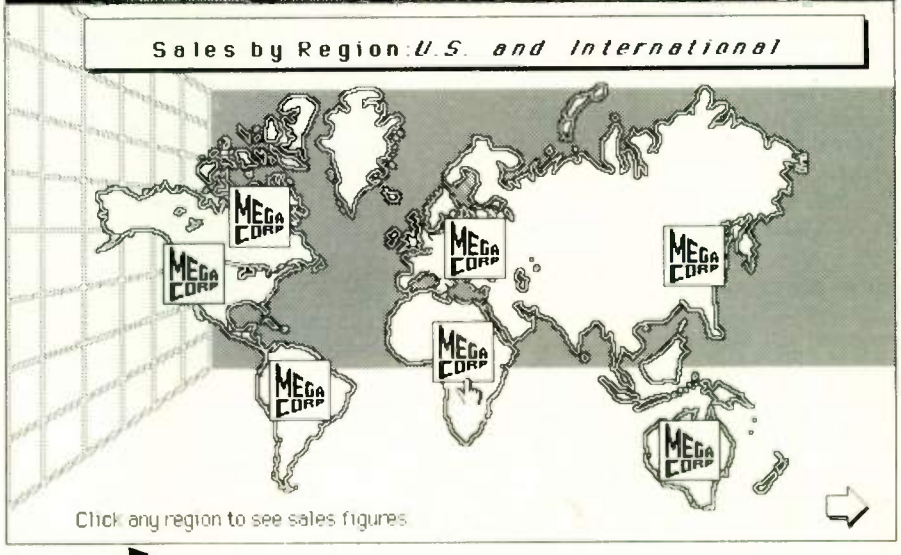
Any card may contain a number of different fields. In our case, we'll need fields for each sound's title, description, sound family/group, disk location, and rating . . . five fields in all.

With the text fields moved into place, you're ready to start the boring part of the job: entering in the data. In the end, every sound will have its own card and you'll have the computer equivalent of a rolodex file for your sounds.

But wait a minute – what about all that interactivity Hypercard is supposed to deliver? It's time to use the buttons to link some cards together.

In our example, for instance, it would be useful to have some way of tying together related sounds. If you need a trumpet sound but the only one in your file sounds like a kazoo, what's the next best thing? A cornet, of course.

You can easily tie your trumpet card to the cornet card by creating a button. Just go to your trumpet card and select "New▶



Button." Choose an icon from the ones available or make one yourself; perhaps the proverbial light bulb icon would fit the bill. Move the button to a convenient spot on the card. Like a text field, you can alter the parameters affecting any button just by double-clicking on it while you're using the button tool. Now, the final part: select "Link To" and move to the cornet card that you want to link. Now pressing that icon will take you directly to the new card.

HyperTalk is Hypercard's scripting language. Don't worry, you don't have to learn Hypertalk to create your own stacks. For most common operations (ie. linking one card to another), Hypercard will automatically generate the required Hypertalk instructions.

Don't be afraid to have a peek at the Hypertalk script of any button though; chances are you'll be amazed at how much it resembles plain old English. Eventually you'll start to customize the script for your buttons, adding visual effects like zooms and dissolves between cards, for instance. Before you know it, you'll be writing your own scripts.

That's it; that's all there is to creating a simple stack. Anyone can. Many will.

Conclusions

INTERACTIVITY IS A buzzword you've been hearing a lot lately. Resign yourself to it; you're going to be hearing a lot more of it because Hypercard has definitely arrived.

One interesting and important trend that seems to be developing as a result of software like Hypercard is that the distinction between the software programmer and the user is beginning to blur. Users can now customize their own software or make it from scratch. That's got to be a good thing!

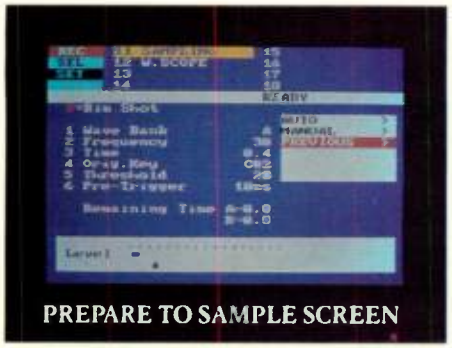
In fact, interactive applications like Hypercard will permanently change the way we educate and train ourselves. Learning can become not only more fun, but more effective too. Never before has one person been able to access so much information so quickly.

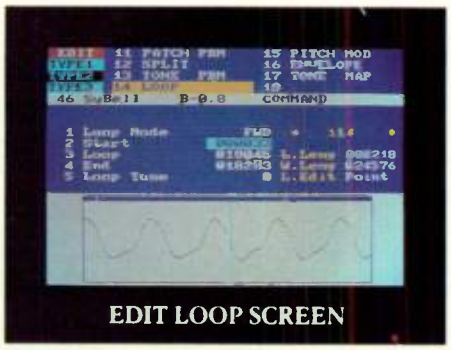
So the final question must be, how does it all relate to music? We can only wait and see . . . ■

PRICE $49

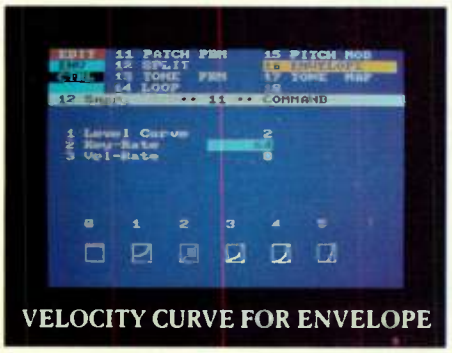
MORE FROM Apple Computer Corp., Customer Relations Dept., 20525 Mariani, Cupertino, CA 95014. Tel: (408) 996-1010

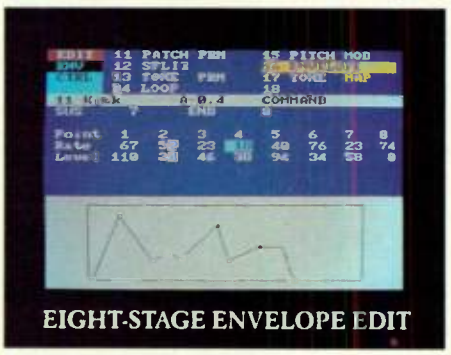
SAMPLE THE POWER OF THE ROLAND S-50


PREPARE TO SAMPLE SCREEN


EDIT LOOP SCREEN

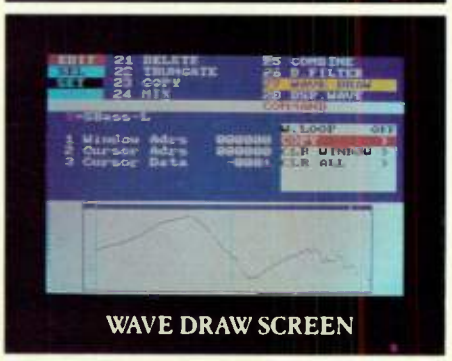

MULTI-TIMBRAL PLAY SCREEN


VELOCITY CURVE FOR ENVELOPE


EIGHT-STAGE ENVELOPE EDIT


TONE MAP GROUP EDIT

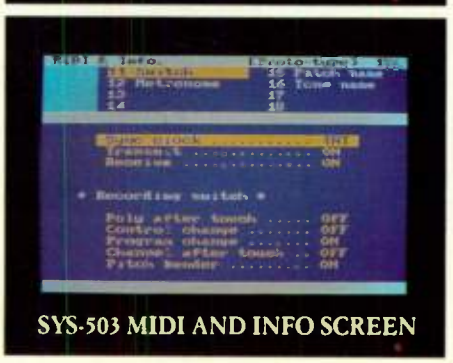

WAVE BANK DISPLAY


WAVE DRAW SCREEN


TRUNCATE WAVE DATA


SYS-503 PATTERN WRITE


SYS-503 SONG PLAY


SYS-503 MIDI AND INFO SCREEN

THE SAMPLER THAT DOES MORE TODAY

THE SAMPLER THAT DOES MORE TOMORROW

The Benefits of Open Architecture

There's no question, digital sampling is one of the most rapidly developing technologies in music today. So when you buy a sampler today, you don't want it to be made obsolete by some new development tomorrow. That's why there's one line of samplers that not only provides top quality today, but allows for future developments tomorrow. Naturally, they're from Roland, the company who always has one eye on the future.

Roland samplers are based around a "clean computer" design of open architecture. This means that the main performance data of Roland samplers is resident in software—software that can be updated with new features and performance just by loading a new disk. Neat trick? That's exactly what our customers think. They've already benefitted from one software update, giving their samplers loads of new features. And more updates will follow.

Professional Performance Sampling

But before we get too far into the future, let's talk about today. The Roland S-50 Digital Sampling Keyboard and S-550 Digital Sampler Module deliver the professional-quality sound and extensive editing capabilities found only in instruments costing many times as much, all thanks to Roland's breakthroughs in proprietary VLSI Sampling technology. The S-50 offers a wave oscillator, amplifier, LFO, and an eight-stage envelope generator for each of its sixteen voices. Although the S-50's features list is far too extensive to be listed here, among its chief attributes are a 512k word of Wave Memory and 16-bit processing, sampling time up to 14.4 seconds at 30kHz, multi-timbral capabilities, and four polyphonic voice outputs.

The S-550 provides all of the S-50's performance with the addition of a 1.5M Byte memory (for up to 64 tone memories and 16 patch memory banks), and expands upon the S-50 by providing eight polyphonic voice outputs.

Quality Sound in the Roland Bank

But we think the best way to judge a sampler is with your ears. That's where the Roland Samplers really shine. Play any Roland Sampler and you'll hear a warm and full sound, with a better bandwidth and greater headroom that especially shows up in dynamic instrument samples. You'll experience an evenness of sound across the entire keyboard without the problems other samplers have of obvious split points. And you'll never run out of sounds, because the purchase of a Roland Sampler gives you access to the Roland Sound Bank—a continuously growing library of great sound samples. Plus, the S-50 is already enjoying one of the fastest-growing bases of third party software support.

Quality In Means Quality Out

When it comes to user interface, Roland Samplers are simply unrivalled. Both the S-50 and S-550 allow the connection of a video monitor to greatly facilitate sampling process and use such new features as Wave Draw. The S-550 even allows the flexibility of mouse-style input by connecting the new DT-100 Digitizing Tablet. This kind of interface makes the experience of sampling sounds as fun as it is productive.

Today's Updates

The new 2.0 Software Update (available to all owners for a small handling charge) can now add in loads of new features—including twice as many tones, Automatic Loop Search, combined Wave Data, Polyphonic Multi-timbral performance, and much more. Not just new sounds, entirely new performance. Another new software program—the SYS-503 Director-S—can turn the S-50 or S-550 into a sixteen-channel MIDI sequencer, playing its own sounds as well as those of other MIDI instruments.

Put all this together and you can see why Roland Samplers are the choice of so many top pros. And why shouldn't they be? Because if Roland Samplers do this much today, imagine what they'll do tomorrow.

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

Roland®
WE DESIGN THE FUTURE

THE FAIRLIGHT COMPUTER

A Primer

You've heard about it, read about it and heard some music created with it, but you probably don't know too much about exactly what this dream machine can do and how it works. In the first part of a look at the Fairlight Series III, we give you some answers. *Text by Peter Bergren.*

A BOYHOOD FANTASY I remember especially well was that of being musically omnipotent. By this I mean being able to play all the instruments of an orchestra by myself. I envisioned myself at the controls of a keyboard instrument with bass instruments coming from my left hand and treble on my right. A battery of buttons, lights and knobs somehow sorted out which sound was where. Just how I could get it to play all the parts of an orchestral score at once I never did work out.

This fantasy came back to me while watching a demonstration of the Fairlight CMI, Series III. (CMI stands for "Computer Music Instrument," don't you know, and a very capable computer it is.) For there in front of me was my dream in the flesh (or should I say silicon?): electric bass on the left, acoustic guitar on the right, and various drums on the middle keys. Not an orchestra, true, but certainly a vast improvement over my experiences with Moog modular synthesizers in the '70s. Now, it seems to me, the limits on how many voices and parts can be played at one time are the number of fingers available to tickle the ivories. And the CMI's sequencer can overcome this limitation, to a degree, but I must say I turned a little pale at hearing the price. You can buy a house (except where I live) for what the sleek machine costs. Besides, couldn't I get the same results using separate MIDI'd voices and a PC/software controller? And have a place to live, too?

I hope to answer those questions, simply by outlining what the Series III is and does. Because it's a specialized computer dedicated to audio production, this will not be a simple task. So, I'll present what I feel is the "big picture," the salient stuff. In the next article, we'll get beyond this primer approach and discuss the Series III from a user's perspective.

Let's start with some history to clarify the problems involved in designing a "one-man orchestra" such as the CMI III.

Requirements

IF ONE PERSON is to command a "facsimile orchestra" (or dance band, for that matter), several considerations apply. First, musical voices have to be available on command, and the means of their control make sense to a musician. These voices can be samples, or synthesized, but they should be easily found, organized, and ready to play instantly with a controller that makes sense to a musician, not to a Ph.D. in Computer Science.

Second, a person can play only so many parts at one time. So some musically acceptable means of overcoming this has to be devised. And in the same vein, it's desirable to be able to hear as many voices as possible playing the same part at the same time. This feature must be part of the control structure.

Third, a realistic string section sound, for

example, is comprised of many components; that is to say, individual players contributing their own nuances to the group effect. Therefore, some method must be devised to create and control such "subvoices."

And fourth, all control information should be easily storable and recreatable and control status displayed in a way that makes sense to the non-technical.

Not an easy thing, this being musically omnipotent. But it's an old dream, certainly.

The pipe organ, the musical automata of the late 19th century, and the Novachord and Hammond organs of the mid-20th century were all attempts at realizing this dream. Then there were the Moog modular synthesizers of the '70s. They were a big improvement on what went before, as one keyboard command could fan out to control many parameters at once. The resulting sounds were sometimes astoundingly real, but they suffered from lack of polyphony and easy programmability, and *really* complex sounds, such as piano chords, alluded them.

With the late '70s came a deluge of programmable polyphonic synths and in the early '80s, samplers began to appear. These solved the problems of earlier synths to a great degree, particularly those of repeatability and duplication of complex attacks. Many machines now can offer more than one polyphonic voice output, and with MIDI, several such units allow performances of rather awesome complexity, especially when accompanying parts are pre-programmed. In fact, you can't open this magazine without encountering *something* really new – either in hardware or software – so why spend a fortune on a dedicated sound production computer?

Because there are potentially more efficient ways of creating "facsimile orchestra machines" than using separate components, that's why. One is the workstation approach, where everything is done via one computer system. This is Fairlight's approach, and its claims for the Series III are impressive. My reaction thus far is positive, with the one reservation being that it seems a rather intellectual, as opposed to a visceral, machine.

"Fairlight's intention is for the term 'workstation' to come to truly apply to the CMI Series III, with it being possible to perform all functions required in a recording session from one control position."

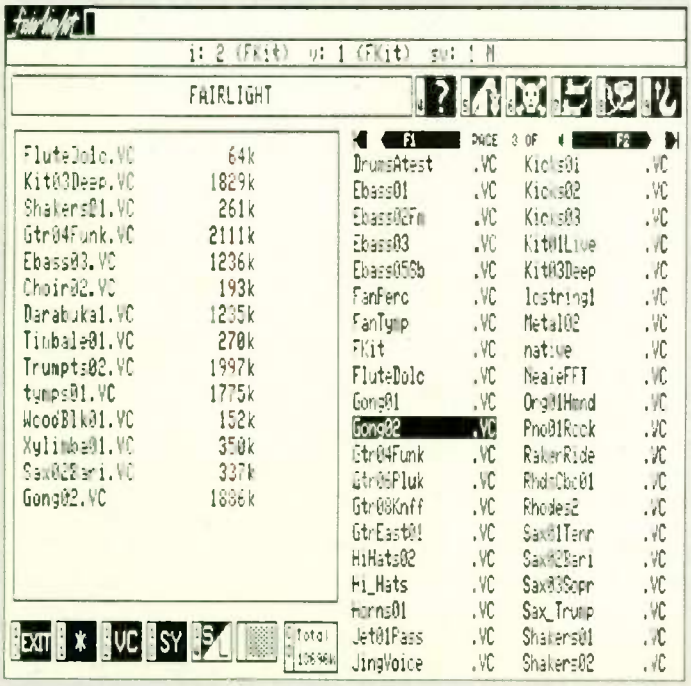
The Series III

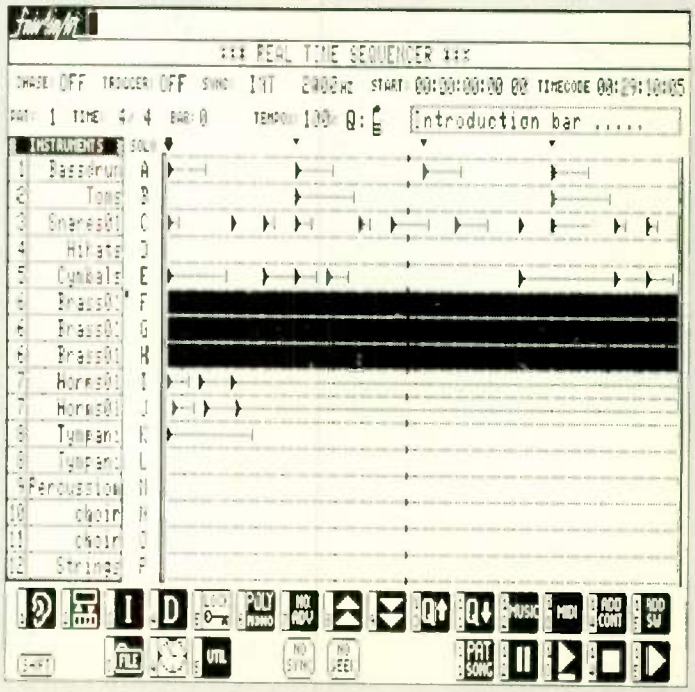
FAIRLIGHT'S INTENTION IS for the term "workstation" to come to truly apply to the CMI Series III, with it being possible to perform all functions required in a recording session from one control position. This would probably include access to large sound libraries and long duration digital disc recording. The CMI isn't there yet, but that direction and destination is certainly possible given what already exists.

At present, the CMI can sample and generate sounds, file them in file spaces organized under a flexible hierarchy, recall them to a large RAM for manipulation, process them, refile them in the same or different locations, and reproduce them manually, by sequencer, or by other means. The sound quality is at least that of a CD. Commands given to the system can be multifarious, and a constant dialogue between operator and machine is possible via graphic screens of the software in use. Moving the "grist" of one program into the "mill" of another is a matter of going from one software page to another, loading the sound by its file name into each in turn. Sort of like word processing for sounds. And all the while the video display unit (VDU) screen charts your progress, so "what you see is what you hear."

Very nice, you say, but let's get back to hard reality again for a second. Just how much *does* it cost?

A lot. A sixteen-voice system with 14 MBytes of RAM and one Winchester hard disk runs $75,000 retail, but this includes everything mentioned above and the ability to run MIDI to the outside world, to work to picture using SMPTE time code, to transmit via SCSI (small computer serial interface) ports to mass storage devices (such as optical disc), and to print hard copy. There's also an eight-voice unit at around $39,000, and a forty-voice "mega ▶





▶system" for $175,000.

The same software will run on all systems, with certain adjustments, and there is no difference in hardware quality between them. Included with all systems is a large factory library of sounds, selectable via a software directory, and easily loaded into the keyboard or sequencer. The instruction manual is well written and organized, though *heavy*.

Obviously, this system is not intended for everyone. Its targeted markets are those needing centralized production control, allowing the most efficient access to the maximum number of options with very high audio quality. People scoring television shows and commercial producers come to mind, as do those producing their own records. You definitely need the cash flow to justify the cost, but my first impression is that such producers would benefit from the machine's organization as a central workstation, especially if other synths are used as supplemental voices.

Using the CMI III's external MIDI channels is one way to cut the cost of extra voices and to centralize control in an existing synth setup. As with any complex technology, however, there's a mind set that must be learned, particularly regarding sound file manipulation, and command words and syntax, and learning these could take a while. But I suppose that's more a reflection of the system's depth than a criticism.

Hardware: Entering Commands

PHYSICALLY, THE SERIES III comprises a musical keyboard, an alphanumeric keyboard, the VDU, and the main card file cage which includes dual power supplies for analog and digital circuitry. Included in the 16-voice system is a 190MByte Winchester disk drive, and 14MBytes of RAM cards, but by the time you read this, the RAM space should double to 28MBytes, due to the introduction of a new RAM chip. At the present, however, the RAM cards each contain 2MBytes, for a total of seven cards, serving 16 voice channels (D/A converters) arranged in an eight-card group. The latter converts waveforms stored in digital form in the RAM (also called Waveform Memory) to analog form. The amount of RAM used for a particular waveform sample depends on its length, which is up to the user, and there are no rigid barriers between blocks of RAM, save for the times you run out of memory, which shouldn't be too often. Sample word length is a nice crisp 16 bits, linear format, and Fairlight tells me that all circuitry prior and following conversion is of very high analog quality. My ears tell me they're right.

The connections available to external hardware are comprehensive, and make allowance for further advances in mass storage (the SCSI port). Sixteen outputs from voice channels terminate in XLRs, including a mix of all voices. A provision for line level stereo sampler inputs also exists, as do connections for SMPTE time code in/out, three MIDI In and four MIDI Out ports, clock and sync out, and external device connectors including two printers.

The musical keyboard is six octaves long, with a good "feel" to it, and it has eight programmable "soft" controls, including footswitches (especially useful for toggling different key loads for playing). Resting atop the keyboard is the alphanumeric keyboard and VDU, and below, in the portable version, is the card cage. All in all, it's about the size of a Fender Rhodes. A rack version – which doubles as a slave voice unit – is also available for studio installation or voice/RAM expansion; the formula being seven RAM cards to eight voice cards. This hardware architecture allows for easy expandibility when needed.

"Commands given to the system can be multifarious, and a constant dialogue between operator and machine is possible via graphic screens of the software in use."

Commands can be entered and executed in a variety of ways, and there's enough flexibility in the operating system's interpretation of same to offer a degree of latitude in working style, though as with any computer, command word choice and syntax is important. For purposes of explanation, here's an example of one work style. A cursor on the VDU, positioned by a special pen moved over a graphics pad, can be made to "hit" a succession of command icons (much like a Mac), on a succession of software display pages. The commands accumulate in an input buffer, and once inputting is finished, the resulting command string can be executed with a single keystroke or function key. Commands can also be entered via function keys (15 in all), or simply by typing. The commands accumulating in the input buffer appear near the top of display pages in what is called the Command Line, and system responses appear below in the Status Line.

This system maintains a dialog with the user, whether he or she's a Mac or IBM aficionado. Other command keys are also available as is the option of actually drawing on the graphics pad. But I'd caution any musician whose sole experience with technology is a DX7 to expect some time to be spent getting into the peak of the learning curve.

Voice Organization

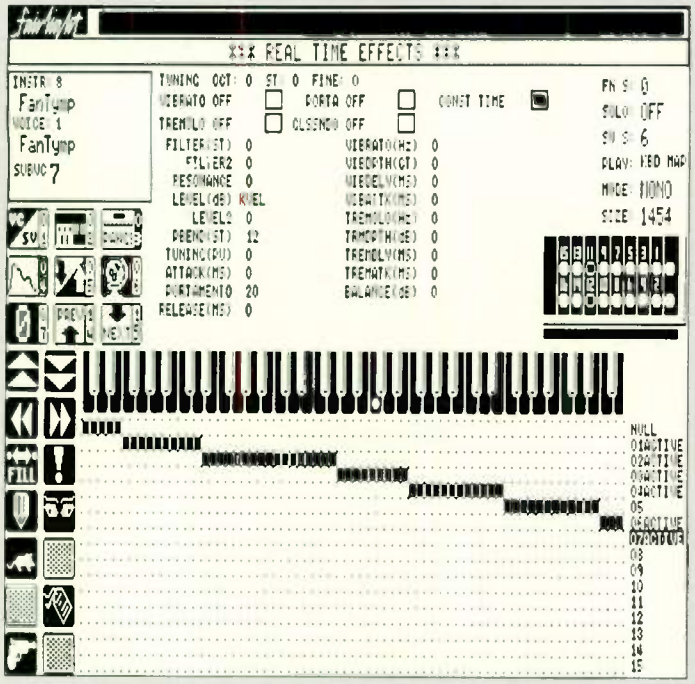
USING THE CMI, you can sample live sounds, resynthesize samples, or create sounds from "scratch." A set of software functions exist which let you change the smallest details of a sound, shaping them

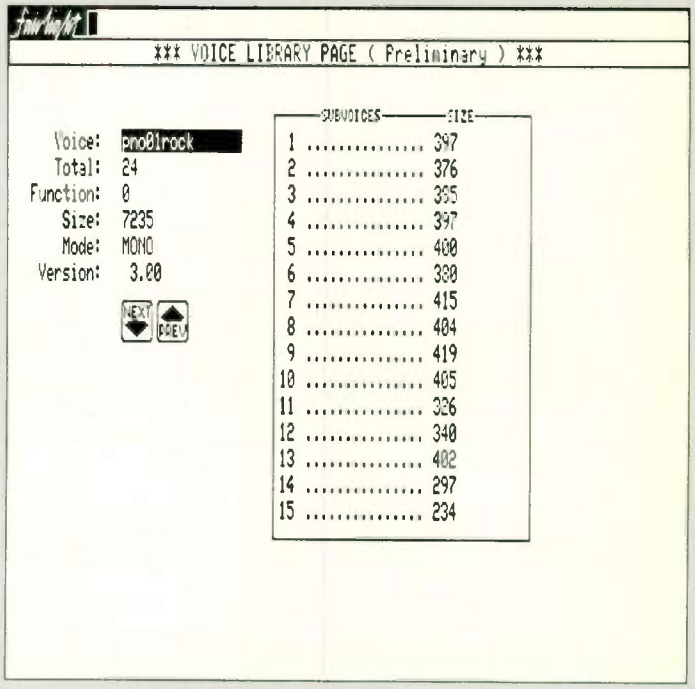
individually. But whether a separate sound is a live sample, or arrived at by other means, Fairlight calls it a Subvoice, which is the prime sound building block. Subvoices can be assembled into Voices, which are playable as one or more sounds and which are also thought of as file categories – allowing subvoices to be related to a name such as "guitar." In turn, Voices can be grouped into Instruments, which are also playing structures and file categories. An Instrument might consist of at least one guitar Voice, consisting of at least one or more guitar Subvoices, each of these being a sampled guitar sound mapped to the keyboard.

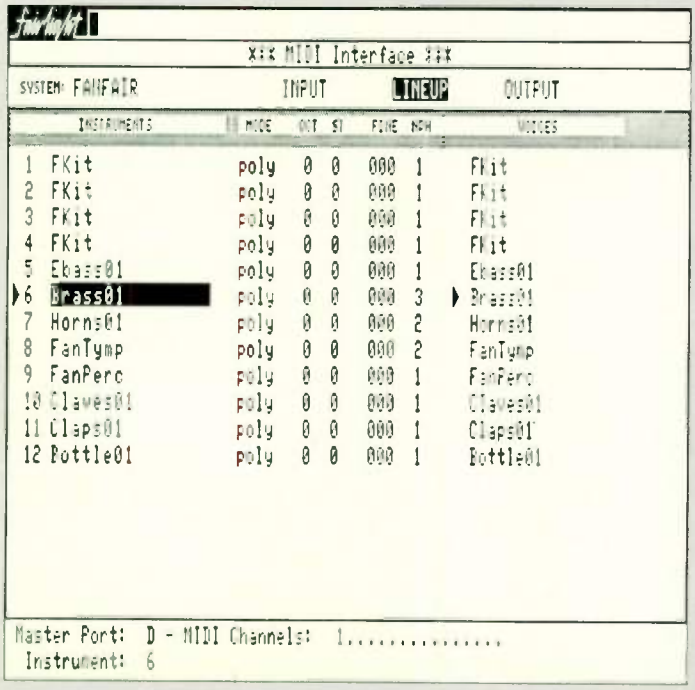
The filing process in this organizational heirarchy is part and parcel of the sampling and playing process. A sound sample has to have a place to reside in the machine (the Subvoice in the Voice), and the Subvoice can only be played by the Voice/Instrument structure recognized by the CMI as being where the Subvoice belongs. So, as the manual says, "to play the CMI, you play an Instrument, which plays a Voice, which plays the sound."

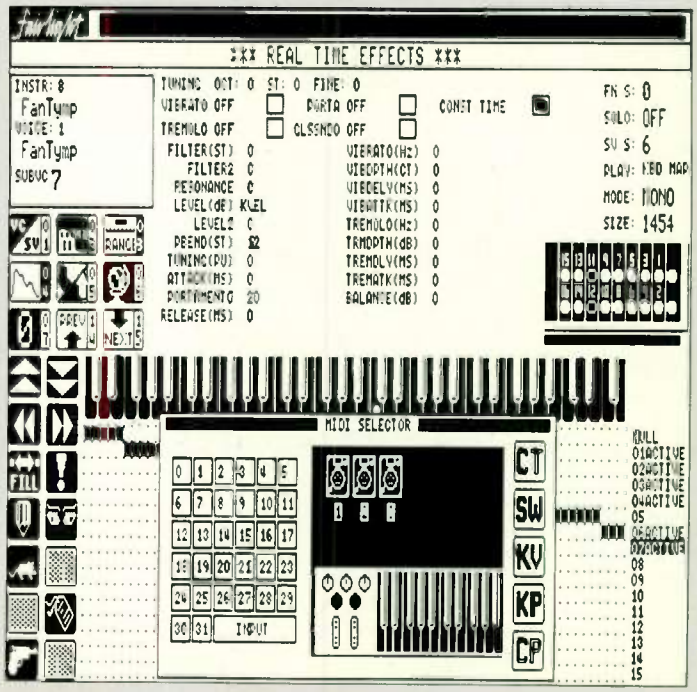
In general, sounds must have a name (file position) before they can be sampled or otherwise processed/mapped by software. Sounds occupy this file position when in RAM, but this slot is temporary, as RAM is for short-term storage only. Long-term storage is in user files on the hard disk, where sounds are filed according to their RAM file addresses, assuming this is desired. If you wish to use that sound (or sounds) again, a load command will transfer the contents of hard disk files to RAM, where

"Sampling can be done at the standard CD rate of 44.1kHz, or at a lower or higher rate (up to 100kHz in mono samples). Playback can be at upwards of 198kHz."









►further modification is possible. If you change a sound, or sounds, these changes accompany the sound sample itself when returning to hard disk. When recalled back to RAM, this stored parameter information is used to restore the sound (whether on the Subvoice, Voice, or Instrument level) back to its changed state.

When you load an Instrument into RAM, you can designate it as the "current Instrument." "Current" refers to whatever sound structure you're working on at the moment, and applies to Voices and Subvoices as well. The current Instruments/Voices/Subvoices have been mapped to playable keys, which could, for example, have a bass guitar available for the left hand, an electric guitar for the right, and an alternate acoustic guitar grouped around middle A. By loading one Instrument after another into the keyboard and recording your playing of each on tracks of the CMI Real Time Sequencer Software Page, you can rapidly build up several Instruments playing in accompaniment at once. This overcomes the present software limitation of being able to load only one Instrument at once into the keyboard. Such collections of Instruments are filed in a category called a System.

All sound data exists in digital form within the CMI up to the point it's actually played. Conversion of waveform data to analog signals is performed by the Voice Channels, which convert the instantaneous output of one Subvoice at any particular moment in time.

Sampling

SAMPLING CAN BE done at the standard CD rate of 44.1kHz, or at a lower or higher rate (up to 100kHz in mono samples). Playback can be at upwards of 198kHz. By juggling record vs. playback rates, it's possible to "stretch" one sample over several keys, thus saving space in the RAM. However, RAM space, as mentioned earlier, extends to cover sample length. The two minutes plus available in RAM at the standard rate (twice that with the new RAM chips) is enough to allow very detailed reproductions of complex sounds, such as pianos, with samples taken every few notes. Lengthy stereo samples can also be taken, such as of swelling synth chords or Tibetan chants.

Whatever the length, samples can be further manipulated to adjust macro level functions such as tuning, envelope, level and looping. Much more detailed, "genetic level" adjustments can be made to the sampled waveform using the Waveform Edit or Fast Fourier Transform software. On the Voice level, certain other changes can be enacted, using the Real Time Effects Subpage. All such macro changes are called parameter changes to Effects. Again, these alterations become part of the Voice or Subvoice file data when stored on hard disk, and when retrieved along with the sounds themselves

to RAM, restore those sounds to their altered condition when played. An option exists allowing you to audition Subvoice samples direct from hard disk if desired, however. With this capability, you can have almost instantaneous access to any library sample.

Voices

BY DEFINITION, A Voice must contain at least one Subvoice. Up to 63 Subvoices can be assembled in the file space assigned to a particular voice. This is a neat way of making available different groupings of related sounds, for when the keyboard is loaded with a particular Voice/Instrument, half of it could be filled with clarinet Subvoices, and the other with harpsichord Subvoices. What this really amounts to is a handy way of dividing the keyboard into sections that are musically useful. But there's really no reason why a sound effects editor, for example, couldn't map those same keys with 20 different door slams, and 30 different gunshots. Assignments of this sort would change upon loading of the next current Instrument from RAM.

Voice Channels

THE 16 VOICE channels are mappable to particular Voices in the Waveform RAM files. That is to say, at one moment in time, a voice channel plays one Subvoice from a particular Voice in a particular Instrument. If more than one key in the Instrument is depressed at one time (playing chords, for example), several Subvoices will "sound," requiring several voice channels to perform D-to-A conversion of the waveform data. In a like manner, if the default polyphony of an instrument (normally 1) is changed to 3, then each key depression will require three voice channels, one for each note in the chord.

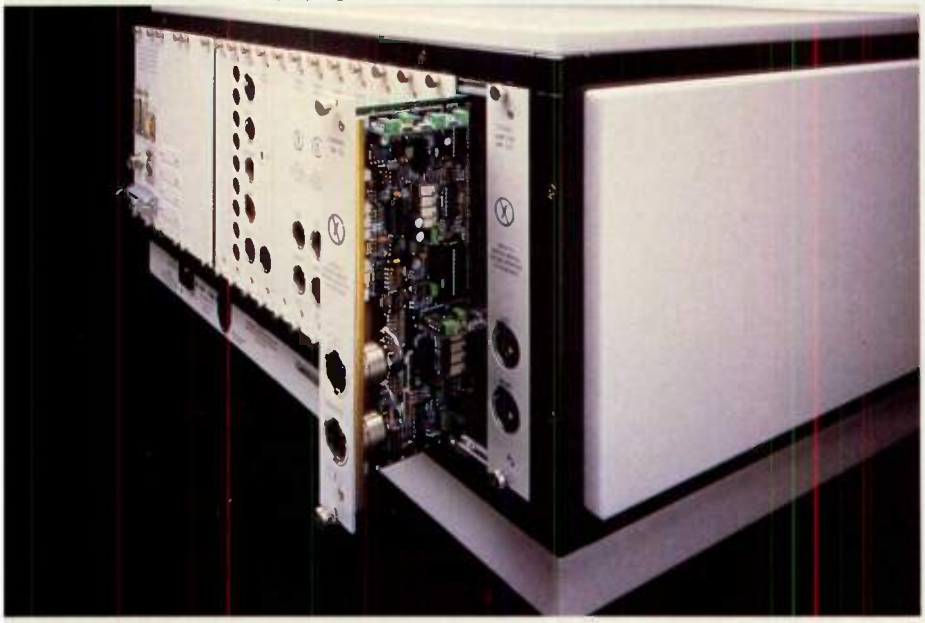
While playing one Instrument may not tax a 16-voice system, playing *several* Instruments, each of which may require polyphonic reproduction of several Subvoices simultaneously, very easily could cause a shortage of voices. The solution? Add more voices – up to 40 can be accommodated at present, with 80 in the offing . . . or you can use inexpensive synth modules controlled by MIDI parameters transmitted by a "dummy" Instrument. That's an Instrument structure with no purpose other than to convey MIDI pitch, level and other commands to external hardware.

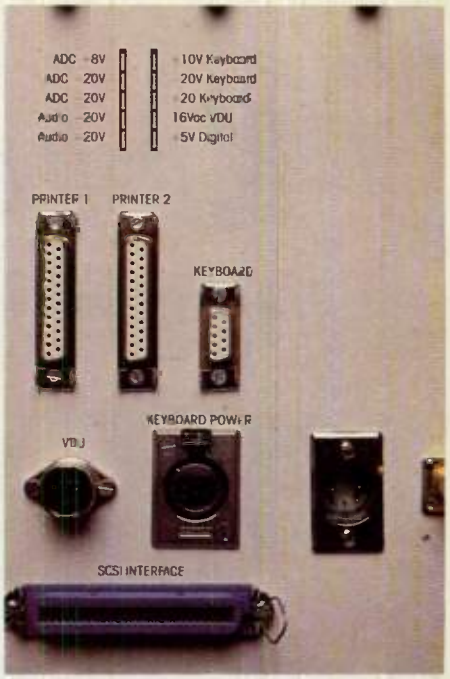
Most of the time, running out of voices shouldn't be much of a problem, however. After all, you can load up a multitrack from the sequencer. And, from the perspective of a sound effects editor (or a musician requiring a lot of variety), 16-voice channels assigned to 16 separate Voice files, each containing the full complement of 63 Subvoices would give you 16×63=1008 separate sounds playable sequentially!

Instruments and the Keyboard

UNDER THE PRESENT software scheme, the keyboard can only play one Instrument at a time. To add texture to the resulting sound, it is possible to "buddy" voices (a very Australian term), or stack them, if you prefer. This simply means that a file Voice is added to the Voice already in the "current" Instrument loaded into the keys. Then, when the keys are depressed, Voices will play in concert. It's really just like ranks of organ pipes being activated by common keyboard commands when stops are drawn out. This stacking arrangement doesn't show up on the Real Time Effects Voice and Subvoice mapping display, but does show up on the System Configuration page.

About the time you are reading this, there will be new software that allows groups of Instruments to be played in real time from the keyboard. In addition, there will be a new software display page showing the Instrument to Instrument, and Instruments to keyboard relationships as well.





Real-Time Sequencer

ALSO KNOWN AS an expanded version of "Page R," the intention of this software is to allow rhythm tracks to be built quickly, though any Instrument loading of the keyboard can be recorded. The resulting musical frameworks can serve as "stand-alone" compositions, as parts to be accompanied by others on multitrack, or as input for the CAPS Sequencer Software, which is still under development.

In using the RS, you start with a click/metronome track and in general work in 4/4 bars. It seems to be a very fast program to work with, as all you do is load up the keyboard with an Instrument, play a one or two bar pattern while recording on one track of the sequencer, and then have that track play back while you record another line on another track. You can record up to 16 different Instruments total, and arrange them in over 255 patterns. These in turn may be arranged in different orders in each of 26 sections, labeled A-Z. These could, of course, correspond to the sections of a song, or longer composition, for easy recall. Patterns and sections can be made to repeat in any way you desire (for example, a continuous repeat of the same two-bar pattern on each of several tracks).

It's possible to "solo" individual tracks, and using other software, go in and make expression changes on individual phrases, notes and sounds, in various degrees of detail.

In the next installment, I'll describe some of the other software and investigate the CMI Series III from a user's perspective, pro and con. ■

REVIEW

Keytek CTS-2000
Crosstable Sampled Synthesizer

The newest instrument from an Italian company offers an interesting method of synthesis as well as extremely flexible editing capabilities. The question is, how does it sound?

Review by Chris Meyer.

Photography Scott Peer

AMERICAN AND JAPANESE synths; that's all we seem to see anymore. The Europeans also make synths, however, like they make automobiles, but we just don't get to see them much over here. European synths are generally characterized by their abundance of features, arcane user interfaces, strange looks, unique sounds, slightly higher prices, uneven factory presets, and limited distribution and service networks – most of which could also be said about EMS, Wersi, PPG, and OSCar.

The Keytek CTS-2000 is made in Italy and distributed in this country by Gibson USA – yes, the guitar people. In the most neutral of spec-sheet terms, it is a wavetable synth with a plastic, five-octave, velocity-sensitive keyboard, allowing up to five splits. It has stereo outs, 48 programs arranged in two banks of 24 (and a cartridge slot), eight voices featuring two oscillators (each with a pitch envelope), two DCAs (digitally controlled amplifiers) and a resonant four-pole DCF (digitally controlled filter), each with six-parameter envelopes, three LFOs, and a nice MIDI implementation. It has a considerable number of features, only slightly dubious looks (some find it cheesy, I don't – look at the photo for yourself), and a midrange price. How does it fare? Read on . . .

Controls and Editing

EDITING ON THE Keytek is a nice advancement on the now all-too-common parameter select and data slider method. Twenty-four "families" of parameters are available, with each family consisting of two to six parameters. Along with this are six(!) data entry sliders, a pair of inc/dec buttons, and a backlit 16X12 character LCD. Whenever a family is selected to be edited, its name appears in the LCD, and moving a data entry slider brings that parameter to the LCD. The display always shows the old values and the new value of the parameter you are editing. The data entry slider has to be moved a bit to register a change (a bit too much for my tastes, actually), but as a result, it's easy to check out the values of everything without changing the parameters.

The five splits, or "zones," are hard fixed to the five octaves – C through B. Each zone

may have completely different sets of parameters. Editing may be per zone or globally adjusted across all zones. To say this is flexible would be a moderate understatement. However, I was unable to find a way to edit more than one zone in zone mode at once. Therefore, I tried starting from scratch (there is an "initialize preset" feature), and editing a full five-octave keyboard in global mode. The process ended up being slightly frustrating, though, as in global mode, parameter values are shown only as plus or minus offsets from the last time you edited them. In other words, every time you go to filter cutoff, for example, it says "0," even if it's wide open. That's a bit disconcerting.

I quickly changed to editing just one octave in zone mode, and copied that zone to the other four. As flexible as the copy utilities were – more later – there was no way to do this full copy in one fell swoop. Also, the copy utility insisted on throwing me back into preset mode after each operation – unnecessary extra button pushes. When done, I had five identical zones – pitch range and all. Achieving a normal keyboard setup required resetting the octave for each zone. I wouldn't be surprised if some future update (or manual tip) makes that a whole lot easier.

After the zones have been programmed and copied, global parameters can be used to tweak the whole set, or individual octaves can be adjusted (such as making the lowest octave a touch brighter). This is very similar to the way some samplers edit a keyboard of multi-samples, and many parallels to editing samples show up throughout the machine. But however familiar territory this may be to sampler users, it may be daunting to many "instant gratification" synthesists.

The 24 membrane-style parameter family switches double as program select switches when in preset mode. Other front panel switches include bank select, cartridge and internal select, MIDI on/off (nice touch), MIDI local control on/off (for master keyboard or sequencer applications – nice touch again), and write and enter switches. Rounding out things you touch and move are a volume slider, a spring loaded pitch-bend wheel, and an unsprung mod wheel. Everything is clearly labeled (including a line along the top showing where the back panel connections are), making the first glance at a CTS-2000 very friendly indeed.

Crosstable Sampling

DON'T LOOK NOW, ma, but it's another buzzword. "Crosstable Sampling" (as briefly explained in "A Deeper Wave Than This" feature, MT August 1987), is this: take a complex waveform, sit on it for awhile, and then fade it out while fading in another. Sit on that for a while, and fade it out while fading in a third. Stay there until the player lets go of the key and the amplifiers die out. The result? A lot of motion in a sound.

With the CTS-2000 you double this capability because there are two of these oscillators, and a total of 333 waveforms to choose from. Before you get too excited about that, however, it's time to remember the split structure explained above. Each zone may have a different set of waves – which is great – but each wave only covers one octave (or "zone"). Therefore, it takes five similar waves to cover a keyboard. Not all families of waves are five strong, though, so splits become not just an option but a necessity on some sounds. Also, the two banks of programs have different sets of waves, and waves from one bank cannot be combined with waves from the other. What this boils down to is less than 333 different full-keyboard sounds that can be combined in any way, but it is still a staggering number nonetheless.

Features, Features, Features

IT'S TIME TO look at what all those parameters in the 24 families do in the context of building a patch.

The first thing I did was read the manual, which consists of one long continuous dot matrix computer printout. Imagine my shock and surprise when it ended up being the *best* manual I have read for any electronic instrument, period. It was logical, clear, and brief, with important information repeated wherever needed. High marks on this point indeed!

I started by finding a preset I wasn't fond

REVIEW

of and initializing it, which yields a basic, static, two-oscillator sawtooth patch with no velocity. I went over to "Osc Levels" and turned off one of the oscillators so I could concentrate on one at a time. For "cross wave 1", I selected my octave, and picked an infinite "delay" on the crossfade envelope so I could concentrate on the first wave. I then scanned the waves, mentally picking out ones that sounded interesting, and then chose three for my progression. I set my first wave, reduced the delay (the amount of time it stays there before moving on towards the second wave) to 1, and started playing with the envelope.

REVIEW

Editing *"Editing may be per zone or globally adjusted across all zones. To say this is flexible would be a moderate understatement."*

The envelope structure is an expansion on Korg's "breakpoint" idea. Essentially, you are traveling from one point to another (wave 1 to wave 2). The user gets to set a "break point" between these two points, and then picks two rates – one from wave 1 to the break point, and one from the break point to wave 2. After my initial confusion with this new concept, I quickly fell in love. I started out this patch by picking a fade that rushed most of the way to wave 2, and then eased in the rest of the way a bit slower.

The above, with the exception of the octave selection, is repeated for wave 2, and then the finishing wave – wave 3 – is selected. Having become used to the one parameter/one slider paradigm on many other synths, I found myself reselecting the same family whenever I wanted to adjust another parameter within that family. Silly; all I had to do was move the appropriate slider (again, there are *six* of these). Having the sliders on the left and the parameter lists on the right resulted in a lot of eye sweeps at first, but I quickly learned to leave my left hand put.

Next I moved to the "Tune and Frequency EG" family. Here, the pitch of the oscillator may be offset by +/−5 semitones (strange – no augmented fourth) and a fine tune amount. A simple attack/decay envelope on the pitch that occurs at the start of the sound is available, with direction and depth being selectable. It's nice for adding more articulation to the attack of the sound.

After the aforementioned "Osc Levels" family are the "DCA" families. An amplifier envelope per oscillator is available (a la the ESQ1), which is, again, a nice feature. Envelopes are of the Korg attack/decay/break point/slope/sustain/release variety, with an additional second release function (a different release time when the release pedal is held – like sustain on a piano). Second release is usually glossed over in favor of a simple lockjaw "hold" on most synthesizers, and is appreciated here.

The voice path now leads to the "DCF", or digitally controlled filter. The manual claims it is of the four pole (24db/octave) variety; it sounded gentler than that. Its sound, in general, was unlike other filters I've encountered – as opposed to being obvious, it was more like the high-end EQ on my mixing console – all I really noticed was that the high end was either there or attenuated. Now, this is a compliment for equalizers; but somehow, I expect a synth filter to be more rude. The resonance had a different quality too – neither the weedy sound of most Japanese synths nor the growl of American synths. It was more like a light "vowelish" touch on top of the sound. Not unpleasant, but again I tend to subscribe to the "in your face if I want it" school of synthesis.

The filter sports the same type of envelope as the DCA, and has the conventional cutoff, resonance, and envelope amount settings. The envelope amount may be inverted for reverse wahs, which is yet another nice touch too seldom seen. It also has a keyboard tracking parameter (which affects how drastically the filter changes across the keyboard) – yet another unfortunate ommission on some synths. After that, a master level and pan position may be set for each zone.

And then (whew!) it's time for oscillator 2! Even with one oscillator, having three different complex envelopes for wave, filter, and loudness gave a lot of motion. At this point I had a sound that started as a cross between a wood xylophone and an organ with full percussion, quickly moved into being a muted woodwind (with a quick "wah," thanks to the filter), which then slowly faded into a bright synth tone – not bad!

Now I want to remind you that all of the above can be programmed *per octave*. We're talking pure craziness to be able to set up each octave differently in this way.

As for LFO's, there is one dedicated to amplitude per oscillator (again, generous). Each may have a separate waveshape (triangle, square, or random), speed, initial level, final level, and fade time from one to the other. Those who have not experienced delayed LFO's are missing out on an important feature for making synth sounds more natural. A third LFO is available which is cryptically labeled as being for "osc 3," when actually it can be routed to the overall level (DCA 1 and 2 together), DCF, or both, and has all the features of the other two LFO's.

The rest of the families have to do with performance parameters, MIDI, and edit utilities. In the "wheels" family, the pitch-bend wheel can have different depths for the two oscillators (again, the power and flexibility begins to stagger one), and the mod wheel can be turned on or off for the three LFO's. A family also exists for selecting the keyboard mode (monophonic or polyphonic, again per zone), type of portamento, and portamento speed.

The velocity sensitivity may be routed independently to the two DCAs and the DCF. It does this by assuming that the highest velocity is always the highest level, and the user chooses how low the quiet ones go. Unfortunately, there are no inverted amounts for velocity crossfades between the two oscillators. There is, however, a feature very similar to velocity switch on samplers, where thresholds may be set independently for when oscillator 1 shuts off and oscillator 2 turns on. Each oscillator is complex enough to be a sound onto itself, and this allows different strokes to switch between those sounds.

The MIDI family allows selection of mode, the MIDI channel a zone transmits on, the one it receives on(!), what program change is transmitted per zone (master keyboard capabilities there), and a "transmit transposition" amount (+/−48 semitones) per zone. No function exists to enable or disable other parts of the machine's MIDI data sources (such as wheels, etc), but this is a small point, and feels like niggling after all that the Keytek gives you.

A family is dedicated to transposition (whole keyboard) and what Keytek misnames "dynamic allocation," which is really whether or not to steal voices still releasing at the price of accepting or ignoring new notes. The final families are for copying (again, thoughtful, given the number of parameters to adjust), naming your creation (ten characters as per the DX7), and comparing your edited creation with other existing ones you may want to replace. It is worth mentioning that the copy utilities let you pick "from" and "to" zones, and in the case of copying a whole zone, "from" and "to" which preset.

In general, I found editing a voice a lengthy but fun task. A lot of parameters are available, but the voice structure is logical, and accessing the power does not require any secret incantations.

Other thoughtful features include a master tune control on the back, stereo outs (with one doubling up as a mono), and a real MIDI Thru jack along with the standard In and Out. The manual claims that it comes with a release pedal; my pre-production model did not. I made a point of leaving the unit on for over 24 hours, and it

Sound *"Ignoring the factory patches, I was able to come up with numerous sounds that excited me, but the audible stepping of the envelopes simply ruined my fun.*

did not get unduly warm (in contrast to, say, Prophets, which I've burned several fingers on no matter how much I love them).

But How Does It Sound?

WHEN ONE GETS inside a new car for the first time, it is easy to be impressed with the upholstery, all the instruments on the dash, the powered mirrors and the like, as well as the occasional nice touch, such as a convenient map pocket or ignition key light. Often it's enough to make you nearly forget to check out such things as the ride, the cornering, the acceleration, the engine noise, the brakes and so on . . .

When I was sent the CTS-2000 for review, I was warned that many others had, ahem, *not admired* the factory presets, and was beseeched to try to program the machine before passing judgement. Indeed, I didn't even turn the machine on for a few days, studying the front panel features and manual first.

Well, the wait must not have made much of a difference because to these ears, the factory presets are indeed bad. In fact, not a single person I've had contact with has come away from the machine with a positive impression after hearing them. They consist mostly of overly buzzy electric pianos (the CP70 imitation almost passes because the real thing is so bright), buzzy strings, brass, organs, and guitar imitations. There is the occasional almost-nice vocal or intriguing multi-split, but I'm afraid that your average first-impression buyer (and salesperson) will quickly pan the instrument based on the factory sounds.

Even worse, it seems that the hardware just wasn't implemented cleanly. Even straight sawtooth waves seem to have a bit of fuzz around them. The output level was very low too, harming the signal-to-noise ratio. Worse yet, the envelopes seem to lack any smoothing – particularly the wave crossfades. One can easily hear the sound change in stepped increments from one wave to another, as opposed to smoothly (such as on the Prophet VS, the first machine I compared this to). The clicking of a PPG switching waves has become its trademark and is rather quaint on many sounds, but the Keytek just sounded plain unrefined.

This lack of resolution seems to permeate the machine. The LFO's and velocity resolution are a bit rough, and almost all the parameters have too coarse an adjustment – from 32 to only four levels in the majority of cases, as opposed to 32, 64, or 100. Consequently, programming the instrument required a bit of the art of compromise here and there, and the machine in general has a rough, unfinished sound to it. A touch of digital reverb helped to put some clothes on the nakedness of the beast, but it was still never right to my ears.

Conclusions

IN GENERAL, I felt like I had a Mercedes with a couple of cylinders missing. The features are among the best and most comprehensive of any synth I've encountered at any price. Editing, although a bit strange in places, was friendlier than on the vast majority of current synths. However, the machine just didn't sound good – not for lack of an interesting voice structure, but because the hardware couldn't pull it off. Ignoring the factory patches, I was able to come up with numerous sounds that excited me, but the audible stepping of the envelopes (and to a lesser degree, the characterless filter) simply ruined my fun.

The qualification I'll give to this negative conclusion is that I'm very choosy about sound quality, and we all know how subjective sound is. I also had serial number 1002, which is one of the first models in this country. On the chances that Keytek has sorted out the hardware bugs before heavy production starts and that the stepping noise doesn't bother you as much as it did me, ignore the factory patches and go try to program one yourself (maybe even through an inexpensive digital reverb) before closing the books on this babe. ■

PRICE $1995

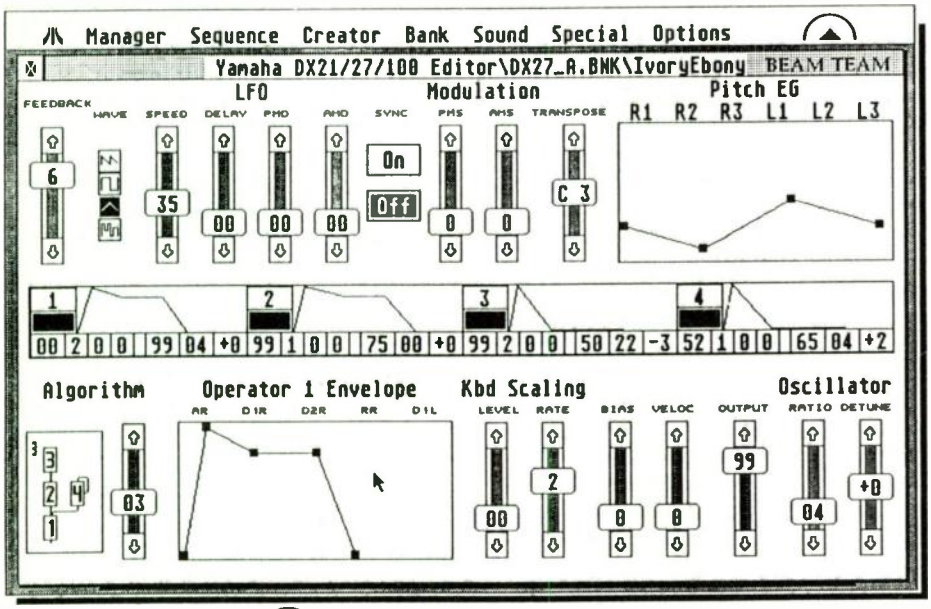
MORE FROM Gibson, 641 Massman Dr., Nashville, TN 37210. Tel: (615) 366-2400

REVIEW

Transform XSyn
Sound Editing Programs

REVIEW

Beam Team's first round of modular voice editors for the Atari ST are finally available. They just might make patch editing and organizing less of an ordeal. *Review by Bob O'Donnell.*



BEAM TEAM • XSYN-DX27 • SOUND EDITOR MODULE

WORKING WITH VOICE editing programs always proves to be an educational experience. There's nothing quite like seeing all (or at least most) of your synth's parameters intelligently laid out on the screen of your computer and having quick access to them. All of a sudden, those odd synthesis methods lurking within your instruments begin to make sense and the subtleties of various patches start to become more apparent. (Strange how you have to sometimes see things before you can hear them.) Of course, if you are into actually *programming* synths – some people still *are*, believe it or not – voice editing programs make the process infinitely easier, faster and more rewarding.

The only problem with this kind of program is that once you have one for one of your synths, you'll want one for the rest of your synths. Often that means learning an entire new program, to say nothing of the hassle of having to quit one program in order to boot the next one.

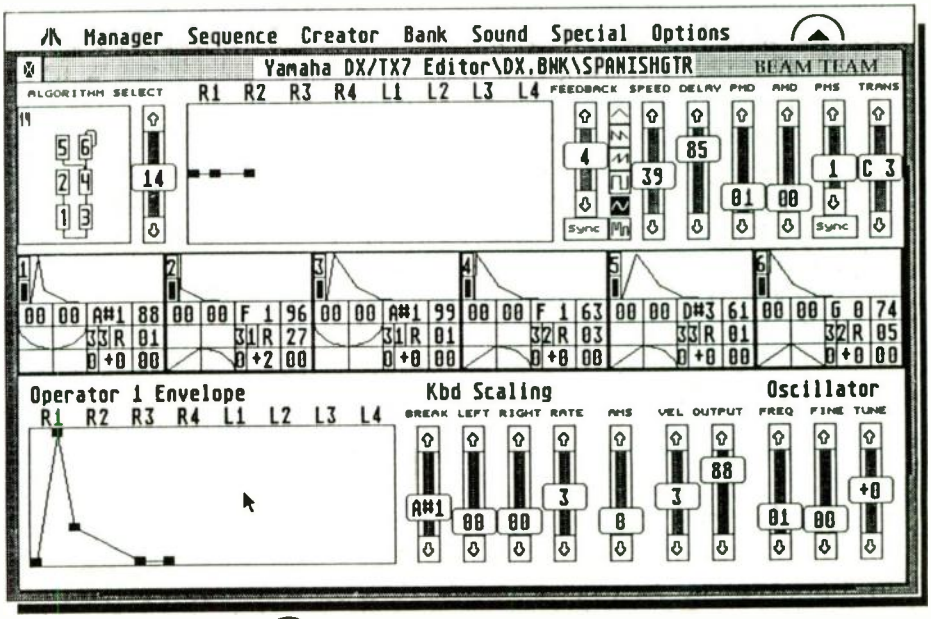
The folks at Beam Team (a German company who have a similarly named office here in the US) have come up with an alternative, however – a system-oriented group of modular programs, referred to as XSyn modules, all of which are available from within a main "Manager" program. Each of the modules works with a different synth and they're each considered separate programs – that's also how they're sold – but they can all be easily added to the main program to personalize your own system. In other words, Transform can be thought of as a series of programs within a program, any of which can be accessed from any other.

The complete Transform Modular Music System, the total system's official name, will include a sequencer program called XTrack, a music score layout program entitled XNotes and a laser printing program called XScript, as well as the XSyn voice editing modules. The only parts of the system which are currently available, however, are the voice editing programs, so for the time being we'll concentrate on those.

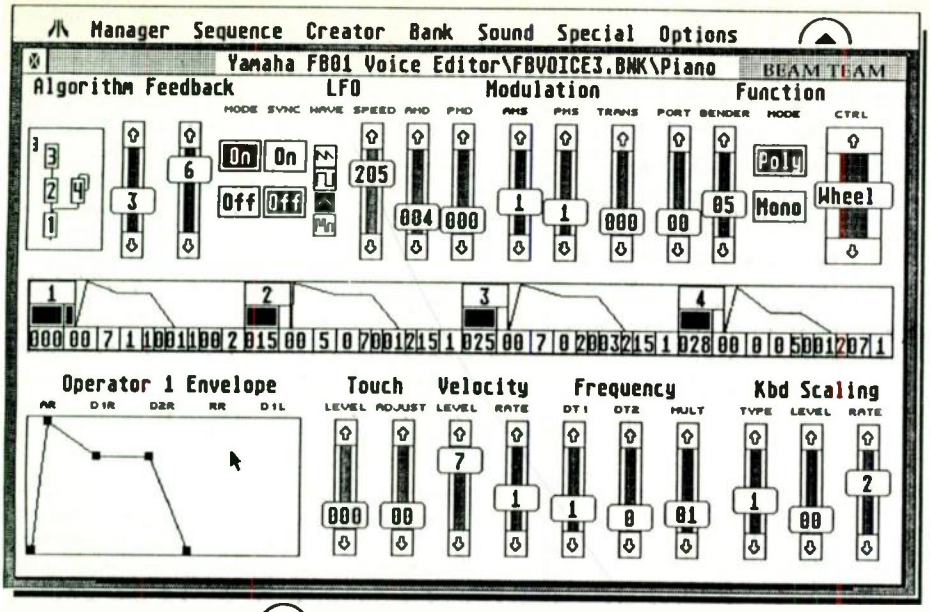
The Basics

EACH XSYN MODULE consists of five parts: a librarian, a voice and function parameter editor, a patch creator, a one-track MIDI sequencer and the program manager. The package also comes with a RAM Disk program which loads automatically if you boot with the program in your disk drive. The RAM Disk takes up 512K of RAM, however, so it only works with ST's with at least a Meg of RAM. (If you have a 520ST, you can simply save the RAM Disk program on another disk, erase it from the original and then reset the computer.) What the program basically does is add the equivalent of an internal disk drive (data is *not* lost after a reset) and speeds up the operation of the various editing functions.

After you've booted the Transform program you're presented with the



BEAM TEAM • XSYN-DX7/TX7 • SOUND EDITOR MODULE



BEAM TEAM • XSYN-FB01 • SOUND EDITOR MODULE

sequencer screen, and from there you can either record or load a sequence – the program comes with six song files – or enter the librarian portion of XSyn. The idea behind presenting the sequencer first – and it's a good one in theory – is to have some musical examples ready so that you can hear your edited sounds in a "real world" context. The program allows you to hear parameter changes in real time and by having a sequence looping as you edit a sound, you can get a very good idea of how the sound will work. Unfortunately, the actual implementation of it has some drawbacks.

The program only allows you to Play, Stop or Record a sequence – which is fine, because it's only supposed to be a basic sequencer – but when you hit Record, nothing happens. Or at least it *appears* that way because there's absolutely no click against which you can record. The program simply begins recording away and you're left to fend for yourself as far as the tempo is concerned. On a more positive note, the sequencer records pitch-bend, aftertouch, mod wheel, program change data, and on a 1040ST has room for approximately 80,000 notes with velocity, but the lack of a simple click (and any form of editing) is a real oversight. The somewhat confusing owner's manual (translated German is not much better than translated Japanese) points out that the sequencer will load files from the XTrack sequencer; but until that's available, you're stuck with this one.

The sequencer screen also has a keyboard on it and you can play the various notes on the keyboard with the current patch by pointing at them with the mouse and clicking. One oddity that came up when testing the program was that occasionally the note would sound repeatedly at a very fast rate for no apparent reason, but this actually proved to be useful for certain percussive sounds.

The next step in using the program involves choosing a synth module. XSyn makes use of the ST's GEM system of pull-down menus and underneath "Manager" are the available choices. The Roland JX8P and JX10, Yamaha DX7/TX7, Yamaha DX21/27/100, Yamaha FB01, and Casio CZ series are currently supported, and an Oberheim Matrix 6 module will be available shortly. One very nice feature of the program is that even if you only buy one editor module, the librarian section will work for the entire range of supported instruments.

After choosing the appropriate synth, the Sound Bank Manager, or librarian screen of the program will appear and, if you've chosen a synth for which you have an editor module, you'll be confronted with the message, "Sound Editor Resident! Check your Synthy for Exclusive Data ON, Memory Protect OFF. Let's go." (Don't ask me, I guess it's German slang.) You'll then need to load a bank of sounds from disk or from the synth's internal memory into one of the eight available bank locations – two are always visible on the screen. Each of the banks hold 32 sounds (except for the FB01, which holds 48 per bank) and you can easily scroll between them.

The Sound Bank Manager's Send on Select option allows you to change programs as a sequence plays by simply pointing and clicking with the mouse. You can also disengage this function, but it works well in giving the program a more interactive feel.

Editing

TO ENTER THE Sound Editor with the currently active patch, you select the Edit Voice option in the Sound Menu. Like the librarian portion of the program, the sound

editor offers you the choice of sending changes to the synth continuously or only on special request. The Send Changes option is completely independent of the similar function in the Sound Manager, however. But again, if you want to hear real-time parameter changes as a sequence plays, you should select this option.

The actual editing screens for the various synths all make excellent use of graphics and are generally well laid out and easy to understand. They use sliders for parameter changes which can be engaged by clicking and dragging with the mouse. You also have the option of moving parameters continuously up or down in value by pointing to the arrows above and below a slider and holding down the left button. Functions which can be turned on or off are adjusted by clicking with the mouse; the display responds by making the chosen value reversed. All in all, it's a simple, effective system for making quick changes.

The envelope information on all of the editors can be viewed in two different ways: drawn out graphically or as a series of numbers. In each program only one envelope can be edited at a time, but all of them can be seen on the screen at once. To edit a particular envelope and the other parameters associated with it, you click on the small version of it and a larger version appears on the portion of the screen set aside for envelope editing. One nice feature that's included with the editors is the ability to cut and paste envelopes to other patches.

Most modules also allow you to adjust function parameters for the sounds, but these are available on a separate screen which, when selected, lays over the top of the editing screen. The same type of editing methods are used for these screens as well, but for some reason these changes are not sent automatically to the synth, even if you have chosen the Send Changes option. You have to make use of a special Send sound function. Admittedly, these generally don't have as important an effect on patches as voice parameters if you're working with real-time changes, but it certainly seems odd that voice parameters can be sent automatically and function ones cannot.

As far as specific modules are concerned, all of the editors seem to function well, but I noticed a few peculiarities with the CZ editor. First, the display allows you to combine two of the resonant waveforms (numbers 6, 7 and 8), which is not possible with a Casio synth. The LCD on my CZ101 actually displayed the same number for both of the waves when I tried this (ie. 6 and 6 or 7 and 7) but there was no audible difference. Even when I tried to combine 6 and 7, the display only showed the number of the first wave twice and at one point when attempting this the instrument

completely locked up. Another problem I noticed was that the portamento time function sometimes sent out odd MIDI data. While I was never able to recreate either of the situations, one time it sent out corresponding MIDI notes as I moved the slider up and down, and another time it sent out program change messages. Very odd.

The XSyn modules all have a Sound Creator function as well as editing capabilities and it can be reached from either the Sound Bank Manager or the Sound Editor. In either case, it automatically generates either a single sound or an entire bank of sounds based on the currently active patch. The actual parameter changes made to the sound are based on Masks which you can easily create and save. The Masks consist of a complete listing of the available parameters which you turn on or off depending on whether or not you want them to change.

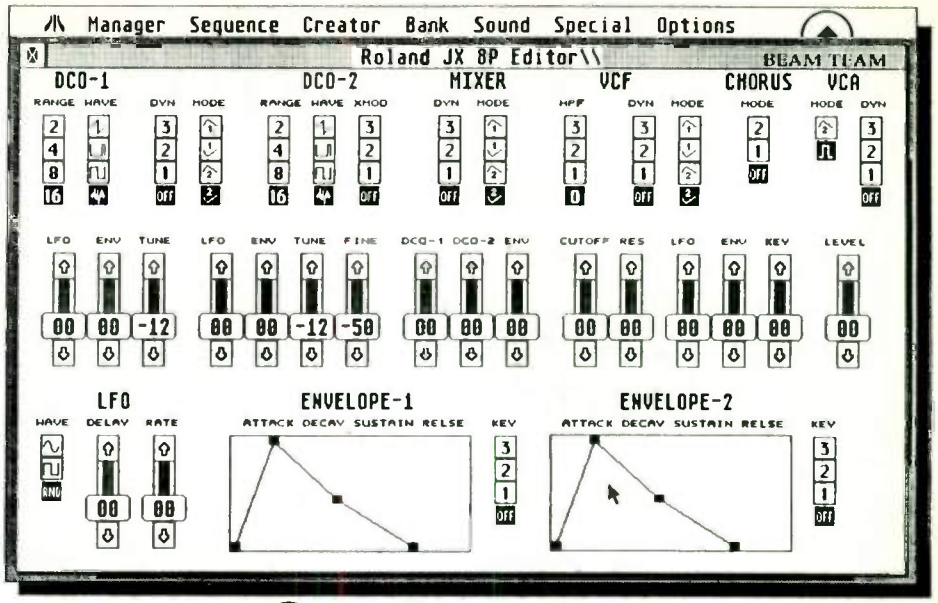
So depending on how many parameters you choose to turn on in the Mask you are working with, the Sound Creator will give you either subtle variations on the chosen patch or drastically different sounds. One other point to be aware of is that newly created patches are not automatically sent to the synth, regardless of the status of the Send Changes function. If you are working within the Sound Editor this can be a bit confusing, because the parameters on the screen will change when you create a new sound, but these changes must be sent to the synth for them to be heard.

Conclusions

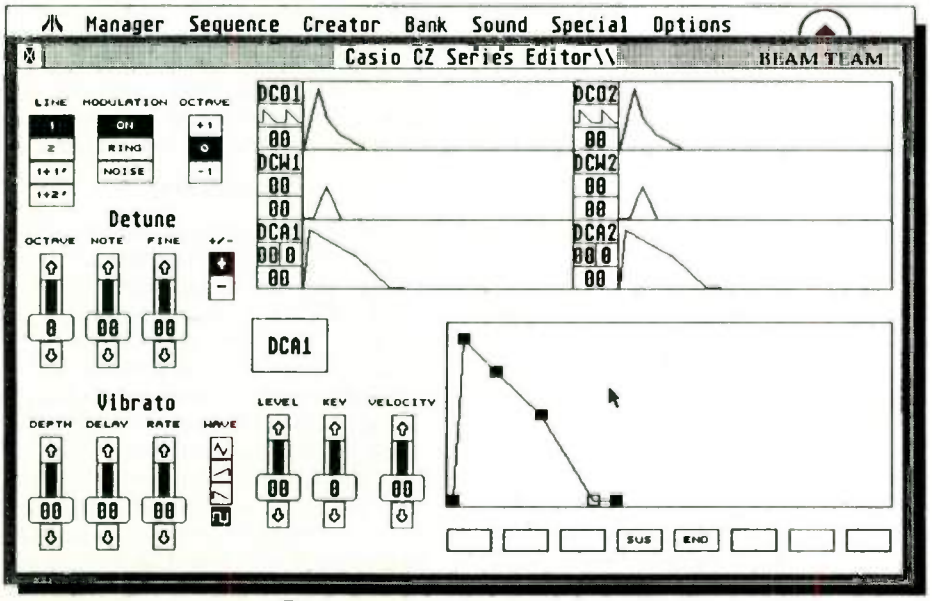
THE TRANSFORM XSYN modules are an excellent idea. A number of similar programs which all fit under one system umbrella sounds great and if you have a number of the synths for which modules are available, this program may be exactly what you're looking for. The only problem is that a lot of the more popular new synthesizers (such as the ESQI, the D50 and the DXII family) are not supported by Transform and if you have one of those instruments you'll still have to buy (and learn) a different program for them.

Similarly, interactive, real-time control over the sounds in conjunction with the sequencer is another big plus for the XSyn modules concept. Being able to hear (and see) the sound change as it plays in a musical context can be incredibly useful, not to mention very informative. Unfortunately, this part of the program is not implemented as strongly as it could or should be, but nevertheless, it indicates an improvement over the options provided on other packages.

Other parts of the program have strange quirks as well, but if you can overlook some of these oddities (which isn't really as difficult as it first appears) the programs should fulfill your voice-editing software needs. And when the complete Transform system becomes available, this could be a really powerful package. ■



BEAM TEAM • XSYN-JX8P • SOUND EDITOR MODULE



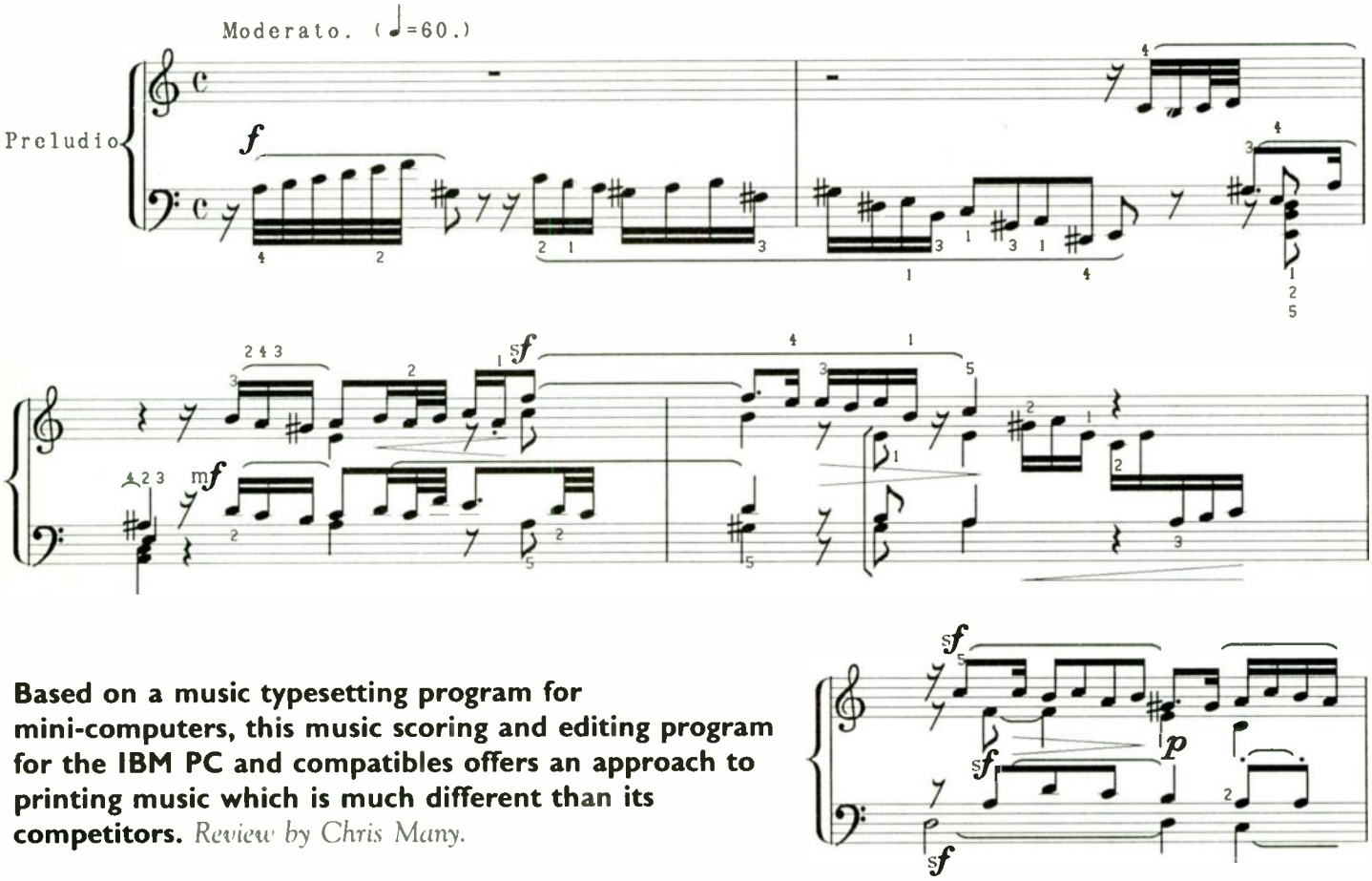
BEAM TEAM • XSYN-CZ • SOUND EDITOR MODULE

PRICE $99 per XSyn module

MORE FROM Beam Team, 6100 Adeline St., Oakland, CA 94608. Tel: (415) 658-3208

Oberon Music Editor

REVIEW



Based on a music typesetting program for mini-computers, this music scoring and editing program for the IBM PC and compatibles offers an approach to printing music which is much different than its competitors. *Review by Chris Many.*

LONG BEFORE SOFTWARE companies began developing programs to print out music scores using MIDI sequencer files, you were able to buy sheet music, collections of composers' works and a wide variety of music, all professionally typeset and published with crisp, clean notation. It's only been recently that the ability to duplicate this kind of quality has been passed along to the consuming public with products from established music software houses. But how does music get typeset by the large music publishing concerns? I didn't expect they'd be using the same programs available for the general public, but souped-up versions geared specifically for them with an even greater resolution when it came to print quality. After all, music publishing has been around for centuries and a pretty high standard has been achieved internationally.

One company, Oberon Systems, developed and marketed a system for mini-computers (the big brother of the micro-computers we're so familiar with) that combined the then fledgling breakthroughs of laser printing with music typesetting. Their system is very much in use today and includes a powerful editor and many music fonts, that essentially allow you to typeset music direct from the computer terminal and create files that are transferable via modem to print out high quality music. To be honest, I was astounded to see that what this high level music publishing system was based on was a Text Editor, where you typed in alphanumeric characters and they, in turn, were converted when you went to print into music notation. What this means is you only see the actual characters you type on the screen, *not* the actual notes, bars, ties, etc. A line might look something like this: ;'$;#;#;D;!;$;;;O;;!,LG:;; which when printed out would be the first line of a staff, plus the top part of the notes, rests and accidentals.

To learn this would seem to require a serious investment of time, but I've been told it's rather easy. Personally, I find that a little hard to accept, but even given some hours or days of memorizing which keys created which fonts, as a musician I still would want to see good old quarter notes on a screen, not a series of %%*Y";;. Of course when you do use a system such as this it gives you a great deal of control over the look and placement of music and any text within the music, which is what interests the professional publishing houses.

The Oberon Music Editor (OME) has undergone some changes and is now being made available to IBM PC/Compatible users at the micro level. The main change is that the WYSIWYG concept (What You See Is What You Get) has been fully applied, so that you will see the actual notation on your screen as you type it in.

Overview

LET ME SAY a few things up front: OME is not for your normal MIDI buyer; it is not and won't be MIDI compatible for at least a year. OME runs only on a laser printer and does not support dot matrix printers at all. It should be made able to do so within the next six months, but it's not the highest priority at Oberon right now. As a matter of fact, it only works with an HP Laser Jet laser printer; no other types of Laser printer have been tried out or any drivers developed if need be to operate them. This is a high-end product whose attention has been on quality of output, not on usability by your home recording

enthusiast or local song writers who want to print out their latest sequenced work.

OME accepts music input in two ways, direct typed entry from a keyboard or from a file sent to it over modem. Forget reading sequencer files. In short, the current market for this product is fairly small, as far as I can see, especially since the actual editor is not as powerful in its micro incarnation as it is in the mini-computer version.

Software implementation is rather straightforward but there are few options actually available for the user. The editor allows you to place symbols and notation on the screen in an accurate manner, each key or key combinations equating to a musical symbol or part of a symbol. For example, the letter "U" gives you a filled in note, the letter "K" lets you draw a stem down. The letter "Y" gives you a natural sign, etc. A series of macros (single keys that have been preprogrammed to execute a number of keystrokes) are also included which speeds up input immensely. It takes three keystrokes to create a properly sized and positioned quarter note, so the macro lets you press a combination of keys (Shift F5) and display the same figure with one stroke. Of course, if you want to erase it, it's still going to take you three strokes or more to position the cursor to erase *each* part of the figure. It would seem from a user's point of view, that an erase function is just as important as the macro feature to keep things speeded up.

The view of what you're working with is great; the symbols are big on the screen and you can scroll the page to see more of what you're creating. Once you learn which keys create what characters, the entry process speeds up, but you will have to invest the time to learn all the corresponding symbols. It's kind of like learning to type, after a while you just know where the keys are and you don't even look at the keyboard.

You can delete or insert lines on the page, change fonts (the actual notational symbols) and save the page you're working on (which you'll need to do before going to another page), and, unfortunately, that's all. Don't misunderstand, you can delete characters and edit the screen to your heart's content, combining music notation with words of different size letters and numbers, but you won't find any copy functions. Nor will you find block move, or other types of character manipulation. You will find it on the mini-computer version, and it will be implemented in the IBM version, hopefully by the end of 1987. But the product as it stands now, is really the most basic of editors, with the normal features we've come to associate with word processing in the offing.

Input *"OME accepts music input in two ways, direct typed entry from a keyboard or from a file sent to it over modem. Forget reading sequencer files."*

Printing

THE QUALITY IS, of course, excellent because it is only laser printer-compatible right now. It's fast, too: eight pages a minute, and up to 20 per minute with the newer, larger HP Laserjets. The problem is, laser printers still cost plenty. Comparing the output of OME to Professional Composer, MESA or Dr. T's The Copyist doesn't really show any astounding quality improvement, at least to my eyes; they all look great coming out of a laser printer. OME does have several fonts to choose from, and allows things like fingerings and very accurate placement of all notation that the others don't seem to have. For example, normally you have a crescendo sign that goes at one specific angle, but if you don't have room to place it between the staves what do you do? Change the angle is the obvious answer, which OME does easily. What if your ties are longer than the normal tie signs, or you want your 16th notes placed closer together? Due to the nature of OME's ►

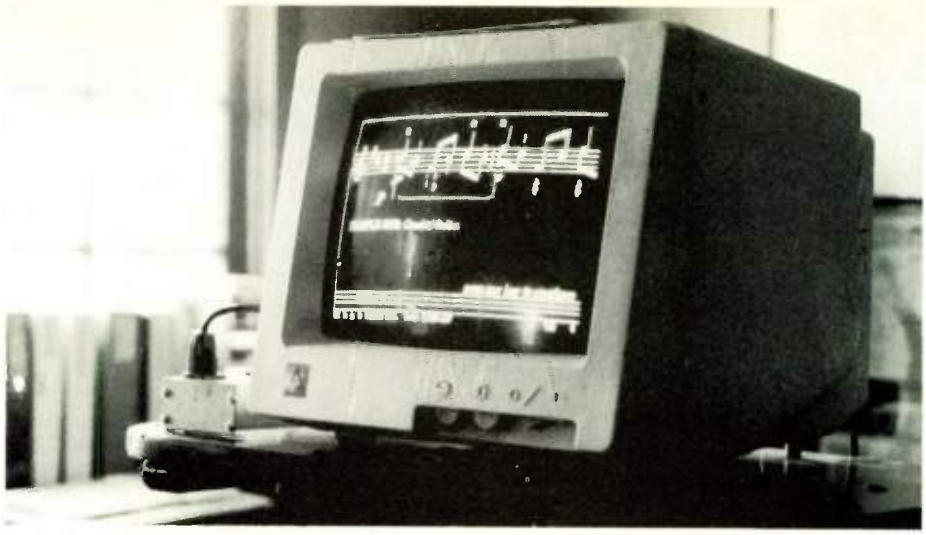
REVIEW

► programming approach, it's a simple matter to correct these, giving one a good deal of flexibility when it comes to placement of material on a screen.

Conclusions

SO, ALTHOUGH THE accuracy and placement of notation is a plus, this is a music typesetting program that is honestly an incomplete product at the micro level, more so than just something that will be updated with new features as time passes. A music editor that does not yet have copy or block move functions installed can only be called incomplete. It's a product that is not MIDI compatible and won't be for at least a year; one that will only work on a laser printer and only one brand at that; and the cost of the program is three times that of other fully functional programs currently on the market. Which brings me to the point I made earlier: this is not for your normal MIDI musician. It may be for the educators who need lots of high school band arrangements of their own, or up-and-coming music publishing companies who aren't ready to invest in mini-computers, or composers who want their scores crisply printed or other people who are fairly serious about music publishing. Oberon seems to have carved out a niche for itself at the mini-computer level, but I think they may find it a bit more competitive than they expect at the micro

REVIEW



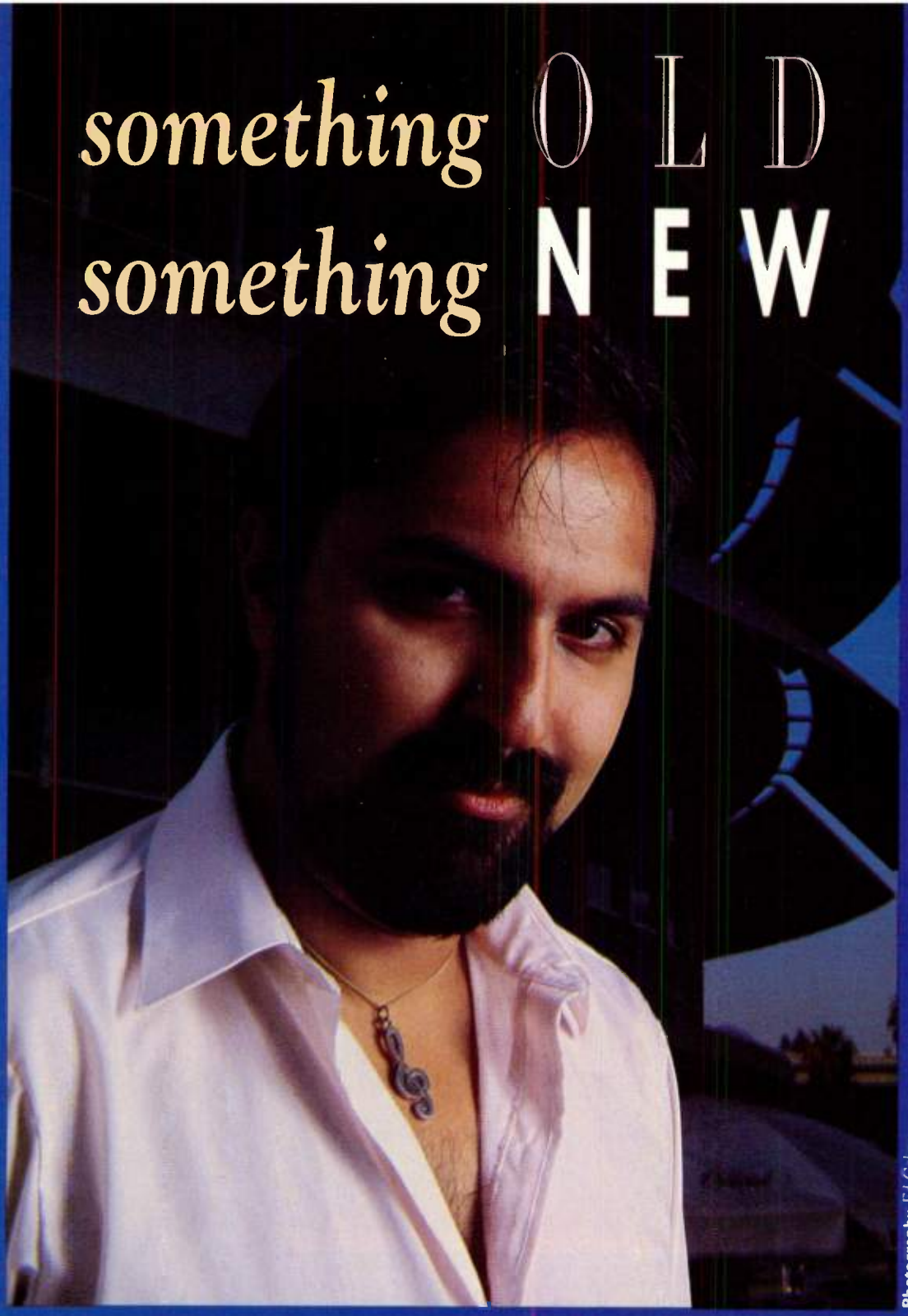
level. And for the price being charged, any music typesetting program should include every feature when it ships, not months down the line.

In all honesty though, and despite the omission of these key features in its initial release version, I must say I liked the basic approach. The manual was still in preparation while I reviewed the program, so it was easy enough to learn without any documentation, a real plus in this day and age. And it's hard to put down a product that so heavily emphasizes the quality of output, so much so that they'll keep their market smaller by excluding dot matrix printers from the initial release. The company has been in business for some years now, and they seem sincere in their effort to bring this program to the IBM world. I just think that by the time they've caught up to the rest of the MIDI'd music world, other software products will have surpassed the efforts they are making in the name of a standard music typesetting product. ■

PRICE $695

MORE FROM Oberon Systems, Inc., 3815 W. Burbank Blvd, Burbank, CA 91505. Tel. (818) 954-9591

something OLD *something* NEW



Photography *Ed Colver*

The first album by Canadian composer Amin Bhatia combines symphonic extravagance with electronic effects. The startling surprise is that he created this "filmless" film score with only a few synthesizers – and *no* samplers. *Interview by Deborah Parisi.*

THE TERM "NEW AGE" is becoming a catch-all for any kind of music that doesn't readily fall into the mainstream pop bag. From the early days of atmospheric simulation to today's championing of electronic sounds for their own sake, many have decided that if it isn't AOR, MOR, R&B or a variation on them, you should call it New Age. Well, what else can you call it?

So it isn't *too* surprising that Amin Bhatia's *Interstellar Suite* has been grouped with Cinema Records' initial New Age (aka "New Progressive") roster along with Pete Bardens, Patrick Moraz and Michael Hoenig. After all, he uses synthesizers and MIDI processors; the music is aimed at the cultural and aesthetic elite; and he obviously aims to communicate more than a toe-tapping good time. Truth of the matter is, though, regular buyers of New Age discs could be a little disgruntled if they bring this one home sound-unheard, because it takes its origins more from John Williams and Stravinsky than from Steve Roach and Kitaro. Even Bhatia is a little confused ▶

with the classification – but he's not complaining . . . much.

"I'm curious to see what kind of category they're going to give this one," he says, smiling. "There have been comparisons made to Tomita and Wendy Carlos, which is great because they've been my idols for years – but my music's more film oriented.

"I think the timing is right for the record because we now have an alternate music market. Call it 'New Age' if you will, but I'm really terrified of those two words – these days they imply alfalfa sprouts and BMWs. That's unfortunate, really, because the people who founded the whole New Age movement were people like you and me, who wanted alternative music. Now, unfortunately, we've got to do something fast or it's going to swallow itself alive. I'm grateful for New Age opening doors in the first place but now that the doors are open, what kind of a room are we in?"

Bhatia initially rejected the concept of pushing his music on the open market. "When Cinema called me, I laughed. I said, 'You're sick; I'm a film scorer; I want to do film music. Who on earth is going to listen to this stuff on vinyl?' They came back and said, 'Look. It's got dynamics, it's exciting, it's new, and we want to do it.' After looking at their concept I finally realized that between their movement and Capitol's support, if there's any way of promoting this stuff on vinyl, they can do it."

No doubt about it – the music *does* have dynamics. Soft, subtle strings nearly burst into shattering sound effects – guaranteed to blast you out of your car. Bhatia jokes about putting a warning label on the album cover to prevent home speaker systems from being blown.

"There are places in the album which are equivalent to about 128 separate tracks," he says. "Like the first sound on the album – Steve Porcaro from Toto flew me down to LA because of things like the first sound on the album, because of my sound effects. We had to bounce like crazy to get them down to 24 tracks. All the sound effects are 24 tracks, created on a separate strip of tape. You should see our track sheet! You've got Whoosh, Whine, Flanged Whine, Whoosh Return, Pre-Whoosh, Crack . . . my engineer and I had a riot with this sort of thing."

Bhatia's extensive experimentation with various instruments has created some obvious prejudices towards equipment, which vary depending on the effect he is trying to achieve.

"Oberheim gear is amazingly suited for special effects programming. The noise generator on the Oberheim is fantastic, and with all the phasing you can do with the noise generators . . . it was a combination of my Minimoog and my Oberheim that allowed me to create some of the deepest, gutsiest space sound effects that I have yet to hear. The Jupiters were very musical – it's hard to get a noise generator out of them. The DXs – well, if you're using them as part of the substance, great; but to hit a DX button and have this deep explosion, forget it.

"Anyway, once we had that effect orchestrated, it was locked away in a two-track mixed form so that we could do anything we wanted with it. The obvious would have been to store it on a Synclavier or a very advanced sampler, which I could have done; but the moment I put Synclavier or Emulator or S50 on the credits, people would immediately have thought I created other sounds – like the strings – on the sampler. So we did it the old-fashioned way, by God; we had two tracks with leader tape on it and we flew the tape in. When we were recording things back to the 24-track, we would run the tape, and when the label reached this particular part of the idler wheel, that's when we'd start the two-track machine. If the sound effect didn't quite catch it, son of a bitch, we'd do it again!"

Bhatia becomes more and more animated as he relives the birth of the album. It's unusual to see someone of his age – under 30, that is – so enthusiastic about *avoiding* the use of sampling technology. His stance resembles that of someone taking pride in a moral position.

"That's what it is," he admits, "it's a moral stance. I love samplers . . . I adore them. I've seen the S50, I've worked with the Emulator, and I'd love to own 50 of them. The point I wanted to prove though, with today's sample-trendy crowd, is that there is no substitute for sitting there and grinding it out. There is no sample in the world that can match the beauty of 20 different random string sounds coming together, and I wanted to prove that point on this first album. Now that I've proven we can do it that way, I'm looking forward to using samplers to help in the next album. I think I've gained enough credibility to experiment with samplers without copping out."

"Call my music 'New Age' if you will, but I'm really terrified of those two words – these days they imply alfalfa sprouts and BMWs."

Bhatia pauses, contemplating. "I guess what I'm really looking for is a sense of permission from symphonic players. I want them to say, 'Keyboard players should be shot, except for Amin Bhatia because he understands orchestration.' That would be great. When I buy my first sampler – whether it's the S50 or the Synclavier or the Emulator – I hope I will adhere to my promise of not having any samples that were acoustically stolen. I would like to work for hours on a Minimoog French horn sound and sample that, to make it more convenient, and to work on some of the most killer DX/JX combinations of strings and then sample that, so that even my sampler has completely electronic examples. The day I walk up to a cellist and say, 'OK, play a low E for me,' and record it on a tape – well, I hope I never have to reach that stage."

BHATIA IS NOT averse to using sophisticated sequencers throughout the composing and recording processes. He enthusiastically shares information about the intricacies of his home set-up.

"The heart of my system is the Roland MC500 Microcomposer, which is wonderful. At the time Roland sent one to me for a trial, I was hunting for an Apple Macintosh or an IBM or the latest Opcode gigamatic system to do what I wanted it to do, and every one of them looked like I had to invest $5000 and a hell of a lot of keystroke work to get at. The Microcomposer was relatively inexpensive; it was a small, tiny, cute little cuddly thing, and it accomplished what I wanted to do in a relatively short amount of time. Sure, you can't tweak with it – there are things you can't do with it – but if I want those things I can MIDI dump to a system that can."

Bhatia understands and appreciates the capabilities of his equipment, particularly the speed and ease with which it can be operated. "I've seen other people that have amazing diagnostic drawings of all their patches laid out in their room – you know, complete organization. While I like to be organized, I also like to access something quickly. In this industry, I find that to be sadly lacking. People will have state-of-the-art systems, but they need five technicians to work in the room. That doesn't create music, that slows you down and the creativity starts to disperse. After awhile, you start giving up on ideas."

Bhatia is also enthusiastic about a rather unique piece of Roland equipment. "The MKS900 is a display unit that shows you where all the notes are, which is great," Bhatia explains. "It thrills your girlfriend to see all these MIDI lights flashing all over the place – hey, there's a lot of sex appeal there! 'Wanna get a girl? Show her your MKS900.' Really, more than anything else, it's great to be able to watch orchestration and see how it's done. Sometimes I'll put in a grace note or something and not know whether I wanted to put it there. Sometimes I can't even hear it. And that damn MKS900 will quickly show me what I'm doing. Plus it has MIDI soloing on it –

it's got a filter on it. And that's great, when you don't have to reach over to your console and play with all your faders or hit the solo button. You just hit MIDI solo and you can hear one particular module."

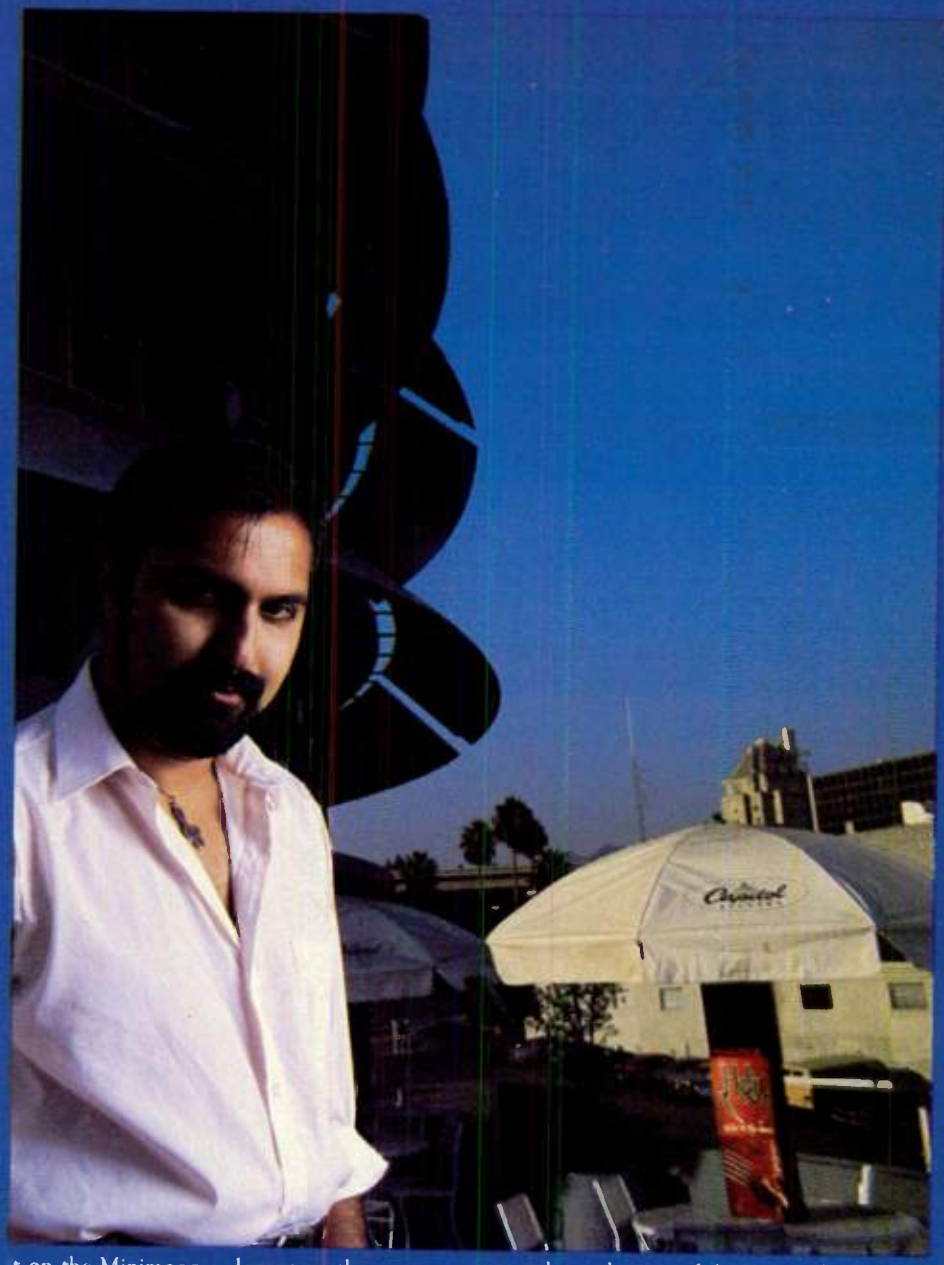
Bhatia is primarily concerned with creating rich, thick, realistic instrumentals, and he uses the tried-and-true method of combining different synths to approximate the real McCoy.

"I'm glad I'm not involved in Roland on an endorsement level," Bhatia says, alluding to the demo projects he has done for them, "so I don't feel obliged to stick with their equipment. Roland gear has a certain gut to it. The JX strings, for instance, are heavenly; but they don't have the bite, they don't have the rosin, which a DX will give you. But the DX won't give you the analog stuff. The beauty of a DX bite, with a JX horn, with a Minimoog gut – all those different combinations, played separately, played humanly rather than MIDI'd together – that's where the beauty comes from."

Bhatia uses the word Moog over and over again in the course of his conversation – in reference to sound effects, to strings, to horns, to flutes and just about any sound imaginable.

"The Minimoog is great!" he exclaims. "It's amazing how many keyboard players have come up to me and asked, 'What are you doing to make your flute sound so great? How much System Exclusive aftertouch information are you programming into the flute?' And that's not it at all. I just don't play keyboards like other keyboard players. When you're a piano player for years, while that can be amazing, it can also be the very same thing that restricts you from understanding how other instruments work. Any polyphonic patch that's labeled 'flute' is incorrect right there. My flutes come from my Minimoog, because a flutist can only play one note at a time. My French horn solo is one Minimoog, unless I'm doing a whole group of them, but even then I will play the Minimoog three times as opposed to calling up French Horn on my DX, because it's a monophonic approach.

"Of course, it makes me the worst keyboard player in the world – I'm walking with two fingers on various notes," he laughs, demonstrating his stiff-knuckled stabbing technique. "I'm not doing it out of being epileptic, though, I'm doing it out of a particular approach to the way I want the French horn to sound. On a Minimoog, when you ripple the notes, goddamnit, they ripple! And if each one ripples just a little bit before the next note hits, it sounds like what a French horn player would do. When you give a French horn player eight grace notes, or give him an arpeggio to run up the scale with, he will never play all those notes for you. I mean, he'll throw the French horn at you before he does that. I didn't learn that from a book on orchestration – I just knew that when I did



it on the Minimoog and went up the keyboard, the Minimoog sounded more like a French horn than when I did it on my polyphonic super-extravaganza.

"I have it tied into the MPU101 (Roland's MIDI-to-CV converter), which allows MIDI to play a Minimoog – oh, velocity-controlled Minimoog is great! When I jumped on the MIDI bandwagon, about a year ago, I knew I had to keep my Minimoog. My Polymoog unfortunately has died and completely come apart. I'm thinking of suspending it from a 20' ceiling, putting microphones all over the floor, turning it on and dropping it so I can sample all the amazing crash sounds. I think that would be the best way to renew its life into the world of sampling. It deserves it."

Bhatia shakes his head at the way some musicians feel compelled to buy whatever new gadget enters the market. "The way people get into buying more and more modules can be ridiculous," he declares. "It's a curse. I hear some of the silliest samples and some of the most ridiculous excuses for music. And the guy will say, 'Well *excuse me*, but it was done to a digital master tape.' And I say, 'Yeah, but it still sounds like *shit*.' You know, there's so much of a fuss. People have come down on me with, 'My God! You used analog tape!' And yeah, I'm sorry, you know, but there are things you can do with analog tape that you can't do on digital."

COMBINING THE BEST of the old with the best of the new is a compositional strength as well as a recording asset on *The Interstellar Suite*. Bhatia often locks himself away and works 18-hour days during his creative periods. He eagerly shares the nuts and bolts of that process.

"First, I created a rough score on the Microcomposer. I would work on one particular section at a time. One synthesizer would be horns, one would be strings, one ▶

would be flutes, one would be whatever. And I would just go through the arrangement, work out the melody, work out the beginning and middle of the piece – I did that in a matter of months, by sitting down, working on each section, in any order I chose. Then we went back to the massaging of each section – I guess that's the word. I programmed my first JX10 for strings, and then the second JX10 I made another set of strings. I made another string patch on the DX7, then on the Oberheim, another string patch . . . on and on until I had eight or nine different types of string sounds from various manufacturers, each one having its own personality, all played separately; no MIDI stacking."

"I'm thinking of suspending my old Polymoog from a 20' ceiling, putting microphones all over the floor, turning it on and dropping it so I can sample all the amazing crash sounds."

He lowers his voice as if sharing some deep secret, and then bursts with enthusiasm again. "That's the secret to orchestration right there – you can't do it with quantization, and you can't sit there fuming over goddamn MIDI delays. The biggest MIDI delay in the world is nothing compared to five frustrated symphony players in a room. You want to worry about MIDI delays? Put Frank Zappa in an orchestra and see if he complains about MIDI delays. OK, there are certain delays inherent to the MIDI system – sometimes you get that. More than anything else it's clogged data. Sometimes when a patch change command goes in, I'll hear some notes freak out. But that's nothing that can't be solved with a filter system – a separate filter system put in right at the front terminal of the Microcomposer so that every MIDI cable only sends that particular channel. That's what I've got to go with next.

"Anyway," he says, "to continue the thought, then I would create a string file based on my original rough score, and I would just work on the strings – solo, if you will. Picking 'em up, playing each different synthesizer separately, humanly, off the KX88 into the Microcomposer. There's my strings. I would then use the SBX80, dump that onto Track 1 of my eight-track system and save that file of strings. Then bring back my rough score and work on my horns . . . use all the synthesizers in all their different forms, program the Minimoog manually and create horns. Then I would do it for flutes, then the others. I'd work on each of these files separately, building it up on an old Otari eight-track, so I had a reference as everything was building.

"After a few months' work, I had a finished rough score at home, with all the documentation stored on the Microcomposer to recreate the entire thing again in a 24-track studio. It was economical; it was musically-pleasing for me 'cause I could experiment every step of the way. All program change commands could be stored by System Exclusive, and then it was a matter of re-sequencing the entire thing. This time on 24-track. So everything is first-generation, everything is thick. We used Dolby SR on the mixdown onto the two-track mixer, so we have all the guts of analog, all that beautiful tape-ripping stuff that analog can do, so the transfer to CD is from off a very, very finely polished, very well noise-reduced analog master."

Bhatia is a big fan of using System Exclusive commands, which the MC500 can record and play back, instead of using program changes.

"I like to stay away from cartridges and program cards and all that kind of stuff because it restricts your system ability. You can only create 32 sounds on one bank of a DX7. Any more and you've got to have the right cartridge in there for the Microcomposer to give the appropriate program change command. Every patch change that I use on my DX or my TX racks doesn't come from a program change command, it's from a System Exclusive instruction one bar before the sound is needed. The Microcomposer's great for that, and a lot of programs are doing that now, too. Whenever you hit a program change on the DX7, it sends a whole burst of information for other TX modules to receive through System Exclusive. The Microcomposer records that. So there's no bulk dumping involved before each session and it's great, because then you can work on a lot of ideas and come back to them months later without worrying about what program cards you were using."

BHATIA IS STILL undecided about the direction which he'd like to go with his music. His past successes include several Clios for radio commercials, the Canadian award for Best Original Film Score of 1986 from the Alberta Motion Picture Industry Association, and album credits on *Thriller* (Michael Jackson) and *Fahrenheit* (Toto). He would love for *The Interstellar Suite* to be presented in a visual medium, either in video or in a touring planetarium presentation, and is of course anxious to do film work. He feels that choosing a specialization within the music industry to a large extent dictates the course of your career.

"In today's technology, you really have to take a position on where you want to work and what you want to do. Do you want to be the world's best sample creator? Do you want to be the world's best orchestrator? Do you want to be the world's best synth programmer? Do you want to be the world's best synth systems organizer? I want to be the world's best musician," he says, breaking into laughter. "Huh? What does that mean? Well, it means I'm going to spend more time working on how the music comes together than on what killer patches I can create on my Xpander. Eventually I'll probably get an assistant to handle the twiddling part of it.

"Right now, there are fuses everywhere, and my direction depends on which dynamite goes off first. Depending on how this album goes, we could be doing another tomorrow. If the album needs more credentials on my part, I'll do a couple of film scores. There is even talk about getting me involved in a couple of segments of the *Star Trek* TV series, which would be right up my alley." He pauses, concerned. "But I don't want to go out there and rip off symphony orchestras; you know – when an android takes over the ships, let's use synthesizers!"

Bhatia's eyes light up as he envisions the future. "Digital workstations . . . I can't wait for that to really take over. To be able to store things on a digital domain, manipulate them, move them around, repeat them. It'd be like MIDI, you know, only better, because you're committed, it's there, it's locked in. With a MIDI set-up, you've got to take the same hardware gear and set it up all over again. With this, you'll get a digital copy. That's the next step . . ."

Whatever future awaits us – in music and in technology – Amin Bhatia will be there, reminding us that we must sometimes reach into the old technology to really create something new. ■

LET AKG GIVE YOU A HAND . . . OR TWO.

Who needs a headset mic? Everybody. Who likes them? Nobody. . . the way they used to be. But all that's changed since the AKG C 410 hit town. Feather-light, snug, comfortable on any head, almost invisible in use, virtually indestructible — tie it in a knot and it'll spring right back into shape — the C 410 is the perfect mic for guitarists, keyboardists, drummers, engineers, vocalists . . .

And no tradeoffs either. The C 410 doesn't buy comfort at the cost of quality. It offers the full-frequency response of a prepolarized cardioid condenser, is plug-ready for anything with 9 to 52 volt phantom power, and there's even a model for wireless applications.

So who needs an AKG C 410? You do. Because you've got better things to do with your hands.

AKG acoustics

77 Selleck Street
Stamford, CT 06902

SOUND READING

Music through MIDI: Using MIDI to Create Your Own Electronic Music System

by Michael Boom

FOLKS FAMILIAR WITH the reality of music stores would agree that initial MIDI purchases can be a nightmare. Uninformed or disinterested sales people can hold out the carrot of potential creativity but just don't come through in teaching and motivating buyers towards mastery of the technology. While the burden of responsibility for learning MIDI naturally rests on the musician, everyone needs someone to show them why – and teach them how – MIDI is worth the substantial effort it takes to learn.

Michael Boom's book convincingly woos the person who wants to learn into buying some MIDI cables and getting to work. He explains what MIDI can do, how it does it, and what you need to make it work in your particular situation. He assumes interest, but little to no knowledge, and uses a clear, easy-to-read writing style which belies the difficulty of his subject. He takes the mystery out of reading an implementation chart, the meaning of SMPTE, and even the binary numerical system.

Boom's approach transcends theory and connector cables, however, by providing the motivation that will get you off your ass and working with your instruments. Lengthy interviews with guys who use MIDI, accompanied by detailed diagrams of their setups and equipment lists, can show you what *you* need for your own application. Tom Scott represents the composer in the studio; David Ocker (once a copyist for Zappa) exemplifies the inexpensive at-home MIDI setup; and Tim Gorman talks about applications in live performance. In listening to their stories, you may find yourself really believing that you can do it, too.

Another surprising addition to this book is the inclusion of equipment reviews. Boom discusses the CZ101, the FB01, the Mirage, and DX7 synthesizers, as well as the TR505 and Pitchrider 4000. Discovering the capabilities of these instruments could provide you with an equipment list – or at least a wish list – for your own studio.

Apart from an error in his definition of a star network (it takes more than just a MIDI Thru box to make a star network), the only really *glaring* error hit closer to home and caused a tremor in the MT office – unfortunately, it could not be solved by referring to the MIDI specs. Perhaps you don't usually pay much attention to "Appendix C: Further Information," but I always feel obliged to read a book to the last page. And the last page of this book offers this description of Music Technology (following a few blurbs about some other magazines): "Music Technology . . . a general interest periodical aimed at the trade audience." Now, I'm not sure who wrote this blurb, but I *know* it wasn't someone who reads or knows our magazine.

Pity, really. He'll never know how highly I recommend this book. ■ *Deborah Parisi*

PRICE $19.95

MORE FROM Microsoft Press, 16011 NE 36th Way, Box 97017, Redmond, WA 98073-9717.

MIDI: The Ins, Outs and Thrus

by Jeff Rona

JEFF RONA IS the current president of the MIDI Manufacturer's Association, and his book reflects a rare mastery of the technical intricacies of MIDI mania. Beginners could easily be scared off by looking at the first several chapters – if you're totally unfamiliar with MIDI diagrams, reading "Note On, 9nH, 1001 xxxx, (from 0 to 15)" is like encountering Sanskrit. But *don't* be intimidated. The second half of the book abandons the Land of Binary and Hexadecimal and enters much more pragmatic territory.

One of the best traits of the book, in fact, is that it's a "now and later" text: you can use it now to figure out practical problems (I wish *I'd* had it the first time I hooked up a drum machine and sequencer), and you can use it later as a reference for theoretical and mathematical data.

Not only does Rona describe modes and channels, bits and bytes, and registered and unregistered parameters, he devotes considerable space to coordinating a MIDI system and problem solving. Practical advice dots most of the chapters and commonly asked questions are answered. While the bold promises on the back cover (my favorite is "Be prepared to fearlessly purchase MIDI instruments and software" – kind of like a Boy Scout motto) may not all materialize at the end of 96 pages, this book does deserve a reading and a space on your bookshelf.

Oh – I almost forgot the most important part! The book is funny. The admirable talents of illustrator Elyse Morris Wyman add a human dimension (translate, "We're all in the same bloody boat") which is usually absent from technical books. Presenting MIDI status and data bytes as cars on a train is nothing short of inspired. In fact, I'd buy this book just for the cartoons. ■ *Deborah Parisi*

PRICE $12.95

MORE FROM Hal Leonard Books, 8112 W. Bluemond Road, Milwaukee, Wisconsin 53213.

HO! HO! HO! Have a MIDI, MIDI Christmas with books from Alexander Publishing.

These great new books by Alexander Publishing make your life easier with fast tutorials. Our new MIDI Musicianship series that makes writing and arranging your music faster and easier.

How To Make Your Drum Machine Sound Like A Drummer

by L.A. Studio drummer David Crigger. Shows you both the music and the MIDI. It has over 289 musical examples fully recorded on the accompanying 52-minute audio tape. Learn pop/rock beats, realistic rock fills, anticipations, shuffles, reggae, Latin, Country, Funk/R&B, synchronizing, quantizing, MIDI note numbers, more... *Order by specific make and model: price includes tape:*

Yamaha RX5 *just $29.95*
Yamaha RX11/15 *just $29.95*
Roland TR505 *just $29.95*
Roland TR707 *just $29.95*
Korg DDD-1 *just $29.95*
Korg DDD-5 *just $29.95*
NEW! *All Make and Model Version for any drum machine now only $21.95 with one hour tape.*

Making MIDI Work By Someone Who Does... *David Crigger*

Shows how to set up and make all this fabulous equipment work. Starts with 2 synths and a MIDI cord and takes you through sequencing, how to hook up and make drum machines work, thru boxes, audio and the advanced MIDI studio. Non-technical, easy to read. Also works well as a basic buyer's decision guide. *Now just $9.98.*

The Murphy's Law MIDI Book

by Jeff Burger.

The only troubleshooting book for the MIDI studio. Takes you beyond the basics to the next level in plain English. Has a one chapter basics review. Next come troubleshooting tips, advanced applications like: patch mapping (a very usefull tool thing for recording), MIDI Controllers, system exclusive transfers, sequencing concepts, sync-to-tape, the complete advanced studio. Lots of helpful graphics. more..*just $19.95*

Make sure you have a very MIDI CHRISTMAS by placing your book order right this second!! Call 818-704-8777 9AM to 5PM Pacific. LAST order date for ontime Christmas Delivery is December 11th. So do it NOW.

Programming Books

These books are recommended by Yamaha, Ensoniq, Korg and others.They're simple and easy to use. Check 'em out.

Yamaha DX7II *supporting E! Update now only $21.95*
Yamaha DX7S *Supporting E! update only $19.95*
Yamaha DX21 *just $19.95*
Programming the DX27/100 *now just $14.95*
MT1X Multitrack Recorder *$17.95*
Yamaha TX81Z *$19.95*
Yamaha FB-01 *$9.98*
Ensoniq ESQ-1 Vol.1 *$19.95*
Ensoniq ESQ-1 Vol.2 *$19.95*
Roland Juno 1 *$16.95*
Roland Juno 2 *$16.95*
D50 Right Now *$9.98*
D50 Programming Book *$19.95*
D50 64-Patch Book *$21.95*
Korg DSS-1 Sampler *$34.95*
Korg DW8000 *$19.95*
Korg DS-8 *$19.95*
Oberheim Matrix-6 *$19.95*
Casio CZ101 *$14.95*

MIDI Musicianship Books

These books help your recordings by giving you the musical techniques you need to make the music sound even better.

101 Songwriting and Composing Techniques *by Jack Wheaton.* These are the top techniques used by all the major writers. When you're done, you'll have over 2,000 techniques at your dispoal.*$9.98.*

Freshman Year *by Peter Alexander* How to build chord progressions (covers enough to do any rock song you want), effective background pads, 4-voice arrangements, writing for string ensembles, building strong bass lines, more.. with 2 hours of recorded music. $39.95

Synth Keyboard Method *by Jack Wheaton* Vol 1 with one hour audio tape Well reviewed in Music Technology. Build keyboard skills for sequencing, arranging concepts on the synth, playing sounds like the acoustic instrument, improv: learn voicings and chord scales, more. *$24.95*

Synth Keyboard Method *by Jack Wheaton* Vol. 2 with one hour audio tape Covers triplet and 1/8 note arpeggios, special left-hand voicings for each chord, left-hand voice leading, playing from a lead sheet, applications to arranging. *$24.95*

For fastest service call our office. (see inset box)MC/Visa.. Add $2.50 Shipping and handling per book. California residents add appropriate sales tax. Or Write **Music Make Publications, 7361 Topanga Cyn. Blvd, Canoga Park, CA91303. Allow 2-4 weeks for delivery.**

Title **Price**

_____

_____

_____

_____

Total + Ship + Tax _____

Name: _____

Address _____

City _____ **St** ____ **Zip** ____

LIVE AT LAST

With **Time Commander**, you can run your technoband any way you like, from any sync code you've got, including yourself, and you can be sure that your sync translation problems are over.

Using real time technology we developed for our Master Beat synchronizer, **Time Commander** listens to YOU and passes that feel along to your technoband. . . no more rigid performances! Your technoband will now pay attention to tempo changes AS YOU MAKE THEM – – WHEN YOU MAKE THEM, and they can be different every night! Your music will breathe again, computers and all, and so will you.

Time Commander also includes all the features of our original Doctor Click! Just connect a click track or any sync code and you'll get MIDI and six simultaneous sync outputs. And if your unit plays late in the studio, no problem, **Time Commander** allows for Lead/Lag, and the adjustment can be trimmed as the piece plays.

At Garfield we know that getting your band to come in on the right beat is just as important as staying in sync, so **Time Commander** provides adjustable count in. And since we know that the owner's manual is seldom where you are, we've printed ours right on the top.

Now, for all of you who have noticed a little something extra on the front panel, for a bit more (and you can add this option later) **Time Commander** syncs to SMPTE *and* MIDI Time Code!

- Live sync to drums or MIDI note data
- Live sync Pause, Restart & Restore modes
- Dual inputs read sync from kick *and* snare
- Internal metronome can operate as master
- Outputs 24, 48, 96, 384, Sync R, Sync K, Click & MIDI
- Sync to 1/4, 1/8 & 1/16th note click tracks
- Sync to Clock, FSK & MIDI
- SMPTE/MIDI Time Code generate, read & translate
- Bulletproof
- Made in USA

Time Commander

Standard Model: $750.00
SMPTE Option: $150.00

For the location of your nearest dealer call or write:

Garfield Electronics

P.O. Box 1941, Burbank, CA 91507 • (213) 434-6643

Our Only Business is Getting Your Act Together.

RSF SD140
Sampling Drum Machine

From France comes a black box that aims to unite the world of the drum machine with that of sampling. Is the fusion a successful one? *Review by Dan Goldstein.*

STROLL INTO YOUR local music store and chances are you will see at least one example of that mid-eighties musical instrument phenomenon, the sampling keyboard. You will probably see several. And you may well get a glimpse of their close cousins, stand-alone sampling modules that communicate with their controllers via MIDI.

But what you probably won't see – unless there are some major instrument launches between my writing this and you reading it – is a sampling drum machine. Simply because there aren't many costing much less than $3000.

If you have access to that sort of money, then one of the current glut of samplers/drum machines/sequencers – anonymous boxes that do the job of several units but are still, in essence, sampling beatboxes – can be yours. The E-mu Systems SP12 and the Linn 9000 were the first of this breed, and now they have evolved into the E-mu SP1200 and the Akai/Linn thingy that keeps changing its name; both also have the widely used Sequential Studio 440 for company.

However, if three grand sounds like stretching things a bit, your choice is strictly limited. Exactly *why* the choice is limited is something I'll come on to later.

You could try to make do with the fairly rough eight-bit sound quality of the Casio RZ1, and have change from $600. Or you could move up-market to the Korg DDD1, add a sampling board, and blow the best part of $1300.

On the other hand, you can go French, and opt for the machine under review here. It goes by the instantly forgettable name of RSF SD140, costs about two-and-a-half times what the Casio does, but is still comfortably cheaper than the big guns mentioned above.

RSF first became known in Europe nearly a decade ago for their Kobol and PolyKobol analog synthesizers, much used (and abused) by the likes of Depeche Mode, Vince Clarke and so on in their quest for Better Bleeps. Since then the company has diversified successfully into digital drum boxes, and in technological terms (and perhaps marketing terms, too) the SD140 is the logical conclusion of that diversification. Here in the US, the RSF is being distributed by Paul Real Sales.

Before we go any further, I ought to stress one point. The machine we were sent for review was in a somewhat "preliminary" state. Its software was susceptible to crashing (in the end, things got so bad that an EPROM was replaced with a new, modified one), and its documentation comprised a French manual accompanied by photocopied sheets of (pretty rough) English translation, some which were stapled together in the wrong order. A quick phone call was necessary to establish the SD140 as a 12-bit machine with a maximum 30K sample rate.

Happily, things have now straightened themselves out. The software is finished – though undergoing frequent and worthwhile ▶

► updates including the addition of support for MIDI song position pointer data. And all the machines now in the stores come with a more comprehensive (and coherent) English manual.

Format

REVIEW

THE MACHINE'S DESIGNERS – by which I mean whoever was responsible for shaping the external aesthetics – are obviously not a very imaginative bunch. More than anything, the SD140's appearance reminds you of a MkI DX7 that's had its length cut by two-thirds and its keyboard lopped off with a hacksaw.

The similarity is more than skin-deep, too, since those banks of pale green and lilac switches are of the same membrane type as their DX7 counterparts – which means they aren't anything like positive enough to inspire confidence in use. Yamaha has seen the light here and fitted proper "clicking" switches to their MkII machine; let's hope RSF follows suit.

As well as the membrane switches, the machine's front panel offers ten more push-switches for voice selection. Two slider controls are located in the top right-hand corner, a two-segment LED display to the left of them, and a rather small LCD screen to the left of that.

All in all, quite a sparse front panel. As you've come to expect, the plain-looking panel design may help things look neater and keep component costs down, but it's a guarantee of operational nightmares to come.

The rear panel is another matter. Here you'll find a comprehensive assortment of connections – perhaps more important on a drum machine than on any other of today's techno-toys. Nine individual voice outputs (on quarter-inch jacks) plus a mix out for those not endowed with multi-channel mixing boards are available as are a single dynamic trigger input; tape sync connections that also act as the means by which sound and pattern data is saved to (and loaded from) tape; MIDI In and Out; a footswitch connection and last, but crucially, the sample input socket.

But back to the front, if you see what I mean. The usual problem with having too few switches doing too many jobs is that designers have to resort to a multi-layered system of "pages" through which the user has to rummage to find the correct function.

The SD140 is no exception to this rule. Its particular multi-function book has four pages, and most of the switches have a different job to do in all four of them. Broadly speaking, the pages correspond to four different families of functions. The first deals with pattern and song programming, the second with input and output assignment, the third with MIDI functions, and the fourth with sampling.

Each page is given its own color in the front-panel legending, but although the system is quite a logical one as multi-function arrangements go, some of the operational procedures it necessitates are none too elegant – as you'll see.

Specification

AS IMPLIED ABOVE, the rear panel's nine separate outputs are not, in fact, capable of handling all the SD140's voices at once, for the simple reason that there are 14 of them. These are preset sounds, mounted in ROM and therefore immovable. They cover a fair range, with a bass drum, a snare, a rimshot, four toms, two hi-hats, a cymbal, three assorted Latin Percussion voices labeled "Perc," and a set of handclaps.

In parallel with these, however, is a bank of RAM sounds that represent the spaces into which samples can be loaded by the user. Thus there are 14 zones – each one corresponding to a preset sound – into which external sounds can be loaded.

But the RSF makes life a bit more complicated than that. For a start, the sample zones are not numbered 1-14, as you would expect. Instead, those corresponding to the machine's four tom presets are numbered 1/1 to 1/4, while those equivalent to the closed and open hi-hat are numbered 11/1 and 11/2. In between, we have sample zones 5-10, and afterwards, zones 13-14. Now, there is a reason for these eccentricities, and it is this: the zones that share a common first number also share output circuitry, so only one sample within each group can sound at any one time – in just the same way that, among the preset voices, only one tom and only one hi-hat sound can play simultaneously.

"More than anything, the SD140's appearance reminds you of a MkI DX7 that's had its length cut by two-thirds and its keyboard lopped off with a hacksaw."

A further complication lies in the fact that since each RAM zone corresponds precisely with its ROM equivalent, not all the zones are of equal length. In fact, they vary dramatically, from a few tenths of a second (the four toms) to over six seconds (the cymbal).

Yet although this system is no substitute for true dynamic allocation of memory – as found in fully-fledged multi-samplers like the Akai S900 – it does at least offer the kind of versatility you need if you're sampling a variety of different percussion instruments. For contrary to popular belief, sticking to percussion (sorry) does not mean all your samples are going to make uniform demands on sampling time, bandwidth, dynamics and so on.

Full marks to RSF, then, for devising a memory allocation system which offers a degree of flexibility, without being too unwieldy (those numbers excepted) in use.

In many ways, it actually makes sense, if you're recording (say) a bass drum, to assign it to the bass drum zone (number 8). Not just because the memory length available should be just right for the sound itself plus a little post-decay ambience, but because there is also a small selection of editing controls available on the SD140 – each one tailor-made for the sound to which it's allied, and each one usable for both the factory ROM sound and the user-sampled RAM equivalent.

The bass drum, tom and cymbal zones all have filters for altering the timbre of their sounds, and the manual (yes, even my copy) gives you a guide as to the "correct" bandwidth and VCF values you should see in the LCD after your tweaking. The open hi-hat, meanwhile, has a more straightforward decay control – useful enough and a piece of cake to use, despite having to press umpteen buttons to access the function in the first place.

On the pattern programming front, the RSF is conventional to the point of being uninspiring. A few nice touches are available, like the Mute function, which allows you to shut an irritating instrument up for the time it takes you to get a pattern just right, without having to lower its level manually. But overall, it's down to the usual patterns-linked-together-to-form-songs scheme of things, with the SD140's internal memory capacity being 100 of the former and 30 of the latter.

Programming can be in real time or step time, and since this is a European machine, there seems to be no bias either way: the two systems are as comprehensively equipped as each other. Before you do any real-time programming, you have to select parameters for volume and accent levels (both are variable per voice and per pattern, but not per beat), length of measure, tempo, auto-correct (quantization) and swing. Swing can be altered while a pattern is playing back, but quantization cannot, so if you're not sure how a different rhythmic resolution will affect the "feel" of your pattern, you have to make an educated guess.

In addition to recording a pattern from scratch in step time, you can also apply step-time editing (to alter things like accent levels) to a pattern that was initially recorded in real time. This is good news, if hardly revolutionary.

A variety of erasing, copying and chaining functions assists in the creation of songs, which cuts down on the donkey-work necessary if you want to use a pattern several times at various points in a song, or if

you change your mind and decide you don't want a particular pattern in there at all. Again, this is hardly revolutionary, though the RSF does take a step ahead of many of its rivals by allowing you to alter the *tempo* of each pattern within a song, as well as program the tempo of the song as a whole.

Patterns and song data – as well as sample data – can be stored on cassette tape. The usual save (confusingly titled "Punch"), load and verify functions are provided for this, but it's no more a convenient and reliable means of data storage with the RSF than it is with any other digital instrument. Roll on the CD ROM.

Alternatively, pattern data can be sent from the machine via MIDI System Exclusive data. No MIDI implementation chart was included with the review instrument, but I presume it can be done, so owners of universal MIDI disk drives (such as Yamaha's MDF1) should be laughing.

MIDI can also play a part in synchronizing the SD140 with the likes of sequencers. Here the machine can act as either master or slave, and the usual 24ppqn MIDI clock is implemented, as is the run/stop command – though I could find no mention of such niceties as song position pointers in the preliminary manual.

Alternatively, you can use the tape sync option mentioned earlier, use the RSF's ordinary internal sync as master, or use an outside instrument's sync to do the same job – in which case you have a choice of 24, 48 and 96ppqn resolution.

Operation

THIS REALLY IS the SD140's Achilles Heel. Even if my manual had been one of the simplest and most comprehensive written, I'd have found working my way through the machine's multi-faceted control system a pain. And I think you will, too.

It's not that things are illogical. Or at least, not most of the time. But I can't help thinking there must be a simpler way of adjusting one instrument's level than pressing the Page Select membrane switch (the only control to have only one function, incidentally), holding down the switch for the instrument in question, and using either the "master" slider or a set of increment/decrement buttons to do the necessary tweaking.

Sampling is even more problematical. In common with most samplers, the RSF gives you a choice of manual or automatic triggering, but either way, you have to do a lot of button-pressing to coax the machine round to your way of thinking. The whole process is supposed to take five seconds, but I generally needed more than that to enter the correct page, select my chosen memory zone, adjust the triggering threshold, alter the pitch, and press Start. Only the last of those actions involves pressing a single button.

Things become less logical (and therefore even more frustrating) when it comes to choosing output and MIDI channel assignments, or adjusting individual volume and accent levels. Here, you have to enter the right page, select the voice you're working on with one of the membrane switches (as opposed to the row of "instrument selectors"), and make the necessary adjustments with the good ol' "master" slider or inc/dec switches. Most other drum machines in this category simply let a dedicated slider do all the work, so why not the SD140?

"RSF's engineers have paid much attention to the SD140's sampling section but they've underestimated the importance musicians attach to preset sounds."

The LCD screen does excellent work endeavoring to inform you of things like input level during sampling, and the LED does likewise for showing tempo values during pattern programming, but no peak level meter is going to be very reliable if it has only seven segments, and no tempo readout should show "..40" when it should really mean "240." In short, the displays are just too small for the complexity of information they're trying to convey.

Conclusions

LET'S RETURN TO our original question. Why, precisely, are sampling drum machines so thin on the ground, at a time when sampling keyboards and modules are in abundance? If my experience with the SD140 is anything to go by, the reasons are threefold.

First, if you're going to make a sampling drum machine, you must also give it a range of preset drum voices. This is in sharp contrast to the people who make sampling keyboards, who seem to have no trouble selling instruments that have no sounds in them at all. The RSF's sampling quality is excellent but its preset sounds are pretty lackluster – usable, but not earth-shattering. Obviously, RSF's engineers have paid much of their attention to getting the SD140's sampling section right, but in doing so, they've perhaps underestimated the importance musicians seem to attach to preset sounds.

Second (and leading on from the first point), if you're going to sell a sampling instrument as a *drum machine*, it must have its fair share of drum machine-style programming options. What the SD140 offers here is fair enough, but nothing like as sophisticated as the range of tricks a talented programmer can perform with the Yamaha RX5, which costs less. It should also be pointed out that the RX5 offers a more convenient storage medium (RAM cartridge) onboard, and doesn't involve the user in anything like the operational mystery tour that the RSF does, though it *doesn't* offer user sampling.

Third, if you're going to make sampling the crucial feature of a drum machine, you must ensure there's a good range of sample-editing features available onboard. Here, too, the SD140 falters. A couple of different memory lengths and a sprinkling of filtering options are no substitute for proper dynamic allocation or a set of looping, truncating and reversing functions. Comparable multi-samplers – even some that cost less than the SD140 – offer most of those possibilities plus a few more besides.

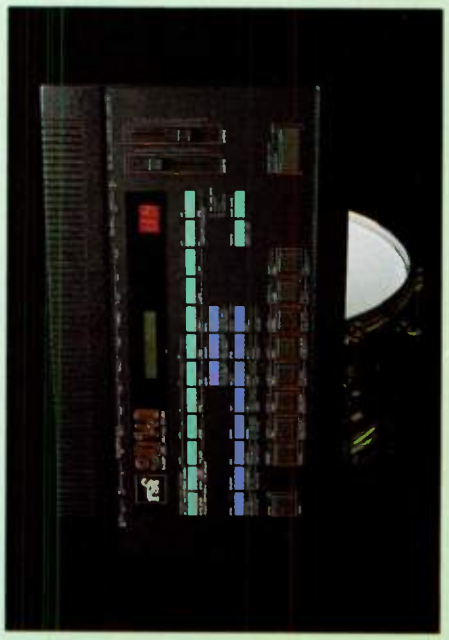
So maybe that explains why so few companies have entered the sampling drum machine fray. Then again, the lack of competition makes the SD140 unique in its combination of beatbox and sampler programming options. I can certainly envisage committed drum programmers opting for this route into the world of sampling, rather than trying to use an instrument that shatters their preconceptions and forces an entirely new way of working onto them.

If the combination of facilities suits you – and you can tolerate the machine's occasionally eccentric behaviour – the SD140 is well worth investigating. It strikes me as being to musical instruments what Citroëns are to the auto industry: a machine that performs a distinctive set of tasks uniquely well, without setting the world on fire.

And though Citroën may never win a Grand Prix, they have always had their admirers. ■

PRICE $1399

MORE FROM Paul Real Sales, 1507 Mission St., S. Pasadena, CA 91030. Tel: (818) 441-2484



a new approach to sound synthesis

Gibson USA

KEYTEK

Cross Table Sampled Instruments

How Cross Table Sampling Works

Keytek professional keyboards use an advanced signal processing technique, Cross Table Sampling. Going beyond sampling, CTS starts with a digitally recorded signal and uses external digital signal processing to modify and compress the original sample. The data is then put into wave tables. Keytek CTS instruments can read from two tables, or cross tables. This gives you all the sounds in the wave tables plus all the sounds from combinations of wave tables. Post sample processing, combined with a powerful resynthesis method using cross tables, allows Keytek CTS instruments to store a complete library of sounds in the space normally required for one sample. These sounds are all in the Keytek ROM*. This means more sounds "on-line" ready at an instant.

*Read Only Memory

Features of the CTS-2000

- 61 key (C to C) Velocity sensitive keyboard
- 6 voice with 2 signal sources per voice
- 320 wavetables in ROM for instant recall
- Analog signal processing on each voice, VCF and VCA each with a 7 stage envelope generator
- 8 data entry sliders and full alpha/numeric display for one of the most interactive, user friendly, editing systems found on a synthesizer
- "Polytimbric" up to 5 zones of 1 octave each
- 48 user programmable presets in internal memory
- External data cartridges
- Full MIDI implementation, each zone completely assignable
- Stereo outputs, with 5 different panpots on every preset

Who is Keytek?

Keytek was started over 10 years ago by a group of engineers who believed the only way to succeed was to build high quality instruments. Since 1976, Keytek engineers have concentrated on designs for real world musicians using sophisticated electronics to provide instruments that are reliable, servicable and affordable.

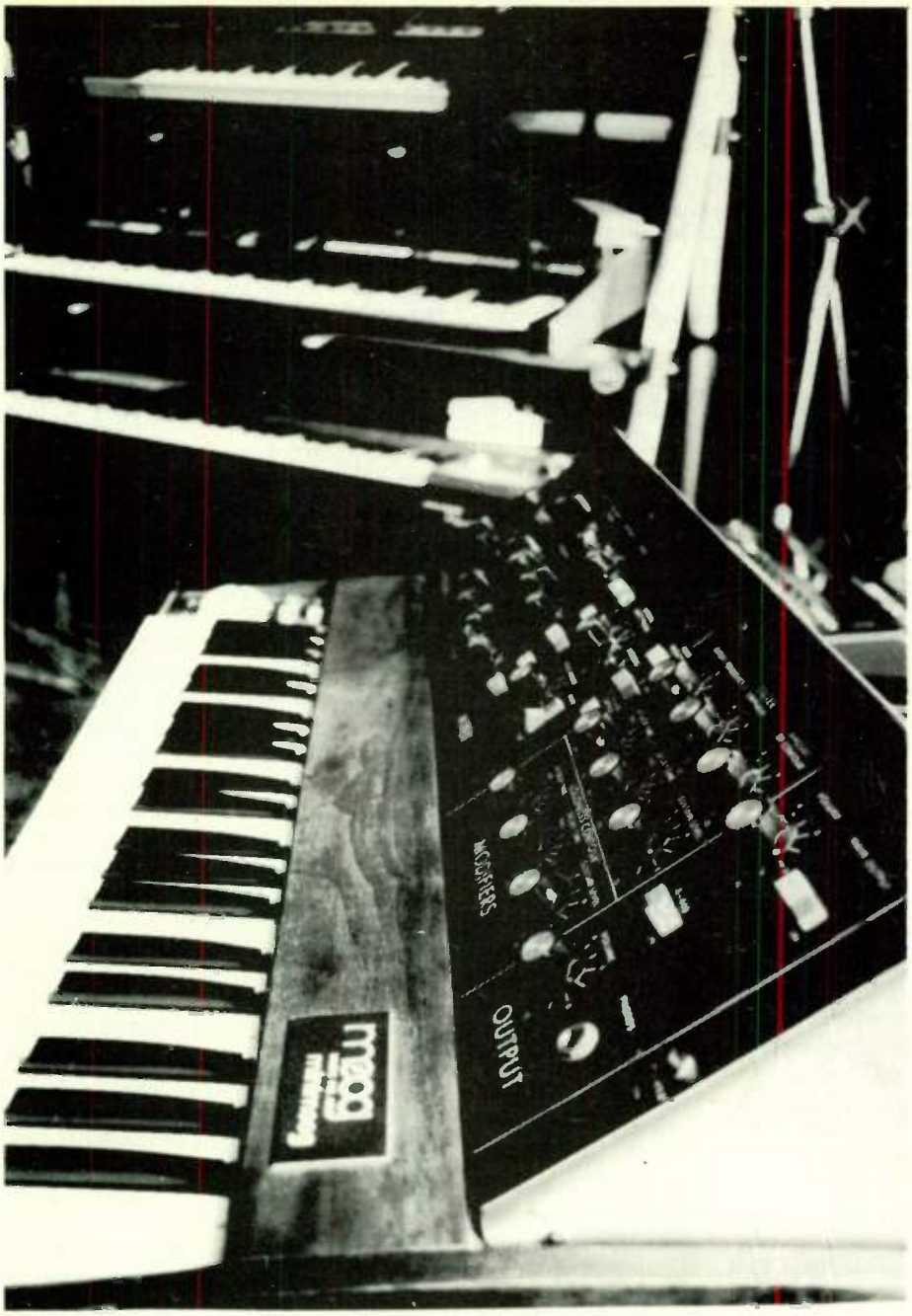
KEYTEK

THE KEYS TO THE FUTURE

ALSO NEW FROM KEYTEK

CTS-5000 (Digital CTS Piano)
CTS-400 (CTS Power, Low Price)

Available only at authorized Keytek Dealers



The Art Of LOOPING

part three

Want to give your old synths a new lease of life? Sampling could be the key. Here are some guidelines on applying theory to the real world. *Text by Chris Meyer.*

CREATIVE SAMPLING TECHNIQUES was supposed to be the topic for an article I was going to do for this magazine. It got pushed back a bit because of the decision that such an article would require some background information (hence the creation of this series). But now that I've covered the basics of looping in the last two issues, it's time to discuss the original idea: sampling and looping sounds for our own uses.

Rather than provide you with hard and fast rules about how to do things, though, I will attempt to relate what works for me, after two years of practice, and hope that it helps you find your own methods.

I usually do things in batches – I'll set up all of the memory, then do all the samples, then trim them all, etc. Aside from it just being a strange kink in my brain, this method gives me the opportunity to keep comparing all the samples of a sound side by side, and ensures that I don't miss a step in the process on any one of them. So I'll be covering all the various topics associated with sampling in the order I'd actually do them.

What to Sample

BEFORE WE DELVE any further into the actual sampling process, it's as good a time as any to talk about sources for samples. One angle, of course, is to try and recreate one's own disks of traditional instruments, such as cellos, pianos, and horns. This requires getting in the close physical proximity of the instrument (and preferably someone who knows how to play it) with microphones and a good recorder. Local university music departments tend to be good sources of the former; the latter are more items to spend your money on and learn. The process certainly guarantees unique samples, and is ultimately satisfying – particularly when it actually works right. If one wishes to take a shortcut courtesy of someone who *has* gone through the bother, more and more cassettes and CDs of sounds intended for sampling are coming on the market. For those making a first pass at trying to record their own acoustic instruments, try recording your own voice sustaining a vowel sound – it's as good a place as any to see if you're up for the task, and sampled vocals and choirs always seem to be in vogue.

Another source (particularly for percussion fanatics) is to sample anything around the house or apartment that makes a good sound when struck, bowed, caressed, or whatever. I personally think *anything* dropped an octave (slowed down to half speed) or farther sounds great. More margin for "error" exists because it is rare that someone will say, "Aw, that's a lousy sounding cake pan – Brand XYZ's factory disk is much better" – uniqueness is ▶

► on your side. No doubt you have been driven as crazy as I with cohorts that go around shouting "Sample that!" whenever you accidentally make a sound. Well, I have started doing sound effects work lately, and developing ears along this line has become particularly handy.

The one source that I am most fond of is sampling other synths. Part of the large turnover in synths is due to the never-ending quest for new sounds. However, another significant reason people tend not to hang on to older synths is that it is financially impractical to have one of each synth around – particularly if it's just for the three or four great sounds that it makes. Throw in the lack of programmability and polyphony, drifting analog oscillators, flaky keyboards and their generally *unhip* stage appearance, and one can see why perfectly valid instruments (or at least a few perfectly valid sounds) get thrown into the closet.

"While it may be tempting to equalize everything to death right now before the sample in jack, I tend to be more cautious and get the straight sound first."

Samples of these instruments saved on disks, however, gives them all the modern features one expects. Synths are great to sample, because they're easy to play (depress a switch masquerading as a key) and record (just plug in), and are easily borrowed from friends with weird passions, or from music stores. For that matter, old analog modular synths can be bought for a dime on the dollar these days, and are an endless source of new sounds.

Preparing the Sound

BEFORE SAMPLING A sound, you have to decide first if there is a way to play it that should be used or avoided. A lot of this has to do with what capabilities you have to change the sound after it has been sampled. Is there a handle rattling on that pot you're going to hit? If it's in a frequency range anywhere near where the rest of the sound is, wrap it up or tape it down now – no filter in the world is going to get rid of it later. The same goes for hiss or AC hum out of a synth – try to cure it now. Any other equalization at this point is fair game too, but while it may be tempting to equalize everything to death right now before the sample in jack, I tend to be more cautious and get the straight sound first. I'm usually able to scare up a second sampler from a friend for a day if I want to re-EQ it later. Otherwise, take at least two samples – one dry, and one "megadosed" with the graphic.

Does your sampler have an adjustable filter or one that can be altered with an envelope? If so, sample an overstruck version of the sound (ie. hit it a bit harder, or open the filters a bit more on the synth). You can always tame it down to some degree later, but rarely can you juice it back up. Also, truncating a bit off the start of the sample later on is another safety net to compensate for a sound struck too hard, so go ahead and stress it out.

What type of vibrato, tremolo, or other washes to the sound exist? Remember that they will become a permanent part of the sample, and transpose all over the place along with the pitch of the sample. Things such as a Leslie or the even wash of a phase shifter on a string synth don't transpose well; natural, random, and/or complex modulation (such as oscillators beating slightly) does. What sort of memory and looping facilities do you have laying around? Sounds other than pure, vibrato-less tones are hard to loop without crossfade looping, and the more complex the modulation, the longer the loop it requires. You may be best advised to strip it off first.

This same rule applies if you're thinking about sampling a solo instrument or a whole ensemble together – the more instruments playing at once, the more complex the beating going on, and the harder it will be to loop. Some samplers even have a chorus, layer, or detune facility to thicken sounds back up. If you have crossfade looping and/or lots of memory, however, sample the best sound you've got now – it'll only come out better later, and will be far more complex and natural than you could recreate with your sampler's own facilities.

Two special notes to those who are indeed sampling other synths, and whose samplers have synthesizer-like envelopes and the like: 1) remember that you can recreate many of your amplifier and filter envelopes with your sampler, and 2) be wary of trying to recreate many of your amplifier and filter envelopes with your sampler. Huh? Well, this one deserves a bit more attention. Keep in mind that transposing your sample all over the place will also change how long it takes to play back. Consequently, your envelope rates will be getting sped up and slowed down along with the sample's pitch. Dropping a sample an octave cuts the speed of all the envelopes in half; raising it an octave doubles them. If they are simply amplifier envelopes or gross filter envelopes (like an exaggerated wah or sweep), keep them wide open when you sample and use your sampler's facilities to recreate them.

On the other hand, keep in mind that your sampler's envelopes and filters may not *sound* like the synth's own, or may not do things that your synthesizer's would. You don't hear people rhapsodize about the Minimoog's oscillators; they rhapsodize about its unique filter. Many older synths have dedicated envelope-generating hardware that produces exponential curves which sound a lot more natural than the linear ones on the majority of current samplers. Also, many older synths have features such as filter modes and inverted envelopes that just don't exist in most samplers. In this case, it is better to use the synthesizer's envelopes and filter(s) when sampling, and on playback, make the sampler's as neutral as possible.

Before creating too many disks or raw sounds, try the following experiment. Program a sound on your synth that has a fairly interesting and complex oscillator setup and a slow, full filter sweep up and down with a touch of resonance. Sample that and play it without using an envelope on the sampler itself. Next, sample an equal length of just the raw oscillator sound with the filter and resonance off. Then try to recreate the original sound with the sampler's envelopes and filter. This will give you an idea of how close you'll be able to get. Chances are you'll find that the envelopes are fine but the resonance doesn't sound right. In any case, at least you'll know what you can and cannot get away with.

If you can get by with using the sampler's own analog (or digital) processing, you'll have a little more work to do, but over the long haul you'll use less memory (loop just a bit of the raw oscillator sound) and will end up with samples that are more consistent across the keyboard (no envelope rates transposing all over the place). I'm personally very particular about a synth's sound and am usually not offended by transposing envelopes in light of capturing that sound as accurately as possible.

Last, remember that quite often you will never hear the natural release of the sound from your sampler. Most samples tend to stay in their loop and rely on their own envelopes to fade them out. Using a sustain-only loop (see Looping Part 1) with the sound's natural release afterwards can be done as an alternative, but this eats up more memory, and it is often difficult to articulate the release of a key precisely with when the loop is at its end (otherwise, the remaining part of the sustain loop will play, and *then* the release).

Sample Rate and Length . . .

. . . IS A FAIRLY basic topic when it comes to sampling, but here are a few tips to add to whatever nuts and bolts knowledge you may already have.

Of course, the higher the sample rate, the higher the bandwidth – that is, the higher the frequencies that hang around in the signal after sampling. Most samplers have a nominal sampling rate around 30kHz, which translates roughly to a

bandwidth of 12kHz – bright enough particularly for old subtractive synthesizers; perhaps not so for some of the newer digital meanies. Because higher sampling rates do eat more memory (the land of compromises), I quite often sample all but the highest notes around 30kHz, and occasionally that highest one at 40kHz or so – it saves a bit of memory for looping and evolution in the sound. Simple, no?

But it's "the exception to every rule" time, again. If you transpose a sound downwards, you're transposing its bandwidth down, too. Playing a sound sampled at 30kHz down an octave drops the bandwidth to a dull 6kHz! Low note samples are the ones that are most commonly stretched the furthest down, so some consideration may be paid to sampling these around 40kHz instead.

The real issues are how much memory you have, how much of the keyboard you plan to cover, and how long the evolution of the sound is. Very few natural instruments or synthesizer patches sound good over five octaves, but there is a constant impulse to fill the whole keyboard anyway (I fall victim to it easily too). This is particularly funny when one takes into account that many old lead synths had only a 3 or 3½ octave keyboard! If you can resist the urge, and know you're only going to play it in the two bass octaves (or whatever), you can balance the decreased number of samples off against longer ones. In most cases, though, the sampler's transpose range won't gracefully cover the entire intended range and more than one sample will have to be taken. Tuck in the back of your mind right now that samples tend to transpose down better than they transpose up (chipmunk effect), and remember this when it comes time to pick which pitches to sample at.

I play in a studio environment instead of a live one, so I don't mind having only one or two different sounds per disk. Thus, I listen to roughly how long the sound takes to evolve from its attack to a steady state (for looping), and decide then if I'm going to try to fit it into half of the memory or if it's going to require all of it. Also, the evolution of the sound and its loop is more important to me than how well the seams between samples match up because I play more sustained chords than quick, long runs.

"Sounds other than pure, vibrato-less tones are hard to loop without crossfade looping, and the more complex the modulation, the longer the loop it requires."

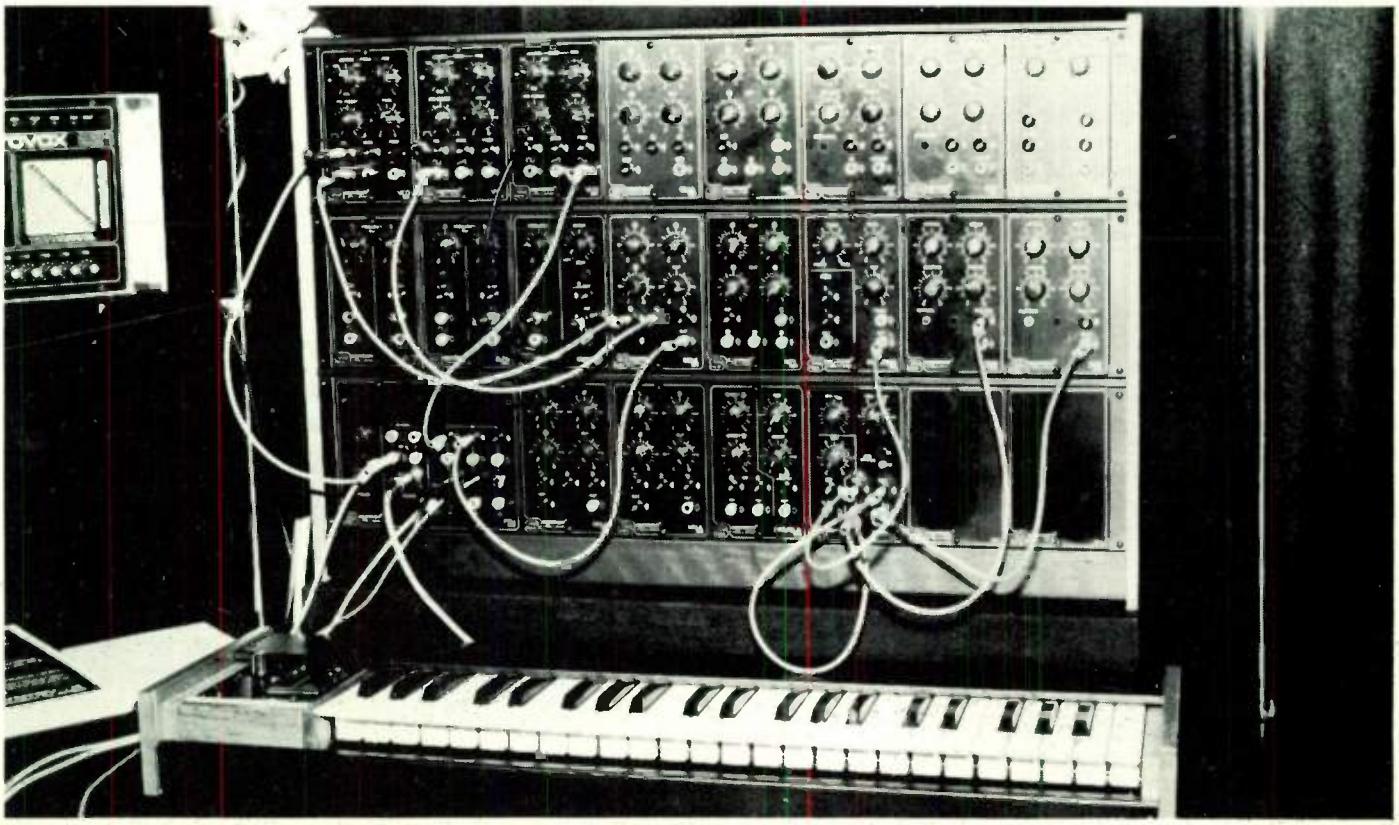
As for the number of samples I need, I try to use four or five per five octaves – four if the sound has a particularly long evolution that I want to capture. For four samples, I divide the memory in fours, place the lower three samples at the three C's centered around middle C, and the fourth around the high F or G to try and lessen the transposition problems. For five samples, I divide the available memory into fives, place them at the five A's, and count on each one to cover their octave. Those who want to hide the seams and transposition effects between samples will have to take more samples placed more closely together.

Two last adjustments are often necessary – first, if the sound seems to evolve more slowly at lower notes than higher ones (the envelope tracks the keyboard, or the beating is slower), I'll make the lowest sample about 20-25% longer and the higher sample that much shorter to let the low notes "breathe." Second, I always take a bit more sample than I think I'll need – it'll come in handy for crossfade looping (see Looping Part II), and acts as a general safety margin.

Sampling and Trimming

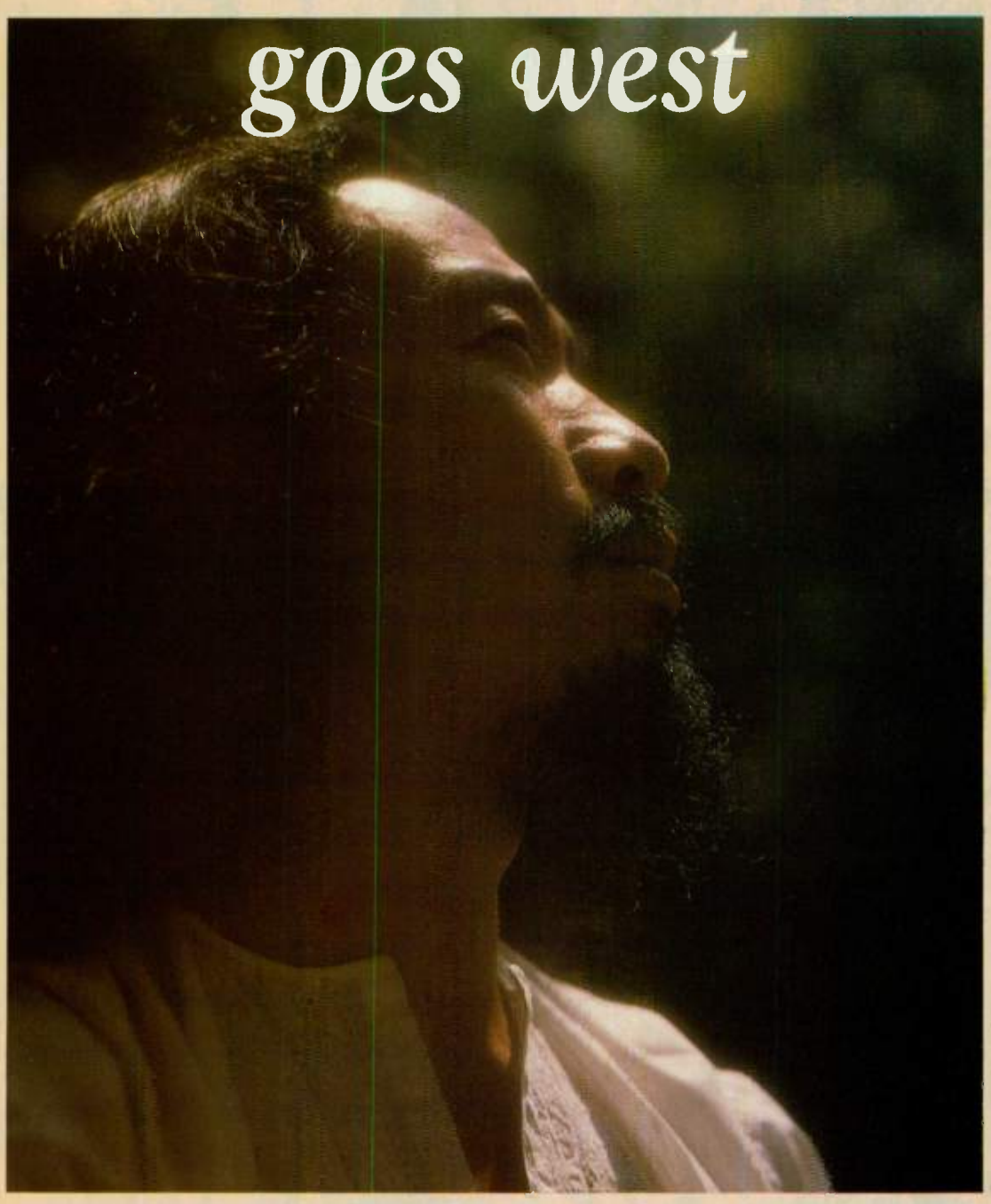
AFTER ALL OF those hard decisions have been made, it's time to start sampling (yeah!). Whatever you do, *don't clip*. The digital clipping of a sampler is far, far nastier than anything you want gracing even the briefest moment of any sound – other than the click at the start of a drum. True, the louder you record a sound, the far better it will be later, but clipping is the gun beyond the door – listen to your sampler's clipping indicator, and listen to the sample itself right after you've taken it. If it doesn't sound right, resample it – you won't get the chance later when it just doesn't sound quite right. And don't be fooled into thinking you can edit it out later – as much as I like visual editors, I haven't been able to convincingly smooth out a clip yet.

There are several other good reasons to listen to a sample right after you've taken it. An obvious one is checking for crackling ▶



—KITARO—

goes west



Photography *Peter Figin*

If you're not already familiar with this enigmatic figure in modern Japanese music, then it's only a matter of time before that changes. His most recent steps into the west include a new album produced by the Grateful Dead's Mickey Hart, and his first US tour.

Interview by Rick Davies, Translation courtesy of Rittor Music.

ANYONE WHO REMEMBERS listening to Kitaro's music in the late seventies is likely to remember how pleasant a change it was to hear his records on PBS stations at a time when progressive music was on its way "out" and New Wave was on its way "in." Electronic instrumental music was a rarity on the radio, and Kitaro's *The Silk Road* was still available on import vinyl only. Digital drum machines hadn't left the R&D labs yet, and digital sequencers were rarities.

The nature of Kitaro's music hasn't changed very much since then, despite the technological advances which have polevaulted many a keyboardist into the realm of the producer. Kitaro, after all, has been doing what he wanted to all along – making music. He is careful with the intent he puts into his music, and a lot of people like it. That some find his style repetitious, or even dated in many ways, is understandable; but there is an honesty and consistency in his music which people just seem to trust.

It is this peaceful nature that has made

his entire back catalog of interest to listeners of modern instrumental music (I refuse to say the "N" word). And right that it should, because of all the musicians producing electronic music, Kitaro is certainly one whose very lifestyle is in harmony with his music.

Kitaro is open about his early influences. He started playing guitar in R&B bands in high school, and eventually took up keyboards when he joined the Far East Family Band in 1970. By 1972 his encounters with Klaus Schulze had shown him new roads in music, and by 1978 his own brand of "impressionistic" music was receiving critical acclaim in Japan.

Here in the US, however, his music was available only on Japanese pressings through import record stores until Geffen Records distributed re-releases of seven of Kitaro's works in 1986: *Astral Voyage*, *Full Moon Story*, *Millennia*, *India*, *Silver Cloud*, and *Asia*. These were followed by the re-release of *Towards the West*, and finally, the 1987 release of Kitaro's first new album on Geffen, *Tenku*, which made a brief appearance on the Billboard pop charts.

Now news comes from Kitaro's publicists in New York that his latest album, *The Light of the Spirit*, is the result of a collaborative effort with the Grateful Dead's drummer Mickey Hart, and that several parts of it were recorded at San Francisco's Fantasy Studios. Next comes news of Kitaro's plans to tour the US in the Fall. Kitaro has assembled a new band for the tour, which he describes as "almost the same as my group in Japan. Two drums, one guitar, and three synthesizers."

Surprisingly enough, Kitaro found his new band members ("not rock 'n' roll," he muses) through auditions in Los Angeles. The new ensemble consists of Ken Park on percussion, Casey Scheuerll on drums, James Behringer on guitar, Stephan Kindler on keyboards and violin, Brian BecVar on synth, Steve Bach on synth, and, of course, Kitaro on keyboards and wadaiko.

For Kitaro this represents the first strong promotional effort to reach a larger US audience. With an entire catalog of domestic releases, Kitaro is no longer a name exclusively associated with the import bins. And while he is certainly gaining ground in the US, his success is not due to any change in his creative output. His music retains its characteristic calmness.

MEETING KITARO AT his management's west coast office confirmed what many have already said about him; he is as pleasant a person as you could hope to meet, and while his spiritual image is certainly not the result of a publicist's plan, he is as down-to-earth as anyone I've met.

About half of the time, our translator spoke with Kitaro in Japanese, but being the personable type of fellow that he is, Kitaro on many occasions would break off from Japanese and address me directly in English. Accenting his often hanging sentences with pauses, quizzical facial expressions, and the occasional "Umm . . . ", Kitaro puts his thoughts across effectively.

"I think it was a concert in last year's Christmas or New Year where I went and heard the Greatful Dead for the first time. At that time I was introduced to Mickey by a friend of mine."

As it turns out the Dead aren't strangers to Kitaro's music. "They have listened to my music a lot, and their children know me as well. Mickey's son is named Taro. That is if you take 'Ki' off from my name, you will get 'Taro.'" Ki is also the title of one of Kitaro's records, which adds yet another source of amusement. "All of the Dead families listen to and like my music. In their office, my music is always played. Also Jerry Garcia, the guitarist, likes my music."

And so the story begins. Kitaro started recording *The Light of the Spirit* at his home studio in Japan, and then came to the US to finish the production with Mickey Hart at San Francisco's Fantasy Studios. "Mickey has been using the studio for a long time, and he is used to the studio, and it is easy for him to work in it."

Obviously Mickey Hart must have had a good deal of control over the drum production, and as Kitaro puts it, he even added some to the instrumentation of the new recordings.

"He played just on two cuts. He has three big drums, named 'Beast.' They look like very big floor toms." Again Kitaro emanates joy as he recalls Hart's drums, possibly because they remind him of the wadaiko, a huge drum which Kitaro hopes to bring into his next collaboration with Hart. "He doesn't use it – yet – but next time he will use the Wadaiko. My Japanese drums' name is Japanese Beast!"

Kitaro's use of rhythm falls outside the course of most electronic composers; this has much to do with his view of *all* types of music and rhythm as having their ties. "In general, I really like to use rhythm. Whenever I travel, like to South East Asian countries, I buy percussion instruments. When I use those instruments in my composition, the Western rhythm does not fit too well. So I use the rhythm to fit the instruments. When I write Western-style music, I would use the instruments to fit the composition. However, in live concerts, I often use big Wadaikos – three of them – along with Western-style drums. My records and live concerts are different."

Saying his records and peformances are different is not just an understatement, however, because in addition to performances with his own band, Kitaro is well known in Japan for extended ceremonial performances on the wadaiko. But this year, his August 12 performance is going steps beyond prior years.

"I will start beating my drums at the outskirts of Mt. Fuji beginning at 7:00 in the evening 'til the sunrise of the next morning. It's a continuous beating of 11 hours. This time, my members will join me, I think. I do this on the day of full moon in August every year. But since I am here this year, I will have to postpone it a little bit.

"I lived in Mt. Fuji for a while a long time ago, and it is like a show of my appreciation to the mountain and a prayer for peace. This event has grown bigger and bigger every year."

This ceremony is something Kitaro began about seven years ago, and each year people come from all around to witness the event. "What is happening now is that many drummers in Japan join me to do the same thing at the same time all over the country. I would like to see this event done globally. For example, at the same time I start drumming, drummers around the earth would start doing the same thing with me regardless of the time zone."

SPEAKING OF TIME gaps, Kitaro's next move may well be to the United States, where he plans to set up a new home and studio. As Kitaro describes his current home studio, it becomes clear that the environment in which he works is very important to him.

"Sometimes I open the windows while I'm recording. As I'm playing sitar and acoustic guitar, from outside the windows bird singing comes in, oh . . . " Kitaro pauses, recalling the sound, " . . . so nice." His mountain home is at an elevation of about 3500 feet, and this is something he is looking for in the US as well.

"I like mountains, so I'm looking for mountains for my studio. I don't know where, but one of the villages in Northern California, Mendecino, is the sister village of Miyasa. Miyasa, 'Beautiful Canabis' village. My village in Japan's name is 'Beautiful Canabis.'" Kitaro laughs.

While we're on the subject of setting up his new studio, Kitaro brings up a few of his favorite instruments. "I am using Kurzweil. Kurzweil and many, many of Yamaha's DX5 or 7, and the Casio FZ1." The latter being one Kitaro enjoys MIDI'ing to his Kurzweil.

"I like analog synthesizers. Do you know the Korg synths? I use the Korg 700S. It's the oldest one. It's nice, *nice* stuff. It's a cheap one, a small one, but it sounds *good*. They'll never make another synthesizer." The 700S has become one of Kitaro's signature sounds, and is featured extensively from album to album.

"The Yamaha factory people try to make this sound for the DX7. You can try, but never . . . " laughs Kitaro. ►

Photography courtesy JLM

► Kitaro also knows how to have a good time. In addition to his interest in American football, he takes great pleasure in building, firing and sampling fireworks.

"I have a Macintosh connected to the Kurzweil for sampling. I'm interested in sampling – real-sounding sampling. So fireworks have a low end, and I go sampling – wow! Lots of times I've sampled big fireworks called 'Sanjaku Dama.' Its diameter is about three feet.

"I take two mic stands, and headphones, the sound is just quiet – pure – it's great!" Kitaro uses Sharp and AKG mics almost a quarter of a mile from the fireworks for a distant sound.

"I have a license to build fireworks. If I am allowed to do here, I would like to include my fireworks in outside concerts. There is one fireworks that you hold on the side of your body with your arm. This is dangerous. The temperature will go up to around 1000 degrees, I think."

Kitaro's production is characterized by a certain regularity, much of it stemming from his preference in sounds and synthesizers. "I like pure sounds you can see through." Still, Kitaro indicates that those things which define how well an instrument fits a piece of music are not just the features. "You can use the same instrument, like a DX, to play rock 'n' roll and also my kind of music. I think it's the way of how you use it rather than the instrument itself."

With programmability such a common feature nowadays, the ability is there to use each instrument in several roles within a single piece, but Kitaro prefers to keep the roles clear from shifting too often. "It will change in each composition, but within one piece of music, I set a synth on strings, another on rhythm and so on." Kitaro features a female vocal on *The Light of the Spirit*. "I think voice is the best of the instruments. So this time I used voice – a woman's voice.

"When I compose, I do not like to spend long hours in a day in a studio practicing. When I feel good on a certain day, I just go into the studio and concentrate. Then I would record the ideas that come up then. After that I would pick up some of those ideas which do fit in a piece of music I am writing then."

Though many other electronic musicians are trying to do it all themselves with sequencers and drum machines, Kitaro doesn't seem to fall into that trap.

"My recordings never use drum machines, I play it alone, myself." Even when it comes to recording the classic analog sequencer-style patterns, Kitaro finds an alternative to automating his music. "I like real-time recording. Sometimes I get the tape machines at half speed," he fingers a slow arpeggio on air keyboards, "go back to normal speed and it plays back faster."

Rather than use sequencers to sketch out his ideas, Kitaro prefers to get it onto tape as ideas come to him. "I use sequencers in my live concerts. When I am recording I always play real-time and record directly on a tape." Obviously that requires a lot of tracks, and in the case of recording *The Light of the Spirit*, Kitaro went for the ultimate home studio. "This time, 32 tracks. Digital.

"Digital is better, because for real sounds . . . gongs and strings – real strings – digital is much better. But synthesizers are *electric*, just on *line*, so I think analog and Dolby sometimes good. This time we mixed down both ways – digital mastered and Dolby SR. Dolby SR almost same as digital, but Dolby has *punch*.

"Last time, we mastered Dolby SR and digital, so I cannot decide – they almost sound the same. If the sound is different, I can choose, I can decide. But it almost sounds the same." Kitaro depicts the scene that took place, "I told the engineer, Tom Fry, 'Which is best?' He can't answer. So 'OK,' he says, 'Toss coins!'" So they tossed coins and the decision was to go digital. So much for A/B comparisons.

This story is a typical example of how Kitaro's success affords him the ability to experiment and try things at random. And Kitaro seems truly appreciative of his ability to make music in the settings of his own choice. "Nobody comes to my studio – only me, my equipment, and a lot of snow. I need a quiet place. I was so enjoying it. Every record is recorded in my studio." *The Light of the Spirit* took four months to record, and "almost all of it was recorded in my studio. The vocal, Mickey, one of the guitars, and the indian flute – four sounds were recorded at Fantasy Studios."

Kitaro seems to have enjoyed recording in the US so far. "Fantasy is a big studio, so I used a Neve – the faders move!" Still, Kitaro is no stranger to automation, and in fact does automated mixing at home . . . "I use different type of computer mix. In my house is a Sound Workshop 38 to 32, and an Otari 32-track digital DTR900. Brand new."

Kitaro finds it amusing that the new Otari machine actually costs more in Japan than it does in the United States. "There is big difference. So now, how to buy some equipment from *here*."

KITARO IS EXCITED about touring the US: "this tour is the first time in the United States, so I just feel try, try, try.

"So also, with this new album, I got a new try. Because before, I never used vocal, but this time I could try the vocals and other musicians. Before I was making music alone, but this time there are other musicians – Indian flute, guitarists . . . "

Unlike a lot of "east meets west" music, Kitaro's music doesn't seem to force the matter. His feelings about musical styles explains why.

"I am not too conscious about whether I am playing Oriental music or Western music. I think my music and other people's music are all worldwide music.

"I like rhythm and blues." But could Kitaro ever see himself playing in so different a style? Kitaro muses over the idea of gigging with the musicians he's met recently. "Now, Grateful Dead and Bob Dylan together, making a show. So someday I hope to make concert with Dead and Dylan.

"I think they're the same, R&B and now this 'Kitaro music.' I think the *same*." Kitaro prepares to make a final point which underscores nearly everything he's said throughout the interview. "The human is sometimes sad, sometimes laughing – *many* situations, so I think music is *music*. So R&B and rock 'n' roll and Kitaro's music . . . " he pauses to check that I'm following him, "the *same*." ■

You Asked For It

Professional musicians like to call 360 Systems to talk about their projects, and to tell us how they've used our Midi products. We've been particularly interested in what they had to say on Midi Bass, because after all, they use it every day in studio work, and we don't.

So 360's Design Group made up a musician's "wish list" with all the new ideas that our studio friends suggested for a new generation of Midi Bass.

Here's What the Pros Wanted:

- **Make it rack mount.** We did. 1¾" high, AC powered, no "wall adaptor."
- **It should hold more sounds.** We agree; now it holds 16.
- **Make the filter and loudness programmable.** We did.
- **How about "zones" for different sounds on the keyboard?** Good idea. Pro Midi Bass has two programmable zones, with two sounds on each.
- **Velocity Crossover™ from one sample to another?** Absolutely. Now choose any two samples, and change at the velocity of your choice.
- **A "pro" product needs a good display of all its settings.** So we've included a full two-line LCD display that tells all. It even looks great.
- **Decay and Release should be programmable too.** And now they are.
- **Keep the price down! I've got enough $1,800 samplers.** We've done better than that. You could buy *three* Pro Midi Basses for the price of a sampler.

A Good Idea Gets Better

Professional Midi Bass keeps the best features of the original. Like being able to play the instant a patch is called—no disk loading time, no lost disks, and no waiting. And multiple-sample bass sounds, with the realism of the original instrument on every note.

You don't have to be a professional musician to use all the new features on Pro Midi Bass. But we think you'll agree that the concept is entirely professional. We can't show you everything on Pro Midi Bass here, so visit your favorite music store and see the next generation of a classic.

18730 Oxnard Street, Tarzana, CA 91356

360 Systems

It's Here

MADE IN U.S.A.

COMMODORE AMIGA
the alternative micro

TECHNOLOGY

Macs, STs and IBMs may be the most popular computers around, but they're certainly not the only ones. Our report clues you in on the past, present and probable future of the Amiga. *Text by Stefan B. Lipson.*

IF YOU'RE A musician researching a personal computer, it's important to know your options. The Macintosh, Atari ST and IBM PC have been receiving a lot of press lately, but there is one microcomputer alternative that has not been widely discussed: the Commodore Amiga.

The first of three Amigas, the Amiga 1000, was released in October of 1985. Touted as the most sophisticated of personal computers, it included independent co-processors for graphics, animation and sound. *Byte Magazine* called the graphics and audio "dazzling." Musicians and artists were thrilled; finally there was a machine designed with the musician/artist in mind.

And then Commodore seemed to drop the ball. In a less than brilliant move, they dismissed virtually the entire Amiga development staff and let the Amiga drift in the marketplace. Third-party developers were leery of Commodore's half-hearted support, so software was slow in coming.

Since then, there have been some changes made. Commodore dismissed some higher-ups (The Prez is gone; long live the Prez) and released two new models – the Amiga 2000 and the 500. It seems that Commodore has finally thrown its support behind the machines; and in turn, third-party developers are providing additional products and support. After two years in the market, the Commodore Amiga has come to represent a solid alternative to the Atari/Mac/IBM options. What follows is a brief summary of the features that the Amiga 500, 1000 and 2000 have to offer.

System Basics

The basic specs on the Amiga line are pretty impressive. All of the Amigas use the Motorola 68000 microprocessor (the same 16/32 bit chip that is used in the Macintosh), running at 7.16 Megahertz. That's pretty fast, especially compared to the IBM PC/XT's eight-bit 8086 chip with its sluggish 4.77MHz clock.

The Amiga series includes a set of custom co-processors for graphics, animation and sound. For musical applications, that means that an additional chip is dedicated to processing audio information, leaving the 68000 free for other tasks.

The Amiga's hardware capabilities make this machine a music synthesizer in its own right. All of the Amiga series machines provide four-voice polyphonic sound with stereo output, two voices per channel. The machines generate sound at frequencies up to 7.5kHz, representing a nine-octave range. While a MIDI interface is not provided, one is easily obtainable for about $50 and can plug right into the serial port.

Voice Synthesis

The Amiga also includes ROM routines that allow for text-to-audio voice synthesis. Given a text input file, it uses the ROM routines to enunciate what has been written in the file. A number of different parameters can be adjusted, such as the pitch of the words, the speed of enunciation, and the gender of the speaker. The AmigaDos operating system even includes a "say" command which allows you to make the machine speak from the command line.

The Interface: Take Your Pick

The procedures required of a user and the screens through which those procedures are entered can loosely be called the user interface. Two basic interfaces are available on micros today; the icon-based interface, originally designed in the mid-seventies at the Xerox/Parc Research Center and made popular with the Macintosh, and the command line interface, which is found in both Unix operating environments and the MS-DOS based IBM PC/XT and AT. Both of these interfaces have advantages and disadvantages as well as supporters and critics.

The Amiga, however, lets you pick either interface. If you don't want to waste valuable RAM on an iconic interface, you can use the command line interface (dubbed the "CLI"). If you simply want to access applications software without having to roll around in the proverbial computer dirt, you can use the "Workbench," Amiga's version of the Xerox/Parc

interface. Changing between interfaces is easy, so you're always afforded the best of both possible worlds.

Multitasking

Another unique feature of the Amiga is its ability to execute more than one job at a time, commonly referred to as multitasking. Multitasking is analogous to washing the dishes and mowing the lawn simultaneously. With AmigaDos, you can have graphics animation running in one window as you edit an accompanying music score in another window.

The Amiga 500: Unique Features

The Amiga 500, which began shipping in June of this year, is selling a lot better than hotcakes.

In effect, the 500 is an upgraded version of the original Amiga 1000. The CPU and the keyboard of the 500 are combined into one unit. With a $649 list price, the system includes the 68000 CPU, a two-button mouse, and 512K internal RAM, expandable to 1Mbyte. You can also expand RAM up to 9Mbytes externally. A 3.5" 880K disk drive is built into the side of the unit. The Commodore 2002 monitor, which at this moment lists for $499 (it may have been lowered as this went to press) is sold separately. While the Commodore monitor does offer stereo outputs, there are better (and cheaper) monitors available. Package deals for the system reduce the cost considerably.

The 1000

The Amiga 1000 is the original Amiga. It differs from the 500 in that it has a detachable keyboard and a larger power supply. Unfortunately, RAM can only be upgraded internally to 512K, and then to 9Mbytes externally. With the introduction of the 500 and the 2000, this machine is being reduced in price and will probably be discontinued as the newer models take hold of the market.

The 2000: A Lot of Bang

With a list price of $1999, the Amiga 2000 is a powerful, open architecture, hybrid machine. It provides all of the features that the 500 and 1000 offer, plus a 68020 expansion slot (with a socket for a 68020 co-processor) and seven internal expansion slots which – get this – may be configured as Amiga slots *or* IBM XT/AT slots. Commodore offers this option to accommodate its new 2088 Bridgeboard, an optional plug-in board that allows the 2000 to fully emulate an IBM. The PC emulation allows you to run much of the software available for the IBM, so that you effectively have the power of both an XT and an Amiga. Combine that with AmigaDos's multi-tasking capabilities and you get an MS-DOS application running in a window under AmigaDos (whistle as you exhale). An awful lot of possibilities are available there.

The Amiga 2000 also comes with 1Mbyte of RAM and two built-in, 3.5", 880K disk drives. Like the 500 and 1000, it uses the 2002 monitor, which is not included in the package.

Software Options

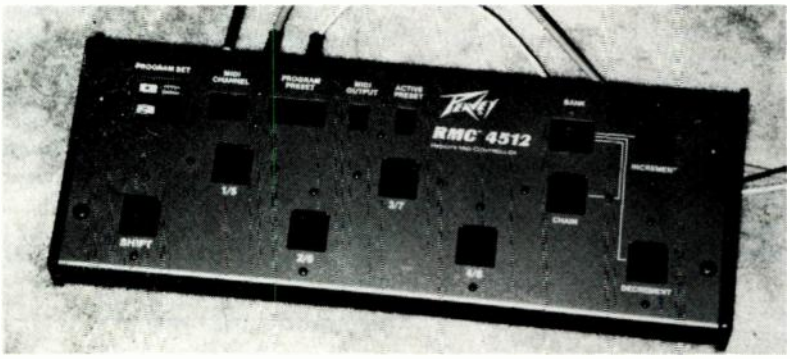
Of course a machine is only as good as its software, and a number of different music programs are available for the Amiga. Texture, written by Roger Powell for Magnetic Music, has been released for the Amiga. Sonix by Aegis is a MIDI-compatible program that includes scoring and real-time playing capabilities. Mimetics Corporation has Soundscape, and Micro Illusions of California is releasing Music X, which includes patch editors, librarian capabilities, and a variety of other functions. The list of available programs continues and the list continues to grow.

The Bottom Line

The Amiga microcomputers are professional, low-cost systems affording the user capabilities which other systems do not. Four-voice polyphonic sound, stereo outputs, voice synthesis, multi-tasking, and IBM emulation are some of those capabilities. If you are investigating a microcomputer system, be sure to check out the Amigas. When compared to the Atari/Mac/IBM options, you may be surprised at what they have to offer. ■

Peavey RMC4512

Remote MIDI Controller



One of the first in a new line of MIDI products from guitar and amp stalwart Peavey Electronics is an impressive MIDI program change footpedal. *Review by Rick Davies.*

THE RMC4512 IS a foot-operated MIDI controller intended primarily for musicians whose hands are too busy playing to hassle with changing programs on multiple synths and effects. Of course, this is certainly not an altogether new concept; Yamaha, Roland, and Meico all have similar devices, though each boasts a benefit or two not offered by the others. Naturally, the RMC4512 does this also.

The primary function of the RMC is to transmit individual Program Change commands over four separate MIDI lines. These program changes can be initiated by selecting one of 512 programs (arranged in 64 banks of eight) – either by pressing one of the RMC's four dedicated footswitches, or by sending a Program Change command to the RMC's MIDI input. In either case, the RMC must be programmed to suit the MIDI system in which it is to operate.

The 4512 is housed in a sturdy metal chassis with a sloping top panel on which are located eight large protruding rubber buttons. They are placed in such a way that any of them can be pressed with a shoe, sandal, or python boot without accidentally activating any surrounding switch. The switches will receive no beauty prize, but they certainly do require more than just a casual tap to activate, and the last thing any musician needs to have happen in performance is to fall prey to the "oops" effect – especially when there are four or more MIDI devices making it clear that you *missed*.

The RMC's back panel provides four individual MIDI outputs (A-D), a single MIDI input, a 1/4" momentary footswitch input, a recessed Program Enable slider switch, and the input for the detachable wall-mount power transformer, which provides the required 9V supply.

Hook-up is simple; if you have four or less MIDI instruments to control, each one can connect directly to one of the RMC's MIDI outputs. If more than four instruments need controlling, you'll need to resort to Thru-chaining. If you have another instrument that you prefer to use as the master program selector (say, a keyboard with limited MIDI capabilities), you can use its MIDI output to drive the RMC's MIDI input.

As with most programmable devices, there are two basic modes of operation: Control mode and Program Entry mode. When the Program Enable switch is set to Control Mode, the RMC is ready to transmit MIDI commands to your instruments. Two of the footswitches – Increment and Decrement – allow you to scroll up and down through the 64 banks of programs without actually changing the current ("active") preset. Four of the footswitches (labeled 1/5, 2/6, 3/7, and 4/8) are what you step on to recall programs, in conjunction with a "Shift" footswitch that toggles between the lower and upper sets of four program numbers.

When a program is selected, the RMC can transmit a corresponding Program Change message (not necessarily the same number) over any one or all four MIDI outputs, on any of the standard 16 MIDI channels. If you need to transmit multiple program changes simultaneously, pressing the Chain footswitch causes all eight programs in the current bank to be transmitted at once. Because guitarists could find the RMC helpful in performance, but might find that having four long MIDI cables running from the RMC cramps their space, Peavey has included an external footswitch input so that the RMC can sit off stage and have chains selected from standard momentary footswitches.

Program mode is entered by switching Program Enable to the Program Entry position. In the upper left side of the top panel are a set of four Program Set switches: "+," "–," "Param," and "Store," which are used for editing the RMC's programs. To the right of these switches is a row of five LED display windows that show the current parameter settings for the active preset. The Param switch selects one of the five parameters (Bank, Active Preset, MIDI Output, Program Preset, and MIDI Channel) for editing. When a parameter is selected for editing, the value in the display blinks and the "+" and "–" switches are used to adjust the value.

What sets the RMC apart from the crowd is its ability to transmit up to any 14 bytes of MIDI data along with the Program Change messages. To enter this "Hexadecimal Program Entry mode," the Param switch must be held down while toggling the Program Enable switch. The Bank, Active Preset, and MIDI output are adjusted as usual, but instead of setting a program number in the Program Preset window, the "+" and "–" switches step through hexadecimal values which correspond to MIDI messages (eg. FAH=MIDI Start). When you adjust the MIDI Channel, you're actually selecting one of 14 possible bytes. This feature makes it possible to turn on a drum machine connected to one MIDI output, select a couple of programs on other outputs, and perhaps center the pitch wheel on yet another. If you've ever wanted to try your hand at MIDI hacking, the RMC is a simple way to get started.

Overall, the RMC works great. The ability to transmit single or multiple MIDI commands with one stomp is a nice option, and the Hex programming mode is a great plus.

My one reservation abut the RMC4512 is that it doesn't echo incoming MIDI notes to the outputs, which means that any synths that the RMC drives cannot be played over MIDI, unless a MIDI merger is incorporated into the system.

Being directed at musicians with MIDI-equipped effects, drum machines, or mixers (such as Peavey's own PKM8128), the RMC fills the need for central control admirably and affordably as well. ■

PRICE $299
MORE FROM Peavey Electronics Corporation, 711 A Street, Meridian 39301. Tel: (601) 483-5365

SDX

16 voice—16 bit sampling percussion system

Up to 88 seconds of sampling at 44.1 KHz

Zone Intelligent drum and cymbal pads—
up to 9 samples assignable per pad—
selected by dynamic and strike position

Real-time recorder option with on-screen
editing and SMPTE

SCSI

MIDI

User friendly, on-screen programming

Unrivalled sensitivity and player control

The ultimate instrument for drummers, and
people who record drums.

Simmons—redefining the standard for
professional electronic percussion.

SIMMONS

Simmons USA, 23917, Craftsman Road, Calabasas, Ca. 91302. Tel: 818 884 2653.
Simmons Electronics Limited, Alban Park, Hatfield Road St. Albans, Herts AL4 0JH. Tel: (0727) 36191 (5 lines)
Simmons Electronics GmbH, Emil-Hoffman-Straße 7a, 500 Köln 50. Tel: 022 366 76 66.
Simmons Japan, Beluna Heights 1B-1, 4-11, Hiroo 5-Chome, Shibuya-ku, Tokyo, Japan 150. Tel: 03 440 0991.

Unisynth XGI and XGIM

Electronic Guitars

Photography *Philippe Eine*

If there's one thing that can get guitarists to warm up to guitar synths, it's a low price. Suzuki's first two electronic guitars are definitely the least expensive instruments in this category, but retain that essential element of *fun*. *Review by Rick Davies.*

JUST REMEMBER, THESE things cost under $300. OK, that's all the justification I'll offer for reviewing these two Unisynth instruments – for the time being at least. Designed and distributed by Suzuki in the USA, the XGIs are obviously aimed at the general consumer market, but they can work in more "upscale" applications as well.

The Unisynth XGI is an all-in-one electronic instrument with guitar-style controls. I say guitar-style because aside from there being strings that you pick or strum, and frets that you bear down on, the XGI behaves like virtually any inexpensive portable keyboard. It can run off of six size "C" batteries or an external 9V power supply and has a built-in speaker. The XGI also has a 1/4" audio output, six preset voices (Guitar I and II, Synth, Vibes I and II, and Brass), a built-in rhythm generator (with preset classic rhythms like Swing, Waltz, Disco and 16 Beat), and three extra bass sounds for completing the rhythm section. Should you choose to use the Chord function, the XGI turns into an automatic music-making picnic accessory – some fun.

As the NAMM report indicated (MT August '87), the XGI's feel is not bad when you consider how it works. Like the Stepp and SynthAxe MIDI controllers, the Unisynths offer one set of six strings for picking. These strings are not tuned, but are merely triggers that fire when a string is picked. They do sense how hard you pick, and do a reasonable job of following the nuances of your playing.

Unlike the Stepp and SynthAxe, however, both of which provide a second set of strings for fingering chords or leads, the XGI's fretboard features raised rubber ridges running up across 12 rubber frets. These ridges give the appearance of strings, and to a certain degree even the *feel* of strings, but when your fingers bear down on them, you're only activating switches that tell the XGI which notes you're playing on each "string." The obvious drawbacks of this system are: a) you can't perform pull-offs or hammer-ons because the XGI only produces a new note when a string is plucked; b) you can't bend the strings because they're not strings; c) it's not obvious whether or not you're pressing down hard enough on the fretboard, because the rubber "strings" don't really move so much as they give in to pressure.

Now, before this starts sounding a bit too critical an assessment of a novelty item, let me say that I had a blast with the XGI. It really is a *lot* of fun.

The instrument also has separate volume controls for the rhythm and bass sounds, as well as a master volume control. A tempo knob completes the variable control section of the XGI. The whammy bar, which can raise or lower the pitch of all played notes, does so in semitone increments – so don't get your hopes up if you're into quarter-tones. Both the XGI and the XGIM also feature a mute switch at the base of the bridge which allows you to simultaneously shut off the signal on all six strings.

"How well does it track?" Someone has to ask. It tracks fine – within limits. If you pick hard enough, the XGI responds instantly. After all, since you're only pressing switches and plucking sensors, no pitch conversion is necessary. However, the XGI does not respond like a real guitar would all the time, not only because of the drawbacks of the rubber fretboard but also because of the sensitivity of the "picking" strings. If your string-picking technique ever involves using an adjacent string to bring your finger or thumb to rest, you'll find that both strings will be triggered. It's conceivable that you could develop a technique to take advantage of this effect, but initially it can put you off.

So why the initial emphasis on the XGI rather than on the MIDI-equipped XGIM? Basically, all of the playing aspects of the XGI also apply to the XGIM. The main difference between these instruments is that the XGIM has a MIDI Out jack and features no sounds of its own, favoring instead to trigger external synths over MIDI. The only controls on the "M" are four knobs that control MIDI volume, pick sensitivity, tremolo range, and note duration; ten numeric membrane switches; and an "Enter" switch. A two-digit LED display shows the current program number and reflects the number sent with a Program Change MIDI command.

Ah, yes, MIDI. While we're on the subject of MIDI messages, it should be pointed out that the XGIM transmits only on channel 1. Using the whammy bar, pitch-bend messages are transmitted, and the "Tremolo Range" knob adjusts the range of pitch-bend values. Because there are no real strings, there are no natural note decays; so you need to make use of the "Sustain" control to set the length of the note messages sent out.

I ran the XGI into a couple of MIDI synths, including Suzuki's own SX500 synth module, and had yet another blast. Sure, I often cursed the way the XGIM forced me to adjust my playing technique to suit the instrument, but on the whole the XGIM struck me as one quick and dirty way to get into MIDI guitar controllers. And while I wouldn't bother comparing the XGIM to any "standard" guitar controller, due to its very limited MIDI implementation, I think it could well get a lot more people into guitar synths. Just remember, these things cost less than $300 . . . ■

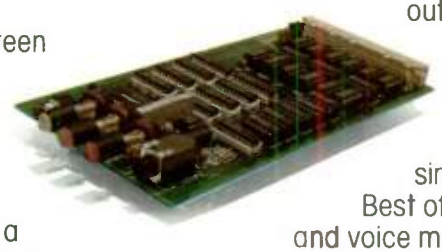
PRICES XGI or XGIM, $299
MORE FROM Suzuki Corporation USA, PO Box 261030, CA 92126. Tel: (619) 566-9710

YOU'VE NEVER HEARD DRUMS LIKE THIS.

THE F16 DRUM COMPUTER.

F16 Now, through the power of 16 bit digital sampling technology, there is a drum computer capable of reproducing any sound with the clarity and brilliance of a compact disk: sounds so amazingly precise it's shocking.

The F16 provides from one to sixteen modular voices that will alter your percussive perspective forever. Full 16 bit samples with 96 dB of dynamic range. Imagine a crash cymbal sampled at 50 KHz that plays for 6 startling seconds; or a tom that booms for 22 seconds or more.



Each voice has independent tuning, volume and pan controls, and is available through direct outs or the stereo mixer. Mic and line inputs make your sampling easy, and editing commands allow you to trim and move samples at will. Save your sounds on the super fast high capacity floppy disk drive.

Driving the F16 from pads, pre-recorded drum tracks or your drum machine's trigger outs, the response time is a blinding 100 microseconds! It's dynamic sensitive, of course, with MIDI in, out and through.

The F16 is rugged, reliable, and simple to use. And, it sounds incredible. Best of all, prices start at under $2,500 and voice modules are easy to add.

Prepare yourself. Forat Electronics is about to propel you and your music into dimensions of sound that previously existed only in the farthest corners of your imagination.

FORAT ELECTRONICS

11514 Ventura Boulevard—Unit 1, Studio City, CA 91604 (818) 763-3007

REVIEW

Roland VP70
Voice Processor

The new multi-faceted signal processing box from Roland combines pitch-shifting and vocoder-like effects with pitch-to-MIDI conversion. *Review by Bob O'Donnell.*

WHAT'S IN A name, anyway? I mean if you call a piece of equipment by a particular name, what effect does it really have on your understanding of the machine's capabilities?

If the gear in question fits into a simple generic category, then probably not much, but if it's a bit unusual or esoteric, then a good name can really go a long way towards creating a strong impression of what the equipment can – and cannot – do. A bad name, on the other hand, can create inaccurate impressions which quite often lead to confusion and frustration.

Unfortunately, Roland's new VP70 Voice Processor falls into the latter category. Besides being more than a little vague, the name is a bad one, partially because it seems to be incorrect. That is, the unit can certainly work with the input of a human voice and it does perform signal processing functions, but it also works with other sources and in fact, performs some of its functions *better* with these other sources. Admittedly, this is not a major point - the unit either sinks or swims based on its intended functions, not what the name leads you to believe these functions should be – but it does cause one to wonder what Roland's actual intentions for this unit are (or were, when they came up with the name). But enough nitpicking; what does the thing do?

Overview

THE SINGLE SPACE rack-mountable VP70 is really two units in one: it combines four independent, MIDI-controllable pitch-shifters (Voice Expansion Mode) with a monophonic pitch-to-MIDI converter (Voice-to-MIDI Mode). The two functions can be used independently or together, and 128 different combinations can be stored as patches (recallable via the front panel, the optional FC100 foot controller or over MIDI) in the internal memory. This memory can be conveniently transferred, stored and reloaded via MIDI System Exclusive commands.

The pitch-shifters can alter the pitch of the input signal in semitone steps up to one octave in either direction, and a detune function can be used for finer adjustments. Activating all four at once allows you to create up to five-note chords from a monophonic input – by combining the direct signal with the effected signals – and three output modes determine whether these sounds will be sent from the VP70's two outputs separately or combined.

Within the Voice Expansion Mode, two further options are available, each of which can be controlled internally or externally via MIDI Note On messages. The Harmonize function tracks the pitch of the input signal and adds "harmonies" to the signal based on either the intervals programmed into the patch (Internal control) or the intervals played on a connected MIDI controller (External control). In the first case, the VP70's output would be parallel harmonies – or a chorus effect, if only the Detune capabilities were used – which follow the pitch of the input signal. In the second case, the harmonies would change according to the intervals played on the MIDI controller – detuning is still possible, but it must be

preset within the VP70. So, for example, if you continued to sing the same pitch but changed the intervals you played on the MIDI controller, the harmonies output by the VP70 would change accordingly.

Unlike the Harmonize function, the Pitch Follow function does *not* track the pitch of the input signal. Instead it outputs a note or group of notes – which again, can either be preset internally or controlled externally – every time it receives an input signal, regardless of its pitch. In other words, the VP70 can function like a vocoder (which I'll explain a bit later). Internal control for this scenario is not particularly useful (except perhaps for special effects) because every time the VP70 receives an input signal it will output the same note or notes. External control, however, lets you create whatever harmonies you want – depending on the notes you play on the MIDI keyboard or whatever other controller you use.

Now if you're wondering – as I originally did – what the difference is between Harmonize with External control and Pitch Follow with External control, let me explain. In the Harmonize mode, the external MIDI note commands determine the *intervals* which the pitch-shifters produce, not necessarily the notes. Middle C (C4) is considered the default setting, so if you play a D4 on the MIDI controller, then the input signal will be raised a full step. For example, if you sing the note C at the same time that you play the note D4 on the controller, the pitch-shifted (effect) output will be D4; but if you sing an E4, the output will be F#4 – a full step higher. In Pitch Follow mode, however, both of these examples would produce a D4 in the effect output because the pitch-shifters "follow" or track the actual notes being played on the MIDI controller, not the intervals.

By the way, an additional benefit of Pitch Follow mode is that if you don't have the greatest singing skills (ie. you're always flat or perhaps in a different key signature), you can still produce reasonable vocal harmonies by playing the parts on the connected MIDI controller and just singing the same out-of-tune pitch over and over. Make sure you don't use any of the direct signal, though, or things could get ugly – fast.

The Voice-to-MIDI function does exactly what you would expect (but unfortunately not very well in some circumstances); it converts the pitch of a voice or any other monophonic source, such as a wind instrument, into MIDI Note On messages. The VP70 can only output one MIDI note per input signal, but it can also send out MIDI Volume, Pitch-Bend and Aftertouch messages.

Operation

AFTER YOU'VE MADE the appropriate MIDI connections, the first thing you need to do with the VP70 is plug in a source. The unit is well equipped to handle a number of different types and levels; it features an unbalanced 1/4" jack on the front panel, a balanced XLR connector on the rear, a three-position attenuator switch and an adjustable input level control for fine tuning. The attenuator can be switched from −50db to −10db to +4db, so that anything from mic level to professional line level signals can be directly connected, and the handy six-segment level display insures that they're properly set.

If you plan on using an instrument like a guitar or a wind instrument as the input source, you'll also need to tune it to the VP70. (The reason for this is that the tuning of the effected signal is based upon the internal settings of the VP70, so the direct signal needs to coincide with it.) To access the unit's built-in digital tuner, all you have to do is hit the dedicated Tune button and play any note. The display will respond by telling you the name of the pitch, its octave and whether or not it's in tune with the VP70. You can also adjust the tuning of the internal reference pitch from A=430.0 to A=450.0.

Like other Roland rack-mount MIDI products, the VP70 has a front panel MIDI message indicator, which quickly and easily lets you know whether or not the unit is receiving MIDI messages. If not, changing the unit's reception channel is only one button away. (Transmit channels – for pitch-to-MIDI conversion – can be stored per patch.) In addition to altering MIDI channels, the System button lets you change

REVIEW

► the VP70's default Mode setting from Poly Mode to Mono Mode. Wait a minute, did you say, Mono Mode? That's right, this guy works in Mode 4; specifically, it seems, for use with guitar-to-MIDI converters like the GM70. Each pitch-shifter can be assigned to its own MIDI channel so that note and pitch-bend messages which are sent per string will individually affect each pitch-shifter. The channels must be sequential, starting with the basic channel; but if you want, you can also set an independent global channel – the basic channel minus one – for program change and hold messages.

Three different output modes are available on the VP70, as mentioned above, and these determine how the direct and effect sounds come out of the machine's two available outputs. Mode I sends the direct signal out of Output A and the effect signal out of Output B. Mode II combines the two and sends them both out of A and B – which also happens if only one of the outputs is used – and Mode III sends the effect signals from pitch-shifters one and three to Output A, the signals from two and four to Output B and the direct signal out both A and B. Mode II is particularly useful for chorus effects. Unfortunately, there's no output level control to adjust the signal from the VP70, but seeing as it puts out a common –10 line level, you should be able to work around this minor limitation.

Once everything's tuned up, hooked up and programmed, you can sing or play to your heart's content. So the question now is, what does it sound like?

Sound Quality *"The chorus programs are very lush and sparkling, and the harmonizing programs are clear and virtually free of glitching."*

The Effects

AT FIRST LISTEN, the VP70's pitch-shifting abilities are very impressive. The chorus programs are very lush and sparkling – the effect signal has a 15kHz frequency response – and the harmonizing programs are incredibly clear and virtually free of glitching. I did notice a very slight delay – not unlike a tight slapback echo – between the direct and effect signal when I set one of the pitch-shifters to unison, but it didn't cause a major problem. In fact, if anything, it added to the quality of the sound by giving it a bit more body.

I tried a number of different inputs with the unit, including vocals, guitar, trombone and synth and, for the most part, the VP70 worked well with all of them. I did run into a problem with aliasing noise in the top few octaves on some bright synth patches, but this is characteristic of many products that utilize sampling (the VP70 "samples" the input signal so that it can digitally process it), and not necessarily something peculiar to this machine.

One of the more impressive effects I was able to produce with the unit was an absolutely monstrous guitar sound that I created in the Harmonize mode with internal control. One pitch-shifter was set an octave down, another at unison, the third a fifth up and the last one an octave up. Talk about a huge, clear sound – wow! Single note lines worked best with this patch, but I was able to play a few simple chords as well and the VP70's output still remained clean. Nice.

Switching over to external control in the Harmonize mode, I connected the output of a sequencer to the VP70's MIDI In – remember, *any* MIDI Note On messages will affect the intervals of the various pitch-shifters, regardless of the source – and tried singing along with some sequenced chords. Though it takes a bit of planning to work well, this method can produce some extremely nice, multiple-part harmonies with changing intervals. Of course, the same effect can be achieved by playing the proper intervals live, but it may take some quick transposition on your part to figure out exactly what notes make up those intervals. (Besides, for those of us who have a difficult

time walking and chewing gum at the same time, the sequencer lets you concentrate on just singing the right pitches.)

Other functions you have available when using external control (in both the Harmonize and Pitch Follow modes) are bend range, for determining how the pitch-shifted signals will respond to pitch-bend data – a very nice feature – and upper and lower key limits, for determining which range of notes will be accepted as valid input by the VP70.

The Pitch Follow Mode produces vocoder-like effects by analyzing the character of the input signal – usually your voice for these applications – and then reproducing that sound at whatever pitches are preset within the machine or whatever notes you play on the MIDI controller. If you sing or even just say "Oooh," and play a Dmaj7 chord on the controller, the VP70 will analyze the sound, look at the notes being input via MIDI and produce a Dmaj7 in "Oohs." According to the manual, the effect will only work properly if the input signal is within an octave of the pitches being played on the controller. In actual practice the range seems to be a bit wider, but one effect I noticed was that the output signal gets brighter as the input source signal goes down in pitch.

One other point to be aware of is that if you do use external control for either Harmonize or Pitch Follow mode the VP70 gives you the option of turning a function called Key Hold on or off. When Key Hold is on, the pitch-shifters will retain the same interval or note settings as the last note messages they received and will process incoming signals at those settings until they receive new note data. If Key Hold is off, however, the pitch-shifters will only operate when they receive MIDI Note On messages; no output will be produced if they aren't receiving incoming messages.

Pitch-to-MIDI

I HAVE TO admit that I was really looking forward to testing this portion of the VP70; the idea of being able to trigger synths from my trombone or sing lines into my sequencer was an extremely enticing one. Unfortunately, my expectations were frustrated. It's not that the VP70's pitch-to-MIDI conversion doesn't work at all, but I did have difficulty getting the machine to work cleanly in a number of different situations.

Before I describe the tests, however, let me explain the unit's capabilities. First, the VP70 will convert any monophonic source into MIDI note information with velocity. If you scoop or slide into a note while singing or playing, it can convert that action into pitch-bend information over a range of up to two octaves. You can also turn the bend function off for chromatic operation. Increases in volume level after the initial attack (ie. crescendos) can be converted

► into MIDI Volume or Aftertouch messages or both at once. (A point to be aware of that isn't mentioned in the manual is that during times when the VP70 is not receiving an input signal, it sends out MIDI volume messages at around a level of zero. The reason for this is to help reduce the possibility of false triggering, but a side effect which can occur if the VP70's MIDI Out is connected to a keyboard is the keyboard may not respond when you attempt to play its keys because the VP70 is turning its volume level down to zero.)

REVIEW

Pitch-to-MIDI *"I was really looking forward to testing the conversion portion of the VP70, but unfortunately, my expectations were frustrated."*

Each patch which uses the Pitch-to-MIDI mode can hold a separate transmit channel, program change number – to be sent out to the connected sound source – and transposition amount (+/– two octaves).

As nice as these capabilities are, however, the real question is how well it tracks the incoming source signal. When I did the obvious and tried singing into the machine, the VP70 seemed to follow some of the notes I sang, but not all of them. Blaming the problem on the unsteadiness of my pitch, I quickly grabbed my guitar and plugged it in. Sure enough, the single note lines on the guitar tracked better than my voice did, but even still, there were occasional glitches. Finally, I tried my trusty trombone – dusted off especially for the occasion – but again, had only limited success.

At this point I began to wonder if there was operator error involved because I had seen two impressive demos of the VP70 in operation, but I just couldn't get it to work well. A few quick phone calls to Roland, however, led me to the conclusion that I was indeed using the unit properly.

The basic problem, it seems, is that the pitch-to-MIDI portion of the VP70 takes a while to get used to; it does not provide immediate gratification via simple and effective pitch-to-MIDI conversion. Instead, the unit requires that users take note of some special considerations. Wind instruments, for example, really need special contact microphones to track properly and vocalists should refrain from scat-singing quick, legato lines. I personally think that these considerations and others like them are limitations of the VP70. Even though I did not have the opportunity to try it with a contact microphone, a machine which is touted to have the capability to accurately convert any monophonic source into MIDI should be able to reasonably do so in any normal playing situation.

Conclusions

WORKING WITH THE VP70 was an alternately fun and frustrating experience. The pitch-shifting capabilities are very clean and, for the most part, quite effective. In fact, when I A/B'd the VP70 with a similar unit, the VP70 was the hands-down winner for audio quality. But the pitch-to-MIDI part of the machine was a constant frustration. Admittedly, I had high expectations for what it could do – perhaps, too high – but I was really put off by the tracking and glitching problems I encountered.

Of course, the ultimate decision on the value and worth of a piece of gear depends on what you can do with it musically. With that point in mind, the harmonizing section of the VP70 was awarded high scores. Though it is a tad expensive, it is capable of producing some very nice effects which could be used in a variety of recording and performance environments.

If only they'd given it a better name. ■

PRICE $1495
MORE FROM RolandCorp US, 7200 Dominion Circle, Los Angeles, CA 90040. Tel: (213) 685-5141

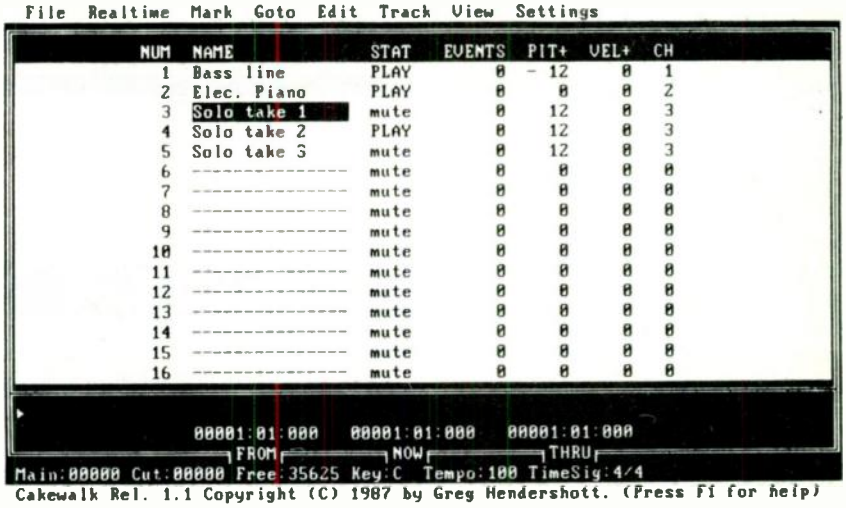
Twelve Tone Systems
Cakewalk Sequencer

This inexpensive new IBM-based sequencer offers a whopping 256 tracks and impressive editing capabilities. *Review by Matt Isaacson.*

REVIEW

FOR EVERY PROBLEM technology solves, it creates another. The main problem which Cakewalk poses for me is how to do it justice in the limited space allotted for a review. I'll begin the whirlwind tour through this remarkable program with a quick overview of its features.

A product of Twelve Tone Systems and the brainchild of one Greg Hendershott, Cakewalk is a MIDI sequencer program for the IBM PC and compatibles. It uses a Roland MPU401 or other compatible interface, and in the current release (Version 1.1) also requires more than 256K of RAM in order to run. Although it supports color monitors and Microsoft mouse, it remains quite usable in the absence of both. The sequencer provides 256 tracks, each of which can hold any type of data. It functions in a linear, single-sequence format – it doesn't provide sequence looping or simultaneous playback of multiple sequences, although the editing functions make it possible to simulate these things without much difficulty. All modes of MIDI sync are fully supported, including song-position pointer generation as master and song-pointer chase as slave (and there's FSK sync too). Recording resolution is 120ppqn, and step-time recording capability is provided. Documentation, both on-line and in the manual, is among the best I've encountered *anywhere*, including non-music applications – at once, it handily serves both novice and advanced computer users. Best of all, Cakewalk comes in at a price which almost earns it a rating of "cheap."



Getting Around

FUNCTIONS ARE ACCESSED via pull-down menus whose headings are always present across the top of the screen. If you aren't using a mouse, these are pulled down by holding the Alt key and hitting the key for the first letter of the desired menu heading – you then use the Up and Down keys to select a function from the menu. Cakewalk also takes advantage of some very logical shortcuts in operating from the keyboard, so if you're nimble-fingered enough, you can probably get around just about as fast this way as you could with a mouse.

Hitting the F1 key at any time invokes the on-line help system, which takes note of what you're stuck on and automatically provides relevant guidance. The available information is extensive, detailed, and above all clear. Between the help system and the very logical nature of the user interface in general, I was able to get pretty far into Cakewalk without even opening the manual, although, as the help system itself points out, the manual does provide a great deal of information not available on-line, and is well worth consulting. As it also points out, if you *insist* upon not reading the manual, it's probably because you've illegally copied the program and don't have one – and that's *not fair*, especially considering the price of the program. Twelve Tone Systems is taking a bit of a gamble here by not bothering with copy protection, but they are obviously more interested in making the going smooth for legitimate customers than in going to great lengths to protect themselves from a devious few.

Often-used functions such as Copy, Cut and Paste, selection of a given sequence, View format, the already mentioned Help facility and the obvious Play and Record, the Set/GoTo Marker and GoTo Time functions, are always accessible via the Function keys F1-F10. In addition, three programmable tempos can be instantly called up via the number keys 1-3, so it isn't necessary to go into a menu for everything. No matter what function you wander into, the Escape key can get you safely out before any unwanted damage is done. Finally, the keys used for moving the cursor around the screen are logical and generally consistent in their use – you don't need to remember a different set of keys for each mode and function.

Looking Around

AS WITH THE pull-down menu selections always present at the top of the screen, certain items are always present at the bottom. One line graphically represents the current sequence in its entirety, and the current position within the sequence is indicated by an arrowhead which marches across the line as the sequence plays – with a mouse, cueing can be done by clicking on this arrowhead and dragging it left or right. One line down are markers which show From, Now and Thru times in bar:beat:clock format. Like the arrowhead on the position line, the Now time is ►

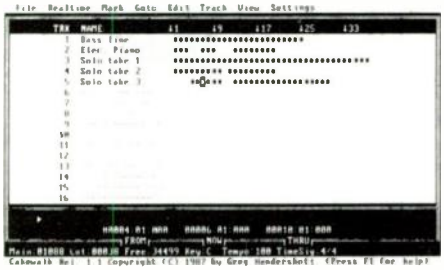
constantly updated to reflect the current position in the sequence, both during record/play and in response to scrolling through the sequence. The From and Thru markers are used to delimit the action of all editing functions, as well as define two points in the sequence which can easily be jumped to via single-keystroke commands. Other single-key functions allow the Now time to be copied to the From or Thru marker – this can be done while the sequence is playing. The Region Memory function lets you store and recall up to ten From-Thru marker pairs to aid in editing complex pieces.

Next down is the status line. It shows event counts for the Main buffer (ie. the sequence itself), the Cut buffer, and free memory, along with the current key signature (discussed later), current tempo (increments of 1 BPM), current time signature, a flashing "P" or "R" if playing or recording, and finally the name of the sequence currently loaded. The Cut buffer is used for Copy/Cut/Paste operations, as well as with the Extract and Merge functions, which allow transfer of info between different sequences. This last capability is accomplished via disk files (because Cakewalk accommodates only one sequence at a time in memory) and is therefore a bit cumbersome and slow, in spite of which it's as precise as any other type of editing (which is to say *very*).

The Sequence View Window

THIS TAKES UP most of the screen. Three different formats are available for viewing the sequence, each of which offers different information and capabilities. Track View shows you your sequence in a list-of-tracks format, one track per line, with 16 tracks visible at any one time. Each line allows editing of a set of parameters which affect the track as a whole. In the namefield you can enter any desired text, to help you keep track of your tracks (pun courtesy of the manual). The Play/Mute status field indicates whether the events on the track will be heard during playback.

Although Cakewalk is an Omni-mode recorder, and remembers the original MIDI channel of the events it records, the Forced MIDI Channel field lets you

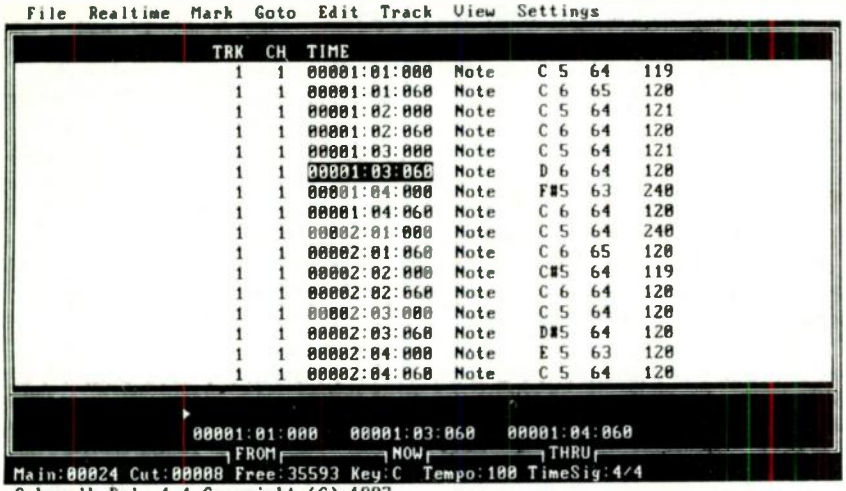


temporarily reroute the events on a track to a different MIDI channel during playback, which can be extremely handy for translating a sequence to a setup different from the one it was created on. In a like manner, the Pitch and Velocity Transpose fields impose temporary offsets on the recorded data. These controls are all the more interesting in that they can be adjusted while playback is in progress, allowing some interesting performance tricks. Finally, each Track View line shows the number of events present on that track.

The Measures View also provides one line per track. Here, the major portion of the View window is given to side-by-side dashed-line displays of the 16 visible tracks over a 30-bar section of the sequence. For any bar in which a given track contains recorded events, the dash is replaced by a small square. This provides an at-a-glance overview of where the action is in the sequence. The horizontal position of the cursor in this View corresponds to the current position within the sequence (aka the Now marker). During playback, it steps along in time with the sequence to show you where you are, while at other times, manual sideways movement of the cursor effects cueing to the indicated point in time. Unfortunately, although this View makes it easier in many cases to set the From and Thru markers to the desired points, there is no visual indication of the position of the From and Thru markers in the Measures View itself.

"Selection" of tracks is done in either

the Track or Measures View. With the cursor on a track line, the Space bar toggles that track's select status on and off. Any combination of tracks may be selected at one time – selected tracks are the ones on which editing functions will operate, and appear highlighted in the View.



The Event View lets you see and manipulate your sequence at the individual event level. In this View, each line presents a single event from the sequence, with events appearing in chronological order. To make this as efficient as possible, only the events on selected tracks are shown. For each event, the display shows the track number, the recorded MIDI channel, the time at which the event occurs, and the type of event (Note, Patch Change, etc). Additional information is presented as necessary to completely characterize the particular type of event – for example, a note event display includes a pitch string (eg. A#2), a note-on velocity, and a note duration in 120 ppqn clocks. All of these event data fields can be edited.

Editing the track number of an event moves it to a new track, while editing the time field moves it to a different point in the sequence – and hitting the Delete key erases it from the sequence. The Space bar, used in the other Views for selecting tracks, here allows you to step through the list of events one at a time, playing each event as you step to it. This takes much of the eye and brain strain out of finding the event you want to work on. In addition, new events can be inserted.

The pitch string notation used by Cakewalk in the Event View and elsewhere is offset by two octaves from what is generally used in the rest of the MIDI world – the bottom key on the standard five-octave keyboard, normally called C1, is here labeled as C3. This was probably done to simplify onscreen notation, because by calling the lowest MIDI note C0, the use of a minus sign for the two lowest octaves is avoided, although it also results in the top octave (the one above "9") being referred to as ":" – a value which in fact cannot be directly entered in pitch string fields. At any rate, although initially confusing, the numbering system is applied consistently throughout. The exact string which is used for a given note is a function of the current key signature (eg. C#2 or Db2). A pretty helpful feature, this is, although only major scales (and their relative minors) are offered as options.

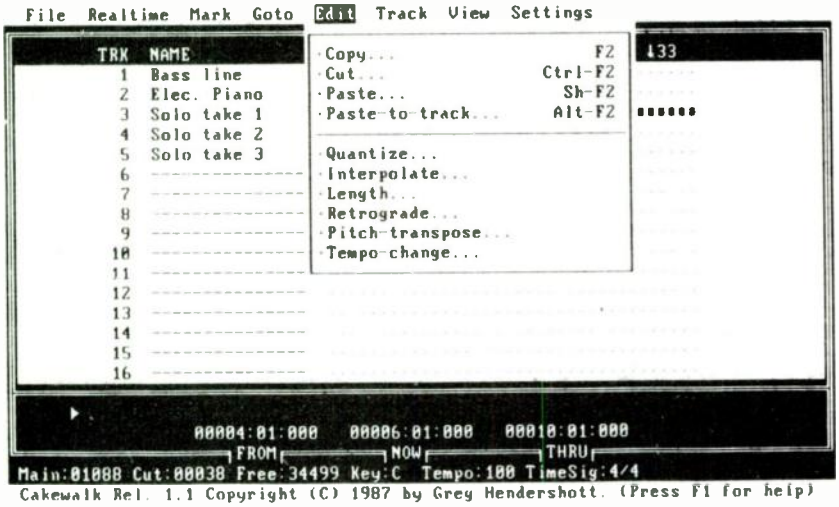
Recording

SELECTION OF A single track prepares you for recording. An optional count-in of ▶

1-4 beats is provided. Recording can be done from the start or from whatever point you have cued to, and is also possible (sans metronome) under external MIDI sync, although not in Song-Pointer Chase mode. Recording filters allow you to specifically exclude any assortment of the event types recognized by Cakewalk, as well as to completely ignore any desired combination of MIDI channels in the incoming data.

With 640K of memory (ie. a fully loaded PC), Cakewalk's recording capacity is roughly 38,000 events of any type. New events are recorded into a separate record buffer which borrows from available free memory. Once recording stops (either manually or upon reaching the prescribed Auto-Stop point), you are given a few choices as to what to do with the recorded data – you can discard it, merge it into the selected track (the usual sequencer mode), overwrite the selected track (more like a tape recorder – existing data in the "recorded-over" section of the track is erased), or if the data contains events on more than one MIDI channel, the new data can be split out by channel and overlaid onto multiple tracks. This last feature is a major convenience for Mode 4 freaks or those who wish to record multi-instrumental live sessions via a MIDI merger.

The Auto-Rewind option causes the sequencer to jump back to a prescribed point (eg. the start of the sequence) whenever the sequencer stops. Used in conjunction with the Auto-Stop option, it also allows you to easily home in on a specific section of the sequence for repeated listening and recording takes.



Editing

AN ENTIRE REVIEW could easily be devoted to an exploration of the editing possibilities afforded by Cakewalk. Here is where this program really shines. The best I can do for now is to give an overview, and to highlight a few of the unique and interesting things of which it is capable.

First of all – all of the usual stuff is here. Arbitrary portions of the sequence can be Copied, Cut, Pasted or saved to disk for merging into other sequences. The timing and duration of recorded events can be quantized to any desired note value. Note pitches can be transposed. An entire track can be quickly duplicated with one simple command – the obvious benefit of having 256 tracks is that you can edit and disfigure a track a dozen different ways and still have the original to go back to. Given a time range and initial and final tempos, the Tempo-fill function automatically inserts the necessary tempo events in the right places to effect a smooth transition between the two tempos over the desired time interval.

Then there's the slightly more off-the-wall stuff. The Length function allows you to shrink or stretch the time duration of a specified section of the sequence by a specified percentage. Optionally, you can also correspondingly shrink or stretch the durations of the individual notes within this section, or even do the latter and not the former. Used in conjunction with track cloning and pitch-transpose, for example, this creates some other-worldly effects. The Retrograde function reverses the time order of the notes in the specified section of the sequence. This has fascinating uses in percussion sequencing, providing a shortcut to some very unconventional accenting and sound/space arrangements, along with the obvious use in turning conventional melodies and chord progressions on their heads.

Even the usual editing functions have a surprising degree of depth. The Cut function allows you to cut with a Jagged edge (note durations are left intact, even though some notes might sustain beyond the Thru marker) or with a Sharp edge (note durations are trimmed as necessary to ensure that none extend past the Thru marker). The hole created in a track by the Cut operation can either be left as a hole (other events on the track are not rearranged) or deleted from the track (events later on in the track are shifted up to fill the gap).

The Paste function lets you copy a multiple-track section of a sequence from place to place with all track separation preserved, while Paste-to-track lets you transfer a section of a track or tracks to a different track. The Pasted info can overlay the destination tracks or wedge into them, pushing existing events back in time, and can be repeated an arbitrary number of times with optional automatic alignment to bar boundaries (which in turn can mean different things at different times, since the time signature can be changed at any time). By activating the Harmonic Math option in the Pitch-transpose function, the transposition occurs as a number of intervals in the current key signature, rather than the usual chromatic transpose (and the current key signature can likewise be changed at any time).

In addition to delimiting edit functions in time (the From and Thru markers) and by track (only the selected tracks), all functions which manipulate existing data offer the option of using the Event Filter, wherein resides much of the real power of editing with Cakewalk. The Event Filter is presented as a somewhat intimidating full-screen table of all event types along with minimum and maximum value fields for each and every parameter associated with each of the event types. In addition, there's a separate section for MIDI channel, beat and clock, since these apply to all event types. Was that clear? Perhaps a few examples will help.

Using the Event Filter, you can: cut out all notes on MIDI channel 3 in the range of C5 through C6 which occur in the latter half of beats 3 and 4 in each measure; pitch-transpose all notes on MIDI channel 12 whose velocity is greater than 75; trim off wild swings of the pitch wheel, or time-quantize your pitch-bends; or, with time signature adjustments, quantize only those notes whose timings fall outside a specified time window surrounding each 16th-note. Neat, huh? Each range parameter in the Event Filter also has a "not" switch which causes the filter to reject, rather than accept, events whose values for that parameter fall within the specified range. However, I was never able to get this to work as described – it seemed to simply defeat all action by the edit function.

But wait, there's more. What do you get when you process the output of the Event Filter through a second Event Filter? You get the Interpolate function, which makes possible a wide range of linear transformations of sequence events. Again, some examples: specifying a note range of C3 through C6 on the first Filter and C6

through C3 on the second will invert the pitches of all note events within that pitch range; a note velocity range of 0-127 in the first window and 30-90 in the second results in note velocity compression. The possibilities are vast, although I was disappointed to learn that event timing cannot be manipulated through the Interpolate function (maybe in the next release . . .).

Problems

YES, THERE WERE a few. What with all of the capabilities of Cakewalk, it came as a great disappointment (although not a complete surprise) to find that it doesn't always deliver all of the goods at once. In particular, having accumulated ten tracks of recorded or edited data, I found that the playback timing on some tracks had gone noticeably askew, even though most of the tracks were muted. Only when the muted tracks were actually erased did things fall back into line. No doubt this is in part because the sequencer must still work through all of this data, even if only to decide that it is on muted tracks, but it seems that it should take more than ten tracks to throw things this far off.

In addition to the "not" switch problem discussed above, the Event Filter came up a couple of times filled with mysterious garbage settings which had to be hand-edited out. Another much less significant problem (reportedly soon to be fixed, at which time the fix will be available free of charge to registered owners) was that the sequencer did not correctly respond to a MIDI start commmand while in song-pointer chase mode. Cakewalk's apparent refusal to allow the playback of layered notes of the same pitch and MIDI channel, even when recorded on separate tracks, is a limitation likely to be most strongly felt in sound-effects applications. Finally, and not really in the bug category, is the total disregard of all System Exclusive messages.

Conclusions

DESPITE THE ABOVE-MENTIONED problems, I'd still have to call this one *hell* of a sequencer. Its editing capabilities are simply too good to ignore, and if using them means ultimately having to banish the unused tracks to disk-file land, well then so be it. Having 256 tracks is a real convenience while editing – it would be naive to expect any program powered by a single 8088 microprocessor to even approach being able to play back this many tracks. The other problems will probably be fixed in time, and meanwhile are far from enough to shut one down. Matt says, "check it out." ■

PRICE $150

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

READERS' TAPES

Reviewed by Yung Dragen.

MUSIC

I HAVE BEEN on a nostalgia trip recently with regards to my listening tastes. "Nostalgia," in this case, includes either early new wave (Urban Verbs, Robin Lane, etc.) or progressive rock (most notably older King Crimson). I marvel how things that seem commonplace today (ie. synthesizers, processed guitars and drums) still seem so fresh and exciting in their earlier "crude" forms. In tandem with this I've done some thinking about the moan that I so often hear, "Music today is so *boring* – when will something new come along?"

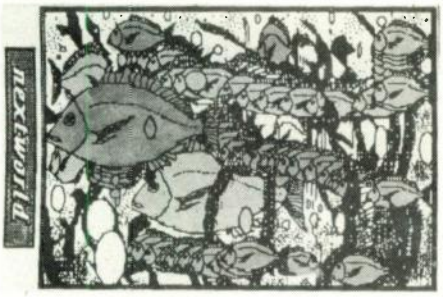
Well, the three tapes reviewed this month are of three styles – industrial dance, new age, and avant electronic – that tend to have roots with experiments a decade or so ago. However, since we have not been beaten to death with them via the radio-waves as of yet, they still seem fresh, new, and "exciting."

Starting with industrial dance, and continuing in the vein of last month's faves Straightjacketbazooka, **dance87** features

Robert Tabb on vocals and saxophone and Damien Gossett on synths, flutes, effects, programming, and vocals. The promotional flyer that came with the tape says, "The artists developed this tape in response to the comment, 'It's nice music but the lyrics are, you know, trite sixties social commentary. You do any dance music?'" So *is* this dance music? No way, unless you're in the throes of some deadly neurotoxin normally reserved for cockroaches and the like in your household. Spastic drum machines weave their way around and through synthesizers in love with static, processed voices, and other mayhem.

The aforementioned Straightjacketbazooka does this style of music well; I have the feeling that **dance87** could use a little more discipline of direction to pull this off (it is better to have fully visualized the path you intend to follow before embarking down it). A few good ideas *are* present in here, though the one relaxing, pretty piece – 'Autumn Leaves' – startled me in how out of character it was with the rest of the music (it also happened to be my favorite). I want to hear these guys' second tape – maybe by then they will have figured out a little better exactly what they intend to do and how to achieve that goal.

Next up is *Nextworld* by **Jaxon Crow**. I live in the electromagnetic shadow of the self-proclaimed first new age radio station in America, KLRS. At first, it was quite pleasant having a station around to mellow out to, and it became the only one I listened

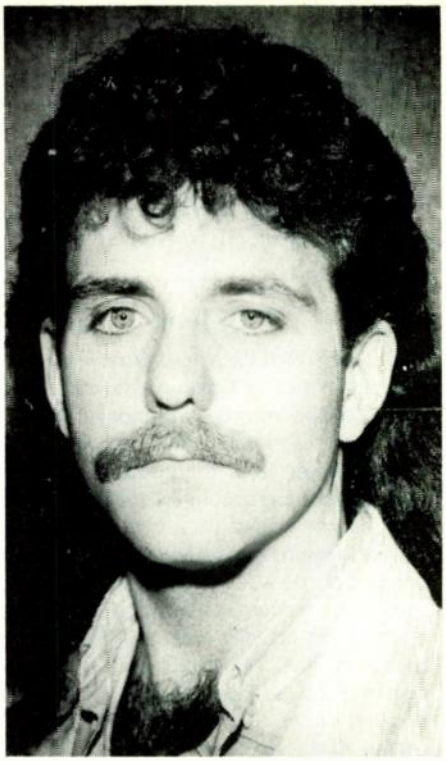


to at home. Lately it has become quite trite, however; the music seems to consist of either overly pompous, grandiose sweeps of instruments or overly simplistic four-note sequences, white noise wind, and "Fairlightish" vocal samples (all elements I like, by the way, but not done in a boring fashion). *Nextworld* is the type of music I wish KLRS would play more often. Sure, it's of the same benevolent, sustained and repeating variety so often associated with the genre, but as opposed to being sugary, it's damn *nice*.

Sequences with dynamics and counterpoint that undulate and surge in some time signature other than 4/4 are present as are drum machines brought up just briefly enough to surprise and move the music along. Languid lines of similar – but different – timbres are interwoven throughout most pieces and enhanced by generous use of reverb. One has the feeling that orchestration went beyond repeating one measure, a drum pattern, a DX7 factory patch, and the player's favorite three sampler disks. There's nothing jarring here, but it's never boring either.

Although no instruments were listed, Jaxon mentions that all was sequenced on a Macintosh with Opcode's Sequencer 2.54, and the very good cover art of fish swimming among plants (hand-colored) was also done on the Macintosh. A very nicely done effort.

Somehow this leads me to our man with two lives, **Paul M. Van Patten**. He works for a "large music company during the day" and plays drums in the, unfortunately, utterly forgettable '70s power-rock band Tour de Force. By night (or whenever), he explains his "intentions are to explore and create new forms of music, utilizing ever-expanding music technology." In this case, the music technology takes the form of an eight-piece Simmons SDS7 kit, an Octapad, an RX5, and a "full complement of MIDI equipment" – most of which is apparently FM in heritage. Imagine if Bill Bruford, with



his current Simmons MTM/Yamaha DX27 setup, took to doing his version of Vangelis' avant garde effort *Beauborg* – these would be the demo tapes.

Shards of triggered FM patches and chords abound, rarely painting a melody or rhythm (in the conventional rock 'n' roll sense – yet there *is* a good deal of harmonic and timing order), but nonetheless, never alienating the listener by being too abstract or strange. It reminds me of "modern" computer music very acceptably done. All the pieces were performed live to cassette, giving further credence to the rumor that with MIDI, drummers may finally be referred to as "musicians" in their own right (an unpopular non-western belief *I*, for one, have always held). If one has an interest in the non-industrial avant garde side of things, or hearing other non-jazz reference points in the new territory Bruford is charting out with the likes of David Torn and Earthworks, bother Paul for a copy of this tape.

To conclude, I really *do* find the vast majority of current mainstream music rather boring. And unfortunately, most people seem interested in using the latest technology to help them recreate more of which there already is too much. The three tapes reviewed here, however, were at least a breath of fresh air (and at most, a pointer at new directions). Speaking of new directions, by the way, what *is* next musically? Do any of you have it out there? ■

Contact addresses:
dance87, c/o Verdant Ventures, 2736 Lyndale Ave. South, Suite 210, Minneapolis, MN 55408
Jaxon Crow, c/o Neon Tetra, PO Box 222311, Dallas, TX 75222 (*Nextworld* is $7.50 ppd)
Paul M. Van Patten, 519 First Street, Liverpool, NY 13088

Take MIDI to the MAX with the MX-8™

Now you can take maximum advantage of MIDI's limitless potential—conveniently and affordably—with the MX-8 MIDI Patchbay/Processor from Digital Music Corp.

Combining versatile 6 x 8 patching capability with an incredible array of processing functions, the MX-8 gives you the power and flexibility of a rack-full of outboard hardware. All in one, easy-to-use device. All for just $395!

No other MIDI peripheral puts this much control at your fingertips:

Patchbay requirements are met quickly and easily with complete routing flexibility for six inputs to any of eight outputs.

Merge MIDI data from any two inputs.

Dual MIDI Delays are independently programmable in milliseconds or as note-value and tempo. You can even specify the number of repeats and their velocity... *and* echo to a different MIDI channel.

Mapping lets you split your master keyboard into four zones to control multiple slaves and set a velocity "cross-switch" point to add or switch sounds when you play harder.

Patch-Chaining enables the MX-8 to send up to eight program changes with each setup.

MIDI Filters can be set on two separate inputs to remove any data from the MIDI stream.

Transpose up or down five octaves on an individual channel or globally.

Channel-Shifting reassigns data from one channel to another... or offsets all channels.

Each function is easily controlled and clearly displayed on the MX-8's front panel. You can even name and store up to 50 setups for immediate recall. Or, use optional software for computer control of the MX-8 with unlimited disk-storage of setups.

It's all the power you'll ever need to take MIDI to the max. The MX-8 MIDI Patchbay/Processor from Digital Music Corp. Available at better music stores everywhere.

The Total MIDI Patchbay/Processor from Digital Music Corp.

DIGITAL MUSIC CORP.

21787 Ventura Blvd., Suite 124 • Woodland Hills, CA 91364 • (818) 704-7879

MIDI 104

The fourth part in our continuing series on MIDI basics dissects a drum machine and discusses its applications in a MIDI studio.

Text by Bob O'Donnell.

AMAZING HOW TIME changes the relationship between words, isn't it? A mere ten years ago the connection simply didn't exist between "groove" and "machine" (some would argue it still doesn't), but today they fit together as well as yuppies and cellular phones. And no matter what people may say, a *lot* of records are still being made with drum machines and they probably will be for a while to come. They've proven themselves useful in countless situations and have effectively become part of our musical culture.

But what is it about these beat boxes that makes them so enticing? Well, the reason varies from person to person, but there is something about playing along with a rhythmic accompaniment which you've been able to create that is uniquely satisfying. Jamming along with a drum machine is just a helluva lot more fun than practicing with a metronome. Plus, every musician feels that they know something about rhythm, and creating your own drum (or percussion) parts seems to fulfill a basic desire to express those instincts. (OK, I admit it, I wanted to play drums as a kid . . .)

What's Inside 'Em?

WHAT EVERY MUSICIAN doesn't know about, however, are the different components of a drum machine; so what follows is a brief overview. Every drum machine consists of three basic parts: the pads, or the controller portion, the internal sounds, and a basic sequencer ("rhythm composer", or whatever label you like) which records the various patterns. Figure 1 should give you a good idea of how the various components of a drum machine fit together. Each one can be used independently of the other for certain applications, but they generally work together to create the drum patterns.

Hitting the pads or drum buttons basically "tells" the computer brain of the instrument – which controls and coordinates all the drum machine's various functions – to either trigger the internal sounds or to trigger and record that particular sound with the internal sequencer – depending on whether or not the drum machine is in record mode. Some machines will also output a MIDI note when you hit a drum button, which you can use to trigger external MIDI sound sources such as a sampler or another drum machine. I'll get into the specifics of this application later.

The sound portion of the drum machine usually consists of digital samples of drum sounds burned into a ROM chip. Don't confuse this with sampling drum machines, however, because many machines do not allow you to sample your own sounds (of course, some do). Regardless of their source, though, these sounds can be triggered either by the pads, the internal sequencer or from a MIDI note message received at the instrument's MIDI In port. Again, I will provide more specifics later, but playing drum sounds via MIDI can basically be done from an external controller, such as a keyboard or MIDI drum pads, or you can control the sounds from another external sequencer of some sort.

Finally, the sequencer portion of the machine records the patterns that you play with the instrument's sounds. Like the dedicated hardware and software sequencers discussed in last month's article, the sequencer found within most drum machines records and allows you to edit "events" that you either play manually in real time or enter step-by-step.

Since drum sounds are generally very short (compared with the range of note durations sequencers are capable of recording), the events recorded by the internal sequencers of drum machines are essentially triggers (usually in response to hitting the pads) rather than incoming MIDI note events, though many newer drum machines will also permit you to record MIDI notes from an external MIDI controller. Some drum machines allow you to *play* their sounds over MIDI, but don't permit you to record patterns this way (eg. the Roland TR505). At present, drum machine sequencers do *not* record pitch-bend, aftertouch and other controller information, though it is possible that they respond to these messages. (For example, the Sequential Drumtraks, used pitch-bend and mod wheel messages to control pitch and volume of the drum sounds.) In any case, the "events" are recorded against a clock source of some sort – usually internal, and made audible via a click or metronome which you record to – and the speed of that clock's output is determined by the tempo control. Most drum machines also permit you to disable the internal clock if you want, and have the tempo of the instrument be controlled from another source. (We'll examine some possible tempo sources further on.)

The actual operation of a drum machine sequencer differs from a stand-alone sequencer in that the patterns you record are generally much shorter than those created on a dedicated device – or program – and the recording method used to create those patterns is easier to understand. (In fact, more and more dedicated sequencers have begun to incorporate drum machine-style program-

ming because most people can easily work with a drum machine but many of those same people have a difficult time understanding and working with a sequencer.) The ease of use stems from the fact that drum machine sequencers continuously loop while you record a pattern and allow you to continuously overdub new parts and erase parts as you go along. Consequently, creating patterns or parts you're happy with is a much faster and simpler process.

As mentioned earlier, some drum machines also permit you to record patterns in the internal sequencer that were played on external controllers. For example, you can play the machine's sounds (and thus record patterns) from a velocity-sensitive keyboard or some other controller. The benefits of this method are many, but high on the list is the ability to record patterns with more realistic dynamics by using MIDI velocity even though the built-in pads may not be touch-sensitive – like on the Yamaha RX5 – and the capability of playing drum patterns from external drum pads. In both instances, the basic concern is to increase the expressiveness of the patterns when you play them back.

To sum the connections up, the sounds can be "controlled," or played, by the pads, the internal sequencer or an external source via MIDI. The sequencer, on the other hand, records the patterns you play on the pads or from an external source via MIDI (on some drum machines).

Hooking Up

UNLESS YOUR TASTES run strictly to pure percussion music, the real fun of a drum machine comes when you hook it up into your MIDI system. There are many possible configurations, and Figure 2 shows one system that's been created over the course of this series. In this scenario, the drum machine acts as the master clock and controls the tempo for the entire system. The MIDI Out from the drum machine is connected to one of the MIDI Ins of the switcher and its output is routed to the input of the sequencer. To permit the drum machine to control the system tempo, it needs to be set to internal clock and the sequencer should be set to external MIDI sync. You do not have to worry about setting Modes or Channels, though, because tempo information (in the form of MIDI Timing Clocks) comes from System messages which are *not* channel-specific.

Now if you want to record into the sequencer while listening to the drum machine, you're going to have a problem with this setup because both the output from the master keyboard and the drum machine need to get into the MIDI In of the sequencer. (The master keyboard provides the note information and the drum machine provides tempo/clock messages.) The only way you can avoid this difficulty is to add a specialized device called a MIDI merger into the system which would combine the two outputs into a single MIDI signal which could then be sent to the sequencer (see Figure 3). An alternative to using an external merger is to use a switcher which incorporates a merging function (eg. the JL Cooper MSB+ and Digital Music Corp.'s MX8).

Rather than complicate the system, however, the other possibility is to have the drum machine "slave," or respond, to the tempo from the sequencer while recording sequences and then switch the MIDI connections after the recording is finished. Figure 4 shows the proper connections for this setup. In this case, the MIDI Out of the sequencer would be routed to the master keyboard and the drum machine via the splitter. Again, Channel and Mode settings do not make a difference for tempo control, but you'll need to switch the sequencer over to internal clock and the drum machine to MIDI sync. (This configuation has other uses, as I'll point out later.)

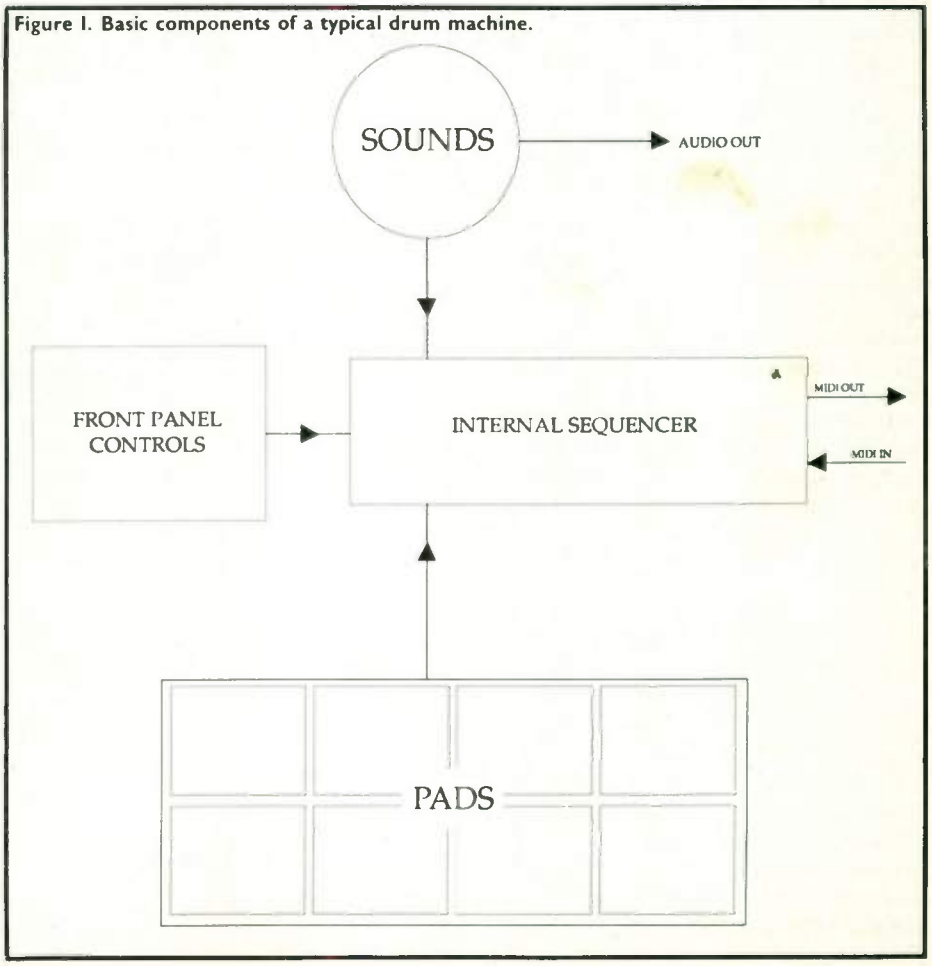
"Every drum machine consists of three basic parts: the pads, or the controller portion, the internal sounds, and a basic sequencer which records the various patterns."

"But," I hear you wonder, "why do you have to connect the drum machine and the sequencer in the first place? Can't they just run independently?" Of course the answer is yes, but the results may not be terribly pleasant. The problem has to do with a subject called sync.

What's Sync?

THE WORD "SYNC" is actually short for synchronization and it has to do with ▶

Figure 1. Basic components of a typical drum machine.



►events occurring at the same time. If you've ever played in a band where the drummer (or somebody else) simply could not keep good time, then you should be able to understand problems that can occur with sync. A sequencer/drum machine system without sync is like a band whose members cannot hear each other. You might be able to *tell* the drum machine and sequencer to play at the tempo and to start at the same time, but without an electronic connection of some sort, there's no way to assure they actually *will*.

The sound-generating portions of a synthesizer or expander do not depend on any tempo control, so by themselves, they do not have any sync problems. On-board sequencers and arpeggiators are another story, however, and similarly, any sequencer controlling the synths in our system relies on a clock source, whether internal or external. Drum machines do as well, so unless the sequencer and drum machine are tied together via a common clock signal of some sort, they will be unable to stay in sync. In other words, even if both instruments are set at a tempo of 100 and you start them together, minor fluctuations are bound to come up which, by the end of, say, a three-minute song, may become unbearable.

The way to avoid sync problems is to have any clock-dependent devices share the same clock source. By doing so, the connected instruments will operate at the same tempo and even if minor tempo variations occur, they will occur consistently throughout the entire system. Problems involved with starting and stopping the various instruments at the same time can also be overcome by connecting them together in this fashion. The instrument being used as the master clock source can send out messages which tell the other system components when to start playing back, when to stop and, if necessary, when to continue playing from a certain point. Again, keep in mind that these messages only have an effect upon instruments which are tempo-dependent.

MIDI Messages

THE MIDI SPECIFICATION contains an entire subset of messages dedicated to sync and other tempo or time-dependent functions: System Real Time commands. Like other system messages, System Real Time messages are not sent on any specific MIDI channel and are thus recognized by all instruments in the system regardless of their mode and channel settings – *if*, of course, they have any use for these messages.

Figure 2. Basic MIDI system.

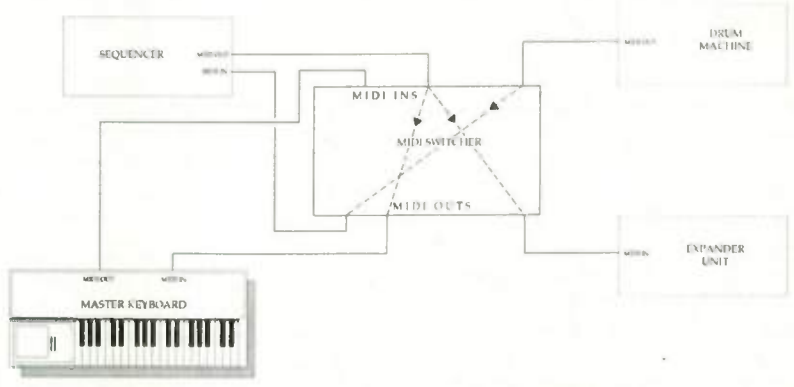


Figure 3. Adding a merger to the basic system.

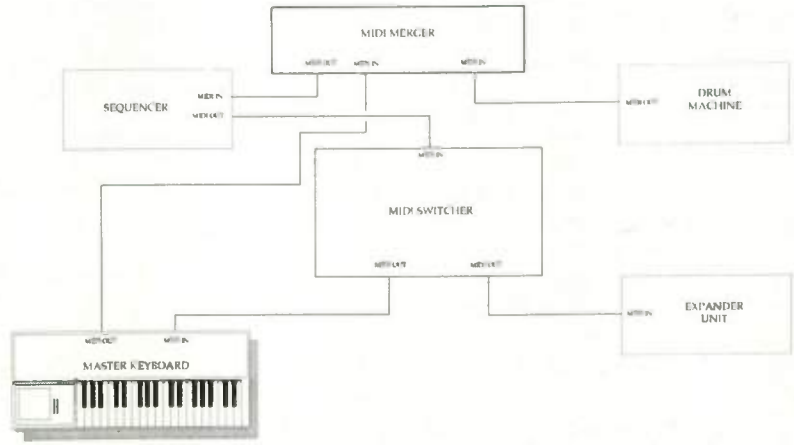
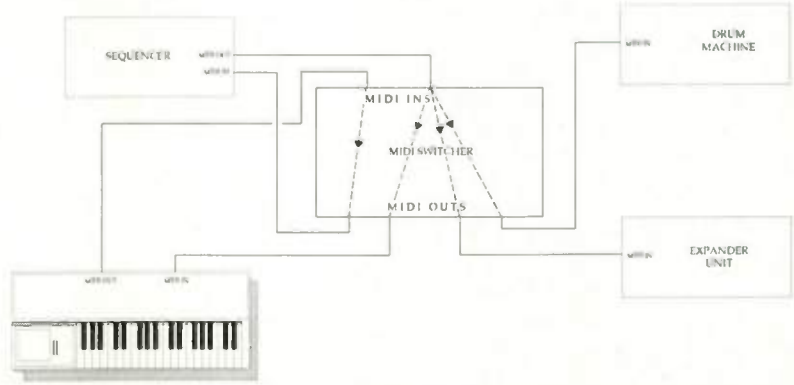


Figure 4. Using a sequencer as the master clock source.



The actual messages which fall under the heading of System Real Time commands include MIDI Timing Clocks, and Start, Stop and Continue messages. As mentioned last month, the MIDI specification calls for 24 clocks or pulses to be sent per quarter note, which is the equivalent of a clock per 64th note triplet – if you're working in a quarter note meter such as 3/4 or 4/4. Some drum machines and sequencers have a higher internal resolution (ie. 96 pulses per quarter note, or ppqn) for recording purposes, but for tempo synchronization with other instruments, 24 ppqn is the MIDI standard.

equally valid. Old Korg and Roland drum machines have DIN Sync ports which use the familiar five-prong DIN socket – *not* MIDI ports – and other machines offer clock outputs of various speeds, including 24, 48 and 96 ppqn. Many sequencers can also generate and respond to these non-MIDI clocks and, if you so choose, you can use them in place of MIDI sync and they will work just as well. Pulses may vary in level and duration from instrument to instrument, however, so care must be taken to avoid incompatibility (another problem which MIDI does a good job of avoiding). You will be limited to starting,

"In addition to System messages, many drum machines can generate and respond to some Channel Voice messages; specifically, note-on data."

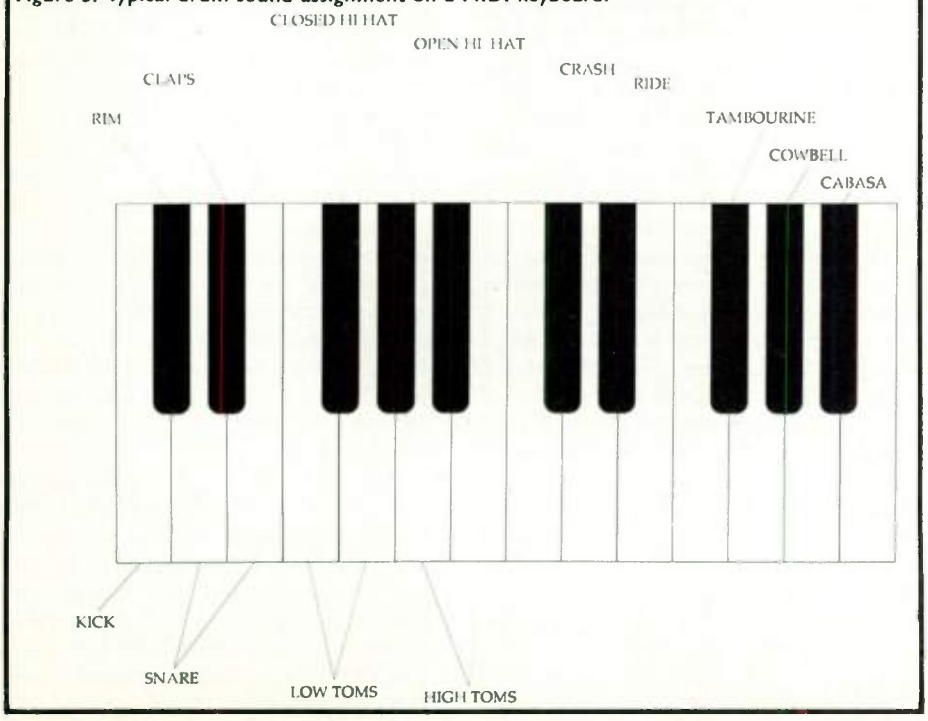
Though they are for the most part, self-explanatory, the Start, Stop and Continue commands do deserve a few words. Start commands causes playback from the beginning of a song or pattern, while Stop commands causes playback to halt. If you want to start playback from the point where you stopped, you need to use a Continue message, since a Start command would return the sequence to its beginning. If you attempt to start a drum machine from its front panel while it is set to sync to an external source, nothing will happen until it receives the appropriate System Real Time messages.

Despite the overwhelming presence of MIDI, many popular drum machines also offer non-MIDI sync methods which are

pausing and basic tempo control, but if you are doing heavy sequencing with a number of sound generators, it does provide a way to clear up the MIDI data stream a bit. (The Sequential Studio 440 uses a separate set of terminals to deal with MIDI sync.)

Should you decide to stick with MIDI sync, though, the possibilities start opening up because of the additional information which can be sent via MIDI. For example, other System messages which many drum machines can send or respond to are Song Select and Song Position Pointer, both of which fall under the banner of System Common commands. Song Select messages tell the receiving device which song to call up from within its internal sequencer memory. Thus, if you could

Figure 5. Typical drum sound assignment on a MIDI keyboard.



select song number 17 on your drum machine, song number 17 would automatically be selected on your sequencer.

Song Position Pointer (SPP) messages are a bit more involved, but as explained last month, they can save you a great deal of time while you are working within your MIDI system. What they basically do is provide information about the location you are starting at within a particular song. A device which sends or responds to these messages keeps track of the number of MIDI beats (one MIDI beat equals six MIDI clocks, which is usually equal to a sixteenth note) that have passed since the beginning of a song and, in the case of the master, sends out a two-byte message which defines this location or, in the case of the slave instrument, immediately moves forward or backward to the specified point within the song. (Like other master-slave situations within MIDI systems, both devices need to be able to deal with these messages; if only one does, the messages serve no useful purpose.) So, for example, if your sequencer is serving as the master, and has a fast forward function and you move to a point in the middle of your song, the drum machine connected to it should automatically move to that same position and begin to play from the same point, at the same time and at the same tempo as the sequencer. If the drum machine can't respond to SPP messages, however, it would simply start from the point where it had last stopped.

Another benefit that becomes available for devices which respond to SPP messages has to do with tape sync. Tape sync refers to the practice of recording a sync track of some sort – one which includes timing or tempo data – on a multitrack recorder and then playing that track back into the device which generated it – typically a drum machine or a sequencer. The sync track thereby controls the starting and stopping of the instrument, as well as its tempo, and permits you to play and record other parts (eg. vocals and guitar) onto tracks of the tape recorder whilst listening to the synchronized MIDI parts.

Most tape sync functions require you to rewind to the beginning of the song to start, though, because if you start in the middle of the tape, parts recorded on tape will start from the point in the middle of the song where the tape was stopped, but the synchronized instruments will start either from the beginning of the song, or from where it was last stopped, rather than from the correct point. As a result, the two sets of musical parts will be – guess what? – out of sync. But if you can somehow tell ▶

► the sequencer or drum machines the point at which you are starting playback, then they can advance the song to the proper point. That is where Song Position Pointer messages come in. These messages "tell" MIDI devices to automatically move to the desired point in the song, regardless of whether it's at the beginning, middle or end of a song. If you're working on something towards the end of a piece, this capability is *invaluable*.

In tape recording applications, one track of the tape is usually "striped" with a time code (typically SMPTE or some similar format), which marks the tape with addresses from which Song Position Pointer messages can be derived. Although many sequencers themselves do not "write" or "read" SMPTE or the like, several "black boxes" are available to do the job. Roland's SMPTE-based SBX80 was among the first, and now the Synhance MTSI and JL Cooper PPSI are providing inexpensive alternatives to SMPTE for less-demanding situations.

"One application for MIDI note messages is to trigger sounds in the drum machine from an external controller or another sequencer."

Triggering Drum Sounds Over MIDI

IN ADDITION TO System messages, many drum machines can generate and respond to some Channel Voice messages; specifically, note-on data. MIDI Note messages, in fact, present some very interesting possibilities for our typical MIDI system.

First, as mentioned previously, many drum machines output specific MIDI notes on a specific transmit channel when you hit the instrument's pads. In other words, some drum machines can work like keyboard controllers. The duration of the note is generally preset to a very short time and, unless the machine has touch-sensitive pads – though, unfortunately, not many do – the velocity level is fixed, so they don't normally make for particularly expressive controllers. But if you're not happy with the sounds in your drum machine and have access to a sampler, then the machine's ability to output MIDI notes will make it worth its weight in gold. Why? Let me explain.

Drum machines which can produce MIDI Note On messages when you hit individual pads will also do so while playing back patterns or songs. So to make use of an additional sound source, like a sampler (which can take the place of or be used in conjunction with the on-board sounds), all you need to do is make sure that the note assignments on the drum machine correspond with the desired drum sounds on the sampler. For example, if the snare drum pad on your drum machine is assigned to transmit MIDI note number 38 on channel 10, then you need to assign a snare drum sound on your sampler to MIDI note number 38 (D1) and set its receive Channel to 10.

After you've done so – and, of course, connected the two instruments via a MIDI cable – every time you hit the snare button and every time the snare drum plays within a pattern, the sampler's snare drum sound will also play. (If you run into a problem with the sound from the sampler only playing for a very brief duration, you may have to engage a function on the sampler which ensures that the sound plays through its entire envelope cycle regardless of the length of the triggering note because, as mentioned above, MIDI notes from drum machines are generally very short.) By following the same procedure for the rest of your drum sounds, you can have the drum machine basically act as a specialized sequencer for your sampler.

Of course, you can be creative and assign different sounds to different pads if you want, or if you have a synth with good drum or percussion sounds you can use that in place of, or along with, the sampler. The important point, however, is to use the drum machine's sequencer rather than a dedicated one for this purpose because it's much easier to use a sequencer that's specifically designed for drum patterns.

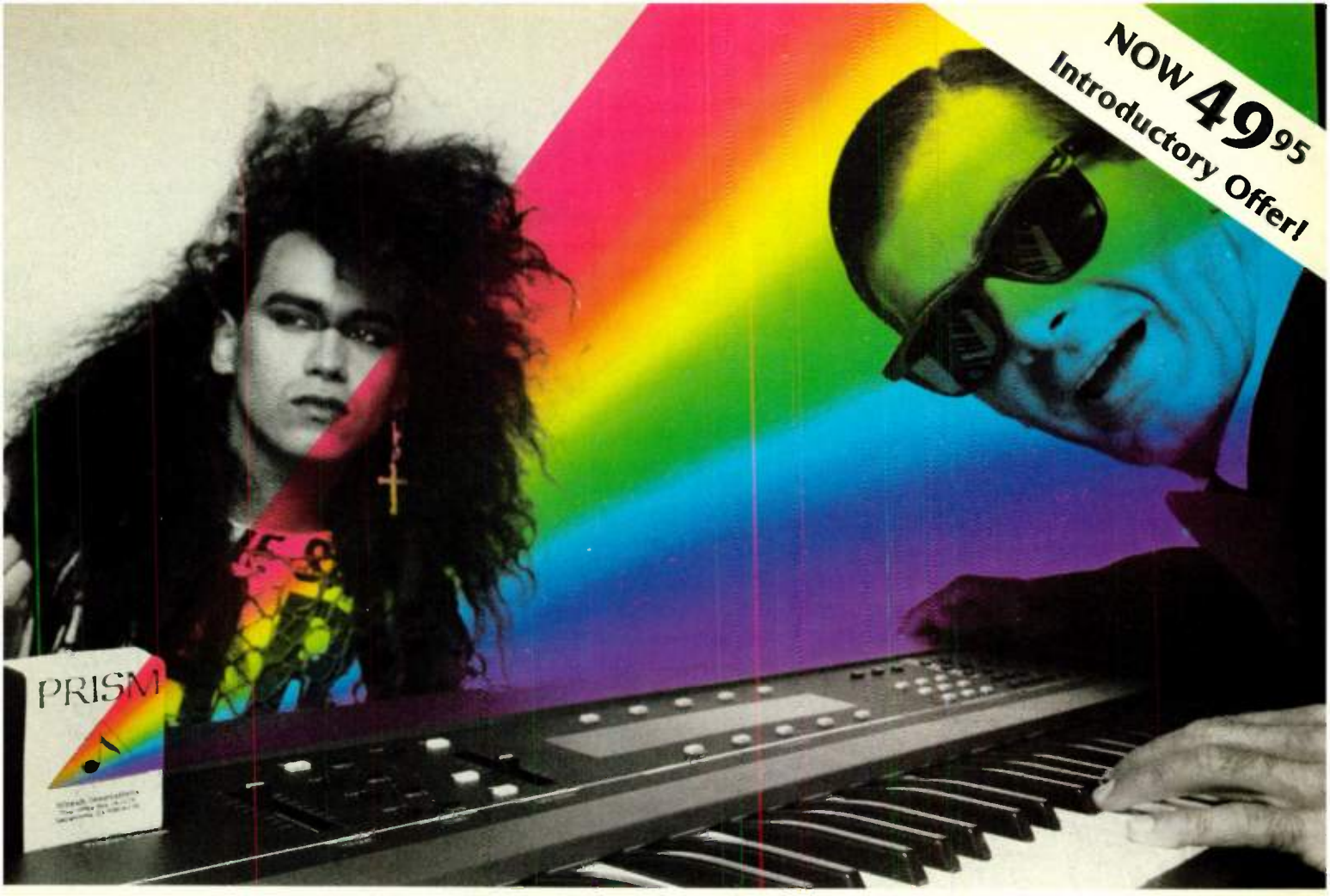
One other point you need to be aware of is that some drum machines allow you to assign drum sounds to any note number you choose and others have preset – and unchangeable – assignments. The former obviously gives you more flexibility, but in either case, you should be able to ensure that note assignments match. Figure 5 shows a typical drum sound layout for triggering from a MIDI keyboard.

If you are happy with the sounds in your drum machine the other application for MIDI note messages is to trigger sounds *in* the instrument from an external controller or another sequencer. As mentioned above, many machines do not have velocity-sensitive pads, but some are velocity-sensitive over MIDI and by sending MIDI note messages from a velocity-sensitive keyboard (say, for example, a DX7) or MIDI drum pads (like a Roland Octapad or Simmons system) with the proper note assignments, you can play these machines dynamically. Some drum machines will also permit you to record the velocity levels from an external controller into the internal sequencer memory for patterns and songs, but if you can't, you'll need an external sequencer to have drum patterns with dynamics.

The basic idea behind this last application is to record patterns with the external controller into the external sequencer and then have the sequencer trigger the sounds in the drum machine. The setup for this requires you to turn the sequencer's "soft thru" (or "echo") on for monitoring purposes so that as you record the data into the sequencer, it will echo back to the input of the drum machine. You'll also need to make sure that you know the note assignments – most drum machines transmit and receive the same note number – and to set the proper MIDI channels. After you've done so, simply record the patterns as you would a normal musical sequence and play them back the same way. Many drum machines allow MIDI control far beyond basic sound triggering, and although those applications are outside the scope of this series, it is worth noting that drum machines like the E-mu SP12 and SP1200 allow you to control instrument tunings with different note combinations, and Sequential's Tom even lets you pan drum sounds left, right, or center with this method.

A small problem that you may run into is the need to constantly stop recording and then overdub and merge new tracks with the original because most sequencers are not structured to allow you to easily record drum patterns. What you lose in ease of use you gain in flexibility, however, because most dedicated sequencers offer more quantization and editing options than are available with the sequencer portion of drum machines. Intelligent Music's Upbeat program for the Mac is actually designed specifically for drum machines and recently several sequencers (such as Hybrid Arts' MIDI Track and Southworth's MIDI Paint) have started to offer drum machine-style recording in addition to the traditional linear method.

Even more applications become possible when you add a drum machine into a MIDI system, but these should provide you with some worthwhile starting points for your own explorations. Next month we'll conclude the series by adding in a MIDI-controlled effects device and talk about tying the entire system together. Until then, experiment! ■



FROM RED HOT ROCK TO COOL BLUES,

THE PRISM VOICE CART

introduces true color sounds for the ESQ-1

All Prism Voice Carts feature our unique "User Switches": simple programming changes that allow the user to modify a voice easily. This eliminates the disappointment of finding that 10 of the 80 voices on a cart are minor variations of the same voice. Other features include an 18 month warranty, masterfully engineered voices, and gold plated contacts. Our E² technology ROMs also have a write protect switch and a data security LED to warn you of the possibility of data overwrite. All this makes the Prism Voice Cart one of the best investments you can make for your ESQ-1.*

Other Products Available from Hitech Innovations

- **Demo sequences—Help you to learn performance technique**
- **E² ROMs—Have read/write capability**
- **Cassette versions of our carts with 40 voices each**
- **Piano 1 & 2 Music Course—Uses the ESQ-1's educational talents**
- **Indian Valley's innovative new MIDI disk drive—128K internal memory, 800K per disk, 4 digit display, 22 key pad, single high rack**
- **3.5" Diskettes for Indian Valley's MIDI disk drive including the same 80 voices found in our carts.**

ORDER NOW:

80 Voice ROM	(A) $49.95 ____	160 Voice ROM	$89.95 ____
	(B) $49.95 ____	Piano Course	(Beg) $39.95 ____
80 Voice E²	(A) $59.95 ____		(Int) $39.95 ____
	(B) $59.95 ____	Cassette Tapes	(ea) $19.95 ____
3.5" Disks	(A) CALL		(2) $35.95 ____
	(B) CALL		(3) $47.95 ____
MIDI Disk Drive	CALL		(4) $55.00 ____

Name _____

Address _____

City _____ State _____

Zip _____ Phone _____

Or Call 1-800-541-0900 (orders only)
DEALER INQUIRIES WELCOME
1-916-362-1449 Info/Dealers

*ESQ-1 is a registered trademark of the Ensoniq Corporation

Hitech Innovations, Inc., Sound Division, PO Box 16-1239, Sacramento, CA 95816

On Stage
SPYRO GYRA

A Moog Micromoog
B Korg DW8000
C Roland RDI000
D Moog Memorymoog
E Yamaha DX7
F Yamaha GS2
G Moog Liberation

Photography *Cindy Manekofsky*

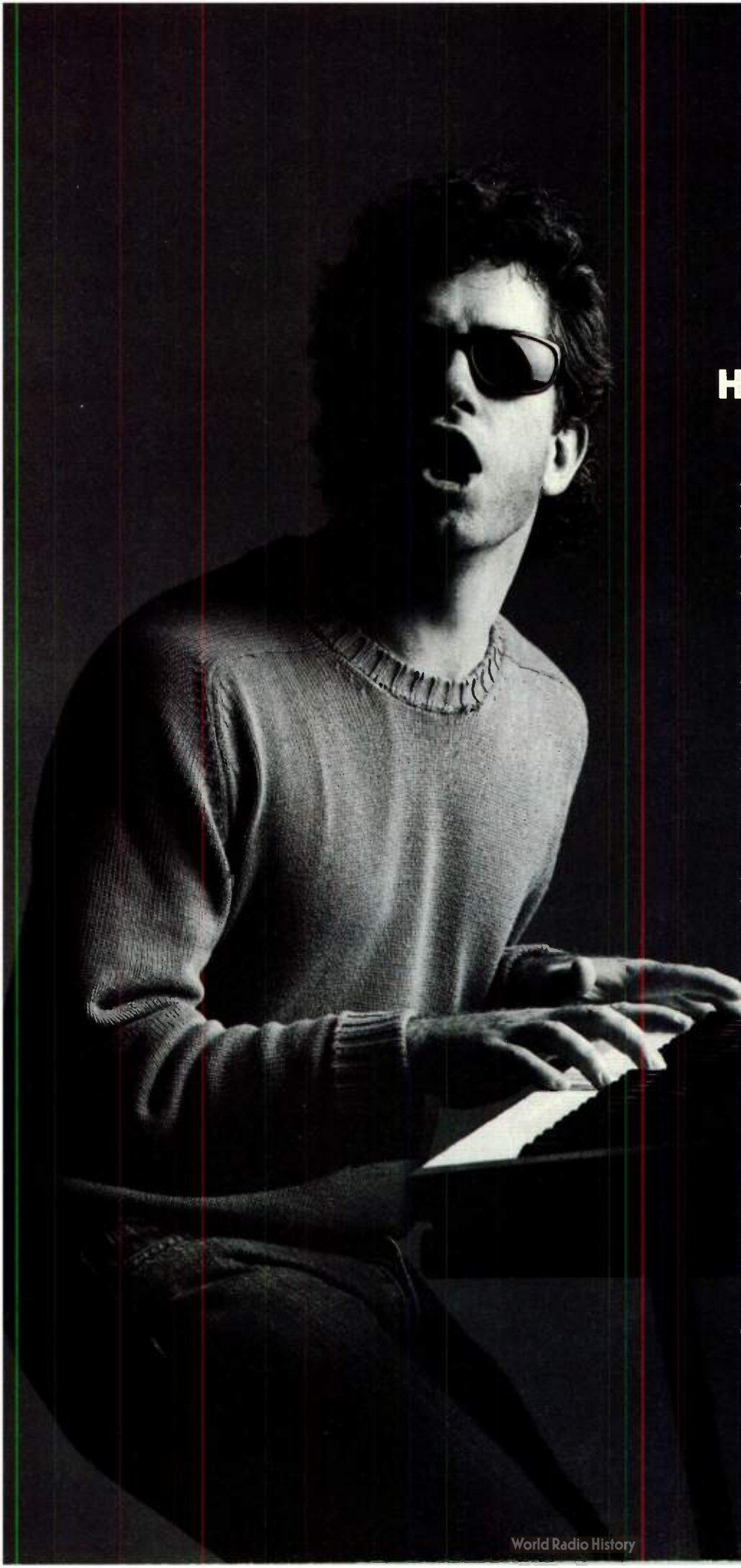
FOR THE LAST ten years we've been kickin' back and enjoying the music and sounds of Spyro Gyra. The man behind the white ivory keys – or in this day and age, white plastic ones – in the fusion group is Tom Schuman, a jazz traditionalist at heart and major member of the seven-piece ensemble.

Tom's background has been as a traditional acoustic jazz player, so his way of adapting to the ever growing technology is to use what he has to the fullest and still keep the sounds as pure as possible. In doing so, Tom has kept a lot of his old keyboards as well as adding what was needed to produce new flavors. Tom's keyboard arsenal currently consists of the following: a Roland Digital Piano (RD1000), which is his main axe (beats dragging around a grand piano and having to deal with the problems of keeping it in tune during outside concerts); two Korg DW8000s; a MemoryMoog (which, yes, stays in tune); a Moog Prodigy; a Yamaha DX7; and two oldies but goodies, a Yamaha GS2 and a Moog Liberation – the latter giving him the freedom to get away from his walls of equipment and get personal with the audience.

During the show I saw on their recent tour I had a great opportunity to watch Tom in action from the sidelines back stage. He used the Korg DW's for a lot of marimba-type patches and the MemoryMoog for some of the smooth, light lead lines. Tom likes the thick fat sounds that the Moogs are able to produce and has perfected performance techniques using the mod and pitch wheels and the ribbon controller.

Tom admits that his first love is the acoustic piano, however, and pointed out that synths only serve to embellish what the player plays. But he might end up MIDIing a grand piano . . . ■

Text by Scott Gershin.



HEADROOM TO BURN.

Let your solos soar.
Let your breaks blast.

No matter how many synthesizer tracks you lay down, no matter how hot the output, a Tascam 300 Series mixer can take it. You'll get greater input sensitivity and more headroom than you'll ever need. So your sound is never limited by what your mixing board can handle.

Thirty years of Tascam engineering make each 300 Series mixer as versatile as it is easy to use. Five complete submix systems cover every multiple or alternate mix situation. Signal routing is easy from the top panel, so you'll rarely need to patch. Back panel inputs allow complete flexibility, providing mic and line, balanced and unbalanced, XLR, quarter-inch and RCA jacks.

But these aren't just studio prima donnas. Pack them up and take them on the road. A solid steel subchassis anchors the electronics, while the rugged casing seals out abuse. They're built to take it, and deliver the same sophisticated sound-handling on stage you'd expect from a prized studio performer.

When you're burning up the tracks, don't settle for a mixing board that can't take the heat. Get a Tascam 300 Series mixer. They love it hot.

TASCAM

REVIEW

Yamaha TX802
FM Synthesizer

Photography Matthew Vosburgh

Flexible multi-timbral capabilities using second generation, six-operator FM voices combined with individual audio outputs, marks Yamaha's latest synth as a winner. *Review by Howard Massey.*

IN MEDIEVAL TIMES, the rack was an instrument of torture. The same can be said today of various synthesizers squeezed into a 19" rack space with convoluted or even non-existent programming capabilities. But here's the good news: the TX802 is one rack-mounted synth that takes us out of the days of the Spanish Inquisition and Cardinal Fang.

Essentially, the TX802 is a rack-mounted DX7IID – not DX7IIFD since there is a cartridge slot but no onboard disk drive. Like the DX7IID – but unlike the TX802's predecessor, the TX816 – there are full editing capabilities onboard. However, the TX802 is capable of generating up to *eight* timbres (called "instruments" by Yamaha) simultaneously – unlike the DX7IID, which only generates two. Each of the eight tone generators has its own audio output (just like in the TX816) and, in addition, there is a stereo audio output.

The Front Panel

LET'S START OUR guided tour with a close-up look of the front panel. Like the DX7II, a back-lit, 2X40 character LCD predominates. It shows not only numeric information, but in a few selected editing operations, actually shows bar-graph displays of data. Immediately below the LCD is a series of eight LED switches. These serve dual functions: in Performance or Voice Select mode, they allow you to turn various tone generators on or off. When a tone generator is left on, the LED in the switch blinks whenever a MIDI Note On signal is received. Happily, unlike the TX816, these switches do *not* blink when a note off is received, making for a less frantic display when you are actually playing notes. In Edit mode, the LED switches allow you to choose various menus of parameters. And, when you press an Edit mode switch, the LCD line immediately above these eight LED switches shows the function of each one, so no memorizing is required.

To the left of the LCD is the power on/off switch and – hallelujah! – a headphone jack. Immediately to the right of the LCD are a series of eight mode select switches. The top four bring you into any of the four main modes: Performance Select, Voice Select, System Setup, or Utility. Beneath these are three switches enabling you to edit performance parameters or voice parameters (these are split into Voice Edit I and Voice Edit II so that each parameter has sufficient display space – the Voice Edit

II parameters are basically what used to be function controls in the DX7), and a "Store" switch for writing data to either internal or cartridge memory.

Below these mode switches is the ubiquitous cartridge slot. It holds the same ROM and RAM4 cartridges used in the DX7II and can also hold original DX7 ROMs and RAMs with the use of the optional ADPI adapter. The TX802 can access voices created in either the DX7 or DX7II but cannot access performance memories created in the II. Its own formatting procedure sets up the cartridge to hold either 64 voices plus 64 performances (as opposed to the 64 voices plus only 32 performances that the DX7II formatting procedure allows for) or the usual 64 fractional scalings or 63 micro tunings. The addition of a cartridge slot alone gives the TX802 an edge over the TX816, where voices could only be transferred via MIDI – and, if you've ever tried it, you'll know that it's much faster and simpler to call up voices by just plugging a cartridge in.

Finally, to the right of the mode switches and cartridge slot is the sixteen-key keypad. This not only contains the usual increment-decrement ("Yes-No") switches and Internal-Cartridge Select (doubling as cursor) switches, but also has ten dedicated number keys (doubling as letter keys when naming voices) and – for the first time on any Yamaha synthesizer – an "Enter" key. Yes, you read right: an "Enter" key. These little beasties, found in almost every other manufacturer's design, have been conspicuously absent from Yamaha synths (OK, so they're on some of the Yamaha digital reverb units . . .) until now. Now, there are two schools of thought regarding the existence of such a switch on a synth. One maintains that it simply adds another annoying step to the editing process, while the other holds that it acts as a safety switch. Whichever school you belong to, you'll probably be glad it's there the first time you either type in the wrong two numbers into the correct parameter or the right two numbers into the wrong parameter. Until you press the "Enter" switch, they won't actually get entered and you won't hear any change in the sound.

As a visual aid, whenever you see a number blinking in the display (as opposed to staying steady), you must press "Enter" in order to send the data change to the microprocessor – at which time the number in the display stops blinking. Note that there is *no* data entry slider on the TX802, since its function is efficiently replaced by the ability to directly enter in numbers from the keypad. Those of you who have long been editing DX voices with the use of an external patch editor are probably already used to direct number entry – and those of you who haven't will just have to take our word that it really does speed things up.

The rear panel is very straightforward, containing MIDI In, Out, and Thru, the two stereo output jacks, and the eight individual audio outputs. All audio outs are ¼" jacks, as opposed to the XLR connectors used by the TX816. While the individual outs make the 802 an instrument worthy of a double rack space in any serious studio, the stereo outs also make it useful in a home or live performance setup – giving you the best of both worlds.

Multi-timbral *"The TX802 is capable of generating up to eight timbres simultaneously – unlike its closest companion, the DX7IID, which only generates two."*

Operation

NORMALLY, THE TX802 will be played in Performance Select mode. Each of the eight tone generators ("instruments") is capable of only two-note polyphony, but you can link instruments together to get up to sixteen notes at a time from any one instrument. Therefore, you could have, for example, one instrument with two-voice polyphony, another with two tone generators linked together for four-voice polyphony, and a third one with five tone generators linked together for ten-note polyphony. Or, in fact, any combination of even-numbered polyphony up to a maximum of sixteen voices.

When you are in Performance Select mode (accessed by pressing the upper left-hand switch in the eight switches to the right of the LCD), all you can really do is call up different performance memories with the use of the keypad or remotely via MIDI, or turn instruments on or off.

By pressing the mode switch below the Performance Play switch – Perform Edit – you can edit the performance parameters. At this time, the LCD changes to show the functions of the eight menu select switches (the instrument LED on-off switches) immediately below. The first of these allows you to set the MIDI receive channel for the individual instruments, each of which can listen to any of the sixteen MIDI channels or to all in Omni mode. When adjacent instruments are set to receive on the same MIDI channel, they can also be "alternately assigned" by pressing and holding Enter while pressing the "On" switch. Doing so will cause a short arrow to appear to the left of the voice number in the display. When you do this, each incoming note will rotate between the ▶

REVIEW

► alternately assigned instruments in ascending order – kind of like some voice assignment modes we used to see in Sequential and Oberheim instruments. The difference, of course, is that you can have radically different timbres for each instrument, or, for more subtle effects, you can set up an alternate assignment between instruments that are only slightly different. The instrument even has a simple procedure that will "confuse" its microprocessor so that the assignments won't always alternate in ascending order – yielding a pseudo-random sort of effect.

Other editing operations here allow you to: turn EG forced damping on and off (kind of like selecting between single and multiple triggering in an analog synth); set the independent volume of each instrument in a performance memory (when incoming MIDI volume change commands come in, the balance set here is preserved); set the low and high note limit of each instrument (thus allowing you to set up involved keyboard splits and layerings from your master keyboard while transmitting on only a single MIDI channel – this is particularly useful when using a DX as your master keyboard); shift the incoming note data by semitones over a range of plus or minus two octaves; select a micro tuning table (and *each instrument* within the TX802 can actually access a *different* micro tuning table); slightly detune an instrument (useful when using multiple instruments which have the same patch in them); and name the performance memory (accomplished with the keypad).

You can also store any performance memory to any of sixty-four internal or cartridge slots by pressing the "Store" switch and then selecting the destination number with the ten-key keypad. Although the factory performance presets are not stored permanently in ROM (though voices are), a booklet enclosed with the owner's manual shows all of the performance edit parameters for each of the sixty-four factory presets, so you can always reconstruct them afterward if need be.

The TX802 also has a parameter here that allows you to assign each instrument to either or both of the stereo outputs. This is particularly useful with alternately assigned voices, since you can have repeated key depressions call up the same or radically different timbres which can then appear from different left-right points in the stereo spectrum. This is no substitute, of course, for the true stereo panning which is available in the DX7II – it's completely *unavailable* here – but at least you can get *some* movement in the sound without having to resort to outboard signal processing. It's worth noting that assignment and volume changes made in edit mode will not alter the output sent from that instrument's individual audio output, which is always sent full blast under all circumstances.

REVIEW

Displays *"Alternate displays are also available for both the operator EG settings and operator output level, and both of these take the form of bar graphs."*

Voice select mode enables you to select individual voices for use within a performance memory. The TX802 contains two banks of sixty-four permanently stored ROM voices – labeled as banks A and B – along with another bank of sixty-four RAM voices – labeled as bank I, for "internal". Additionally, you can access another sixty-four voices from a cartridge in the slot, and this is known as bank C. You specify a particular voice to be viewed or changed by using the keypad cursor keys, and one nice touch here is that not only is the selected voice name displayed on the top line of the LCD, but the MIDI receive channel for that voice is displayed as well. You cannot alter this here – Receive Channel Select is a performance edit parameter – but it's nice to be able to move the cursor along the different voices and instantly see what channel each is listening to – particularly if you're driving the TX802 from a sequencer.

From this display you can also link instruments together to create polyphonies of greater than two voices. You accomplish this by positioning the cursor over the tone generator you want to link, and pressing and holding "Enter" while pressing "Off." In place of a voice number, you'll see a long arrow pointing to the tone generator to the immediate left (obviously, you can't link the left-most instrument), which will then be able to play two additional notes. Instruments that are linked or in alternate assign mode will turn on and off together when using the eight LED on-off switches. From voice select mode, you can also store a specific voice to an internal or cartridge memory slot by using the "Store" switch. It's also worth noting that, like the DX7II, if you call up a voice written with either fractional scaling or micro tuning data and the appropriate cartridge is not in the slot, a visual indicator in the LCD – in this case, a blinking "f" or "t" – will remind you of that fact.

Voice Editing

AS MENTIONED EARLIER, there are two voice edit modes, labeled "I" and "II."

Voice Edit I contains all of the basic FM voicing parameters, including algorithm number, feedback level, oscillator key sync, transpose, voice name, oscillator frequency (coarse, fine, and detune), operator and pitch EG values (the same four levels and rates used in the DX instruments), output level and level scaling (including both normal and fractional modes), operator sensitivity (to keyboard velocity, pitch modulation, or amplitude modulation/EG bias modulation), and LFO settings. Worthy of mention is the fact that the TX802 offers three unique displays in addition to the standard numeric displays which are exactly identical to those of the DX7II.

Voice Complexity *"The TX802 actually allows you to hear all eight instruments while editing one, so those of you with a pioneering and/or masochistic streak can approach it as a forty-eight operator synth(!)."*

The first of these is an alternate frequency display that not only shows the mode (ratio or fixed) and coarse and fine frequencies for all six operators in the one display but also indicates which operators are carriers and which are modulators in the algorithm you are currently using. Unfortunately, there is no visual indication as to which operators are modulating which other operators. To learn this information, you still have to reference the algorithm diagrams, provided on a pull-out sheet attached beneath the unit. The currently selected operator (the one whose data will change if you make a change with the keypad) is indicated with a blinking operator number.

Alternate displays are also available for both the operator EG settings and operator output level, and both of these take the form of bar graphs. In the EG display, the four EG levels for all six operators are shown at once – with no indication of rate values – while in the alternate output level display, there is a bar graph representing the output of the chosen operator over an 87-note range – that is, 29 three-semitone groups – of the keyboard. This clearly shows the effect of any keyboard level scaling being used, and is really useful when working with the relatively new area of fractional scaling.

For example, when changing the offset value in the fractional display, you see all the bars change at once. When working in fractional scaling mode, the currently selected note group is indicated with a blinking bar. True, these are relatively low-resolution bars and no useful numeric data can be derived from them – you can easily get the numbers themselves by simply pressing the parameter switch again to return to the numeric display – but it's great to have some kind of graphic representation of what is actually being accomplished when changing edit data. Now why can't these same displays be included in the DX7II?

It's probably worth mentioning at this point that all TX802 editing operations can be carried out either from the unit's own keypad, or directly from a DX7II which is connected to the 802 via MIDI. In fact, you don't need to even look at the 802 display while using this latter method – though you'd probably want to use the bar graph displays – since simply calling up and making a change to a parameter with the data entry slider or "Yes-No" switches of the DX7II will cause that same parameter to be called up and changed instantly in the 802. Isn't compatibility a wonderful thing?

The Voice Edit II options allow you to program several of the parameters offered by the DX7II edit switches 23 through 26. Setting the key mode is one of these functions, but it's a little different than on the DX7II because only poly and mono are offered here: there is no unison poly or mono – don't forget, though, that you can detune tone generators in performance edit mode. Another display lets you determine the range and step of the pitch-bend wheel – but the alternate high, low,▶

REVI

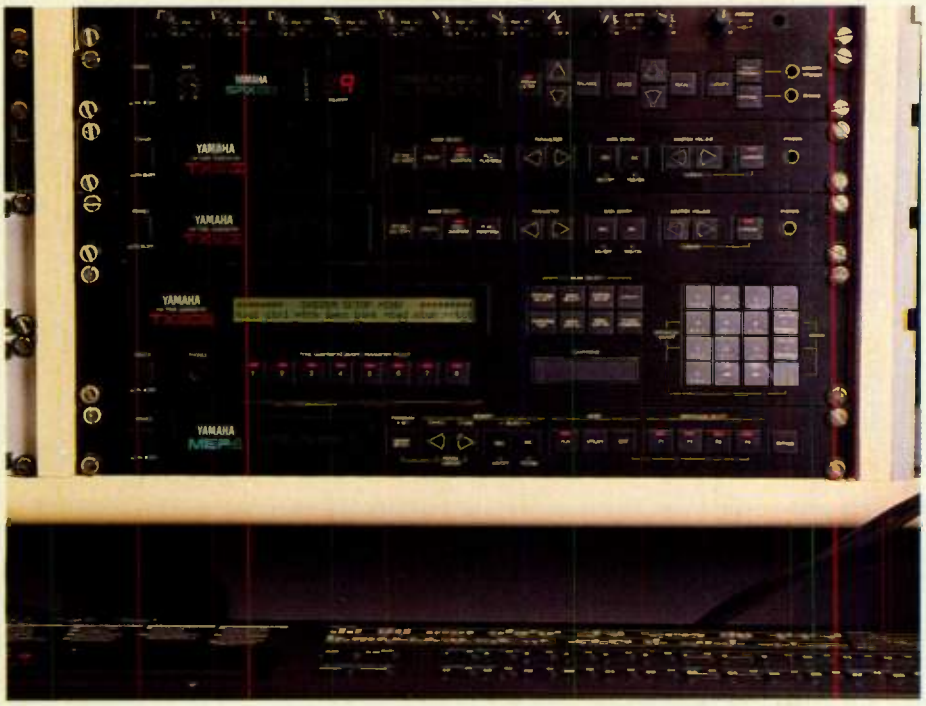
▶and key on modes offered by the DX7II are not supported. Remember, however, that each individual TX802 instrument can have different pitch-bend wheel settings. Like the DX7II, the TX802 allows you to determine the depth of a random pitch effect, and to set various portamento modes, steps, and times. Last, but by no means least, are the routings for the modulation wheel, foot controller, breath controller, and keyboard aftertouch.

Naturally enough, these controller routings seem to presume that you are in fact using a Yamaha keyboard as the master controller. You're not? Never fear – Captain MIDI is here! The TX802 System Setup menu allows you to reassign *any* incoming controller data to take on the functions of the mod wheel, breath controller, or foot controller, or to directly control either volume or portamento time, or to act as either a sustain switch or portamento switch. In other words, you can pretty much adapt any controller to the 802, but it would have been nice to see even more options offered – for example, allowing incoming controller data to take on the aftertouch or direct EG bias functions.

Other Functions

THE SYSTEM SETUP menu allows you to specify how incoming MIDI control change, program change, aftertouch, or pitch-bend messages are assimilated by the TX802. Each of these can be selectively filtered, or any of them can be received by each instrument on its own channel – ideal for sequence control – or via an assignable "global" MIDI channel – ideal for live performance when you're using a single keyboard controller. Like the DX7II, the TX802 can respond to all incoming note messages, or to just odd or even numbered notes – a clever trick that enables up to 32-note polyphony when two TX802s or an 802 and a DX7II are used together with the same voices in memory and one synth set to receive even notes while the other receives odd ones. Another option in this menu lets you reassign incoming program change messages if desired. And last, but not least, is the micro tuning edit display – similar in every respect to the equivalent display in the DX7II – and the master tuning and internal memory protection controls. Interestingly enough, there is no software cartridge memory protection switch in the TX802 – only the hard switch on the cartridge itself.

Finally, the Utility menu allows you to selectively transmit system exclusive data. Like the DX7II, any area of memory – voices, performances, micro tunings or system setup – can be transmitted, and so too can the contents of the edit buffer, meaning that you don't have to necessarily store data before you can transmit it. The



Utility menu also contains options to save to or load from cartridge, as well as the cartridge formatting procedure. Here you can also initialize the voice or performance edit buffer or the incoming program change or controller reassignments. A Recall Edit function – for either voice or performance data – is also available and a copy function. It's worth taking a minute to describe this one because it's substantially better than any copy function available on previous "X" machines, and nearly as good as that available on most patch editing software. Three options are offered: either you can copy EG and scaling data (both rate and level scaling, plus output level) from operator to operator; or you can copy the oscillator mode, frequency (coarse and fine) and detune; or you can copy all of the above, plus velocity and amplitude modulation sensitivity. A good start, but it would be useful to have even more options, because there are times when you want to copy output level only or EG data only without the other parameters coming along for a ride.

Like other DX instruments, there is an instantly accessible Compare mode from any of the editing menus, and this, combined with the Recall Edit function and ease of the keypad makes editing on the TX802 a simple and painless procedure. The DX7II instruments allow you to hear both voices at once while editing either, thus allowing you to approach them as true twelve-operator synthesizers. The TX802 takes things a step further and can actually allow you to hear all *eight* instruments while editing one – thus allowing those of you with a pioneering and/or masochistic streak to approach it as a *forty-eight* operator synth(!).

And, in the Don't These Clever Nippon-Gakki Engineers Think Of Everything Department, try this scenario on for size: you'd like to buy a TX802, but you don't have a standard 19" rack to mount it on. No problem, kemosabe: there's even a little tilt-stand mounted underneath to make table-top use simpler. Whew!

Conclusions

WITH EIGHT INDIVIDUAL outs and an easy-to-use and understand multi-timbral operation, the TX802 distinguishes itself as *the* FM synth to own, particularly if you do any sequencing work. And unlike almost any other rack version of a keyboard synth, the TX802 actually offers *improved* editing displays and simpler programming operations, thanks in large part to its sixteen-key keypad.

Frankly, it's hard to find too many substantive complaints. There's no stereo panning, for one thing – even though, as mentioned above, there *are* stereo outs – and there is a total polyphony of only sixteen voices – as opposed to the over-the-top 128 voices available in the TX816. Naming voices and performance memories is even more of a pain here than in the DX, but how often do you name voices, anyway? Apart from these minor points, however, this really is a hell of a synthesizer.

All in all, the advanced programming features of the TX802 – not to mention its excellent sound quality – make it worthy of inclusion in setups of all sizes and sophistications. Tie me to the rack! ■

PRICE $1895

MORE FROM Yamaha Music Corporation, PO Box 6600, Buena Park, CA 90622-6600. Tel: (714) 522-9011.

Yamaha has been hearing voices again.

And you can hear all 240 of them for only $345.*

Presenting the Yamaha FB-01 FM Sound Generator. A surprisingly compact black box containing the largest selection of Yamaha digital FM voices since the introduction of the DX synthesizer.

Its incredible affordability is even more dramatic when you consider that in addition to 240 pre-programmed voices, the FB-01 accommodates 96 user voices. For a total of 336 voices at the unheard of price of only $1.02* each.

But what is even more incredible is the amazing versatility the FB-01 affords the amateur and the professional musician alike.

Each natural acoustic instrument or synthesizer voice features independently programmable functions such as pitch bend, detuning, octave transpose, and up to eight 4-operator digital voices simultaneously.

Plus four preset configurations that enable the FB-01 to be used as a single-voice 8-note polyphonic sound source, as well as a split sound source and dual sound source. And a Mono-8 mode offers monophonic performance of up to 8 voices simultaneously.

You can also create your own configurations and then store them in one of the FB-01's 16 user memories.

A lit LCD exhibits all performance data, system set-up, and voice names with dozens of operations carried out via a simple key layout on the front panel. The FB-01 is designed for ease of operation by any musician, even those without any detailed knowledge of MIDI or computers.

Its exceptional adaptability makes it well suited for MIDI applications ranging from live performance to computer music systems, MIDI recorder and sequencer systems, and MIDI studio systems.

Weighing in at a slight 4.6 lbs., the Yamaha FB-01 provides left and right stereo outputs and is also capable of microtonal tunings by computer.

All the more reason for you to stop by an authorized Yamaha Digital Musical Instrument dealer today for a complete demonstration. So you too can soon be hearing voices.

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. JOrgan 1
2. Horn	2. SPiano	2. Horn 3	2. Fnk Syn 3	2. JOrgan 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. Sml Pipe
12. EOrgan 1	12. EGrand 2	12. Hard Br 4	12. Mono Syn	12. Mid Pipe
13. EOrgan 2	13. Honkey 1	13. Huff Bra	13. Cheeky	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 Pluk	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. Fnk Gt
18. Oboe	18. New EP 3	18. String 3	18. Sol Bass	18. Pluck Gt
19. Clarine	19. New EP 4	19. String 4	19. Pluk Bas	19. Brite Gt
20. Glasken	20. New EP 5	20. Solo Vio	20. Uprt Bas	20. Fuzz Gt
21. Vibes	21. EPiano 1	21. Rich St 1	21. Frebes	21. Zither 2
22. Xylophn	22. EPiano 2	22. Rich St 2	22. Flap Bass	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 Strg 3	27. Syn Bas 4	27. Sft Koto
28. Harp	28. Perc Pf	28. Lo Strg 4	28. Syn Bas 5	28. Hit Koto
29. SmadSyn	29. Wood Pf	29. Lo Strg 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. 5th Str	31. Marimb 2	31. Huff Syn
32. Timpani	32. Clav 2	32. Piano 1	32. Marimb 3	32. Fantasy
33. Lo Strg 2	33. Clav 3	33. Piano 2	33. Xyleph 2	33. Symphnic
34. Horn Lo	34. Clav 4	34. Flute 2	34. Vibe 2	34. Mt. Voice
35. Whistle	35. Fuzz Clv	35. Flute 3	35. Vibe 3	35. VSAR
36. Zing Pip	36. Mute Clv	36. Flute 4	36. Glockn 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 Be 2	38. Wild War
39. Funk Syn	39. Syn Clv 2	39. 5th Flt	39. Bells 2	39. Ghostie
40. Voices	40. Syn Clv 3	40. Ocve 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. Chick	45. Smash
46. Mars to	46. Clavinet	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

Banks 1 and 2 can be used for storage of 48 original voices each.

Yamaha Music Corporation, Digital Musical Instrument Division, P.O. Box 6600, Buena Park, CA 90622. In Canada, Yamaha Canada Music Ltd., 135 Milner Avenue, Scarborough, Ontario M1S 3R1. *USA suggested retail price subject to change without prior notice. Canadian price will vary.

YAMAHA®

IN PAST ISSUES

NOVEMBER 1986

• Synth king Howard Jones talks recording and programming, David Sylvian comes clean, and we track down The Philip Glass Ensemble, and touring keyboardist Greg Whelchel.

On test are Akai's MX73 master keyboard, the Ensoniq Digital Piano, the Unique DBM controller keyboard, the Fostex E16 multitrack, Dynacord's ADDone drum module, the Korg DDD1 drum machine, Steinberg's Pro24 sequencing software for the Atari ST, Yamaha's FB01 FM synthesis module, and the Roland DEP5 multi-processor.

We get the lowdown on mixdown, examine the SPX90's MIDI Mode 4, and continue our guides to sequencing, FM programming, and creative sampling.

JANUARY 1987

• Synth-pop pioneers, OMD, feature alongside Heaven 17 and avant-garde composer Morton Subotnick.

The review roll-call includes the E-mu Emax and Akai X7000 sampling keyboards, the Kawai R100 drum machine, Yamaha's QX5 sequencer, Stepp DG1 guitar synth, Tascam Porta Two, and Steinberg's ProCreator software for the Atari ST.

We start an overview of hard disk technology, conclude our series on creative mixing techniques, and see how guitarists can benefit from MIDI Mode 4.

FEBRUARY 1987

• The enigma of contemporary music – Frank Zappa – talks tech in a rare in-depth interview and Robert Irving III, keyboardist with Miles Davis, tells how he coped with the pressure of following in the footsteps of Herbie Hancock and Chick Corea. Also featured is Japanese composer/arranger Seigen Ono.

Reviews include the Oberheim DPX1 sample replay unit, Roland RD300 piano, Jam Factory, Dr. T's KCS software, and the revolutionary Mandala video-MIDI instrument.

On the feature front we continue our investigations into the worlds of hard-disk technology and MIDI Mode 4 for guitarists, and take a look at how modern percussionists can keep in time with the times.

MARCH 1987

• We reveal why Adrian Belew is one of the world's most innovative modern guitarists, and the unusual method employed in recording Peter Hammill's latest album. Also in conversation are jazz-fusion keyboardist Jeff Lorber and film-soundtrack composer Michael Stearns.

Being put to the test are the Yamaha DX7II, Sequential Studio 440, Korg SG1 piano, Simmons SPM8:2 mixer, Barcus Berry processors, and MegaMix and Intelligent Music's 'M' software.

Bandwidth jargon is deciphered, MIDI Modes are dissected, and experiments carried out to the Sample Dump Standard.

MAY 1987

• Allan Holdsworth, one of the leading advocates of guitar synthesis, and eccentric composer Holger Czukay, talk to our intrepid reporters, while Kim Ryrie, Fairlight's co-founder, and Steven Randall, designer of the Stepp guitar, speak frankly about their technological achievements.

On test are the Casio SK2100 keyboard, Roland MKS70, Yamaha RX5, Kahler Human Clock, Alesis MIDIverb II and ART DR1 reverbs, and Texture and ClickTracks software.

We assess the Apple Mac II, continue our drum programming series, and explain how your sampler can produce stereo samples. ►

PLEASE SEND ME the following issues of Music Technology (check issues required) at $2.95 each, plus $2.00 shipping/handling.

☐9/86 ☐11/86 ☐1/87 ☐2/87 ☐3/87 ☐5/87 ☐6/87 ☐7/87 ☐9/87 ☐10/87

☐ I enclose a check for $_____

☐ Charge my credit card: ☐ Visa ☐ MasterCard

Card number ..... Expiry date .....

Signature .....

Name .....

Address .....

City ..... State..... Zip .....

Checks should be made payable to Music Maker Publications, Inc. The address to send to: Music Technology Back Issues, 7361 Topanga Canyon Blvd, Canoga Park, CA 91303. Back issues charges to Visa/MasterCard can also be ordered directly on Tel: (818) 704-8777.

JUNE 1987

• Fusion pioneer and former Mahavishnu violinist Jerry Goodman discusses his new work with us, and Bill Bruford, characteristically in the vanguard of new technology, talks about electronic percussion as an art form which is distinct from acoustic drumming. We also interview Richard Horowitz and Sussan Deihim on their unique, new electronic and vocal recordings.

Our reviews check out the Casio FZ1 sampling keyboard, the Roland D50 synthesizer (Part 1), 360 Systems MIDIMerge+, Steinberg's Cosmo software, and the Kawai R50 and Korg DDD5 drum machines.

We also find some ways to speed up Macintosh MIDI workstations, get the story straight regarding MIDI delays and we conclude our interview with Kim Ryrie. The third part of our series on creative drum programming assesses the relative merits of different trigger-to-MIDI converters.

JULY 1987

• Ryuichi Sakamoto, Yellow Magic Orchestra's founder, shares perceptions on his latest album and plans for the future. Geoff Downes, former Yes and Asia keyboardist, discusses his personal history with synthesizers, while Cutting Crew talks about their unique blend of guitars, guitar synths and other music machines.

Our in-depth reviews highlight the Korg DS8 synth, 48 Track PC II, Yamaha TX81Z and MDF1, Passport Master Tracks Pro, as well as Part 2 of our in-depth look at the Roland D50.

The fourth segment of our series on creative drum programming looks at electronic drum pads, and the DX7IID upgrades are examined. We begin a new series on recreating the sound of acoustic instruments with a focus on the trumpet.

SEPTEMBER 1987

• Synthesist and composer Steve Roach discusses the fine art of programming, and Level 42 keyboardist Mike Lindup and co-producer Wally Badarou talk about their rise to the top in two separate interviews.

Reviews uncover Iota Systems' MIDI Fader, Yamaha's TX802 FM synth module, the Bacchus TX81Z Editor/Librarian for the IBM, the Kawai R50, and Grey Matter Response's E! Version 2.0 upgrade for the DX7. We also offer in-brief looks at the Alesis HR16 & MMT8, Sequential's Prophet 3000, and Perfect Patch from Aegix.

The sixth part in our series on drum programming examines new hand-held percussion controllers, and the third installment of our series on synthesizing and sampling the sound of acoustic instruments focuses on the snare drum. We begin a new series on the art of looping and offer an article on the ways and means of just intonation.

OCTOBER 1987

• The incomparable Todd Rundgren talks shop, Vienna's Peter and Ina Wolf discuss their new projects and methods, and Living in a Box reveals the technology behind the tunes.

On test are the Roland MT32 expander, Hybrid Arts' ADAP, the IBM Music Feature, Garfield's Time Commander and Yamaha's QX3 sequencer and REX50 signal processor. In-brief reviews are included on Akai's EWI/EVI wind controllers, JL Cooper's MAGI and the M8000 master keyboard from Kawai.

We say a fond farewell to our series on creative drum programming with the seventh and final installment, and present Part II of our series on looping techniques. Part III of our series on MIDI basics looks at sequencers. ■

There's no doubt that the Japanese musical instrument industry has had a profound effect upon Western music and, in particular, Western equipment manufacturing. But unfortunately, not many people know or understand what actually goes on behind the Bamboo Curtain.

Text by Hugh Ashton.

MADE IN JAPAN – words to strike terror into the heart of any Western manufacturer. For as soon as the West comes up with a good idea – the synthesizer, the polyphonic synthesizer, the drum machine, MIDI, the sequencer or whatever, you can bet your last yen that the Japanese will take the idea, refine it, add bells and climb onto the market – your market – with a better product at half the price.

I was fascinated by this – after all, what tradition do the Japanese have of electronic music and of pop music? So what makes them so successful at producing and selling such an alien form of technology? When I was invited over to Japan on a completely different matter, I decided to investigate the Japanese popular scene in general, and the electronic music industry in particular. The following is the result of these investigations, based on talks and meetings with Japanese manufacturers, retailers and musicians.

Why are Japanese synthesizers so successful? How can they be made so cheaply, and so reliably, with new improvements and models appearing every year? To answer these questions, it is necessary, I'm afraid, to go a little into Japanese industry and Japanese society.

The average Japanese company is a much different affair than its US counterpart, and will have a much more diverse range of interests. Yamaha, for instance, is known as a manufacturer of motorcycles, stereos and musical instruments (the Japanese name "Nippon Gakki" means "Japan Musical Instruments"). But did you know that Yamaha is also one of Japan's largest furniture manufacturers, and also makes bathroom fittings? Neither did I, until I found myself bathing in a bathtub labeled "Yamaha." Even Roland, which is one of the more specialist instrument manufacturers, makes computer plotters and Computer Aided Design (CAD) software. Looking at this from the perspective of music being the number one interest (which may not always be true), the non-musical product lines serve two purposes: first, they ensure a steady flow of income when musical instruments are selling slowly, and second, the low-technology items with their low research and support budgets can help to finance the hi-tech music R&D (Research and Development).

R&D itself is carried out in a way somewhat different to most Western companies. A team will be given responsibility for the design and production of a product. Within this team, all the necessary work will be done, from chip design to printed circuit board layout to software to mechanical assembly to production engineering. The manager of the R&D team will also have responsibility for production of the item. This eliminates some of the problems that may beset a product designed in a different way, such as: the circuit board will not fit in the case designed by the industrial designer, or the software designed for the hardware prototypes will not work with the production version, or even that the finished product is too complex to build to a sufficiently high standard in any kind of quantity. In case you're thinking these are absurd examples, by the way, these are all design faults that I have experienced in my time in the high-technology industry.

The danger of this system is that a number of different groups may simultaneously be engaged in "reinventing the wheel." To avoid this, a controlling R&D group will be set up which holds regular meetings with the project groups to impose a "house style" on the designers, and to save wasteful duplication of effort.

The average Japanese worker is often praised for his long working hours, his short holidays, and his loyalty to the company, by those who should know more about the subject. All these things are true, but the reasons behind the truth are somewhat more complex than what you may expect. The average Japanese office worker does work long hours – factory workers not as long. However, in those long hours he or she is less efficient than his or her Western counterpart. Part of Japan's high employment rate is due to jobs being accepted which would not qualify as "proper" jobs in other countries. Five men to see one car out of a multi-level car garage? One to take the money, one to wave the car on, one to wave the car onto the road (complete with baton and whistle to stop the traffic), and one on each side of the pavement near the exit to halt the pedestrians. There is also a high social status to be gained from working long hours. Men will sometimes spin out their work in order to return home late.

Short vacations? True, but not necessarily from any real liking for work. Japanese people are brought up in a more group-orientated environment than Westerners – co-operation, not competition, is the golden rule, and many Japanese people claim to be only truly happy when in a group of like-minded people. The ▶

► reluctance to take vacations may therefore be a reluctance to leave the group. As a result, the vacations that are taken may be taken as company trips.

This leads neatly to the last point – loyalty to the company. In traditional Japanese industry (though this is changing) once a white-collar worker was hired, he was hired for life, could not be dismissed and was expected not to seek work outside the company. In return, the company rewarded him with subsidized housing, welfare, vacations and even a place in the company cemetery should this become necessary.

An Australian design engineer working for a Japanese company summed it up: "I work the same hours as I did in Australia," he told me. "10-hour days and a six-day week. The difference is that here I feel appreciated." This is probably why employees in Japan feel loyal – it's a two-way thing.

Turning Japanese

ALL THESE REASONS suggest ways in which Japanese industry can produce the hi-tech marvels which come flooding out of *Dai Nihon*, but is there a domestic demand for them in Japan? You bet. Wandering around Akihabara (the electronics retail district of Tokyo), I went looking for the most highly sophisticated useless objects I could find. Prizes went to a hi-fi graphic equalizer and spectrum analyser – controlled by a light-pen (even more ridiculous when you consider the size of the average Japanese room), a cassette player the size of a Walkman which also incorporated a TV set, and a pocket photocopier.

All electrical goods cost slightly less than the average US price for equivalent items, but the Japanese have a much higher disposable income than Westerners. So if you already have a video recorder, there is nothing to stop you from buying this month's model and throwing out your old one, even if it's only six months old. And "throwing out" means exactly that. There's no market for second-hand hi-tech goods in Japan. Many foreigners living in Tokyo furnish their homes with the second-hand microwaves, TVs and videos from the garbage waiting to be taken away. I looked, but I didn't see any musical instruments lying around, I'm afraid.

Hi-tech musical instruments fall into the same category for most Japanese as the other items – "technotoys." And like toys, they're bought primarily by and for the young. Yamaha and Roland claim that their expensive keyboards are bought by indulgent parents for their offspring, but Casio believes that the high end of the market is the kids themselves using savings and money from part-time jobs. Believe who you like – the fact remains that it's the kids who play them.

"Hi-tech musical instruments fall into the same category as other electrical items – 'technotoys.' And like toys, they're bought primarily by and for the young."

The Japanese music scene is in a rather sorry state. You all know that the Japanese make the best affordable synthesizers (hands up anyone who can afford a Synclavier or a Kurzweil), but the best synthesizer players? I'm afraid not. The reasons behind this are again, not simple (nothing about Japan ever is), but are linked in many ways to the description of industry above.

Like so many countries outside North America and Europe, Japan is infected by the West-is-Best attitude, and this applies as much to popular music as anything else. In Hamamatsu, home of Yamaha, Roland and Kawai, a town of some 500,000 inhabitants (somewhat small by Japanese standards), there is a shop selling nothing but Beatles records. It is almost impossible to imagine such a shop surviving in the US (unless it sold to Japanese tourists). This attitude leads to an almost absurd reverence for Western music at the expense of whatever homegrown talent may be available.

The Japanese equivalent of Music Technology, *Keyboard* – no relation to *Keyboard* produced here in the US – (notice how even the title of the magazine is in English though), is dedicated to much the same sort of articles as is Music Technology. One or two important differences do exist, however. Reviews of new products are not reviews in the way in which we understand them. It is not in the best interests of the community to criticize manufacturers' goods. The reviews are therefore more a description of the features available than a recommendation to buy or avoid a piece of equipment. Prices are fixed by the manufacturers (at least in theory), and there is no point in looking through the dealers' advertisements for bargains – there aren't any. Everything is sold at manufacturers' recommended retail prices – somewhat different from the US.

Alongside the equipment reviews, *Keyboard* also contains the full scores of a number of (usually Western) songs, complete with settings for the most popular synthesizers so that the sound of the original can be copied exactly.

Though dealers do not give an open discount, it is possible to get a discount comparable to Western figures by asking for "besto puraisu" (Japanese English for "best price"). In addition, the average Japanese dealer will provide more services than his Western counterpart. Usually at

least one practice studio will be available to potential customers, complete with equipment for rent. (Who needs detailed reviews when you can rent the latest equipment to use in your band?) There may also be a small demo studio, and in the larger music stores there may also be a music school, where you can learn your favorite Eddie Van Halen licks note for note and bend for bend. Singing (including English conversation) may also be taught. As a result, the music stores build up a very strong relationship with their customers, satisfying almost all of their musical needs.

But practicing in the music store rehearsal room may be the closest that most bands actually come to playing in public as the semi-professional music scene in Japan is almost non-existent. Most musicians are aged between 15 and 23 – the age at which Japanese attend high school and college. When leaving college, a Japanese entering a job will give up all popular music – after all, it's kids' stuff, and beneath the dignity of a "sararyman" (Japanese English for "office worker"). The students who use the instruments may well have received them as graduation presents or the like. The new toys may only be played in public once or twice a year, at friends' parties, or at the annual school dance.

In Tokyo, however, there is a wide boulevard that is closed to traffic on Sundays, and hordes of formation disco-dancers, "punks" and rock bands set up along the roadway and do their thing (using generator-supplied electricity). Rock music in Japan is not considered a social protest, it's "just a phase kids go through." This social attitude, combined with the worship of the West and the stifling of individuality, makes for a strange mixture. How else can you explain an immaculately turned-out Japanese punk, in designer-ripped "Anarchy in the UK" T-shirt, listening to a CD of the Sex Pistols' 'God Save the Queen'?

As a result of all these factors, most of the Japanese groups that I saw performing were very talented, very stylized, superbly-equipped, and terribly unoriginal. I was told that a Japanese child's practical musical education would start at about four or five years of age, and that about 30% of high-school students would play guitar. The most striking example of this syndrome came when I attended some of Yamaha's Band Explosion (Yamaha-ese for "rock contest"), and saw a three-girl band called Tone Tone. None of them were more than 18, and they had on stage three DX7s, an Akai S900, QX1, RX5, Roland Juno, electronic percussion and so on. Was all this equipment theirs? Yes, it was. How long had they been playing together? About six months. Were they thinking of turning professional? Giggles . . . They played extremely competently for 15 minutes on stage, had a fair degree of stage presence and were totally boring – a good copy of Western pop.

I'm not advocating that Japanese musicians rediscover original Japanese music, but surely there is room for some experimentation. The few Japanese (Sakamoto, Tomita, Yamashita) who are known in the US are exceptions to the rule, as they have broken away from the stifling mentality of the majority of Japanese musicians. The talent's there, and if two countries as similar as the US and England can produce different styles, there must be some hope for Japan.

"Tone Tone had three DX7s, an Akai S900, QX1, RX5, Roland Juno, electronic percussion, played for 15 minutes, had a fair degree of stage presence and were totally boring."

Bamboo Music

SO WHERE DOES this leave the Japanese instrument manufacturers on the home market? There is no essential difference in the domestic and export product ranges, only computer programs are sold extensively in Japan and not in the West. They command a domestic market in this area for two reasons – Japanese language, and the problems of porting software to a Western language, and also, Japan is blessed with a universal computer, the NEC PC9800 series, which is used for everything from games to business. Only one set of software needs to be written for the Japanese market, while in the West you may need to cater for the Commodore 64, Amiga, Atari ST, Macintosh and IBM PC. I have a feeling that any Japanese software which comes over will be ported to the IBM PC clone marketplace. Though not as good a machine as others around, it may well prove a good standard workhorse for the next few years at least.

All preset noises on instruments are the same as in the US – in fact, most manufacturers seem to have these sounds developed for them in the US as well as in Japan. Samples may be developed either in Japan or in other parts of the world – one room in Roland's R&D labs contained a technician, a Steinway, a sampler, and a lot of computer equipment – I didn't ask if they provided a Stradivarius and twenty years' violin tuition for the technician doing the violin samples. There's not much difference in the choices of sampled sounds between Japan and the rest of the world, with one or two manufacturers providing traditional Japanese instrument samples – but then they also provide Indian instrument samples, so perhaps we can't read too much into that.

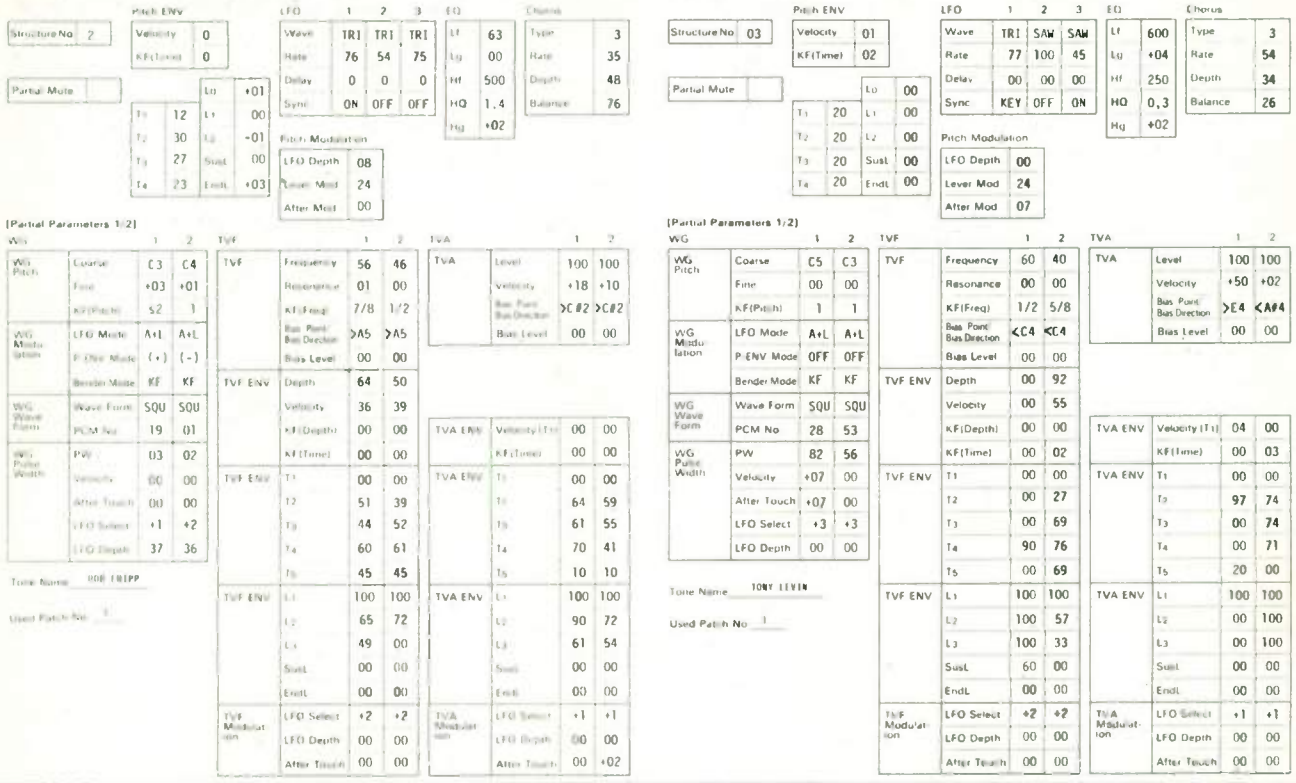
One thing that did strike me about most of the companies in Japan, which ultimately has a bearing on the final quality of sound, is that they all used the same monitor speakers. Without naming the speakers in question, I can tell you they have all the subtlety of an elephant tap-dancing on a tin roof outside your window at three in the morning. Any monitoring done with them is bound to affect the brain after a while, and may account for some of the names produced as presets. Of course, there are those who like them, and if any of you are out there, all I can say is that loudspeakers are a very personal preference – I'll continue to listen to music.

MIDI is important in Japan, but in a different way than it is in the US. Obviously there are advantages to a standard buss system, but the more esoteric functions of MIDI are held to be for specialists and professionals. Oddly enough, however, ▶

PROGRAMMING

ROLAND D50
King Crimson Court
Doug Fietsch, Hawthorne, CA

Though we're not quite sure about the name (it doesn't really sound like a 4-op patch), this analog-type sound is really unique. It has a built-in delay, created with the LFO's sawtooth down waveform associated with the AM sensitivity levels on operators 4 and 5, and a reverb-like decay, created with a number of different values for the envelopes' rate 4's. One point worth mentioning is that the patch is out of tune, but Richard suggests retuning the DX instead of changing the frequencies of the six operators because of the difficulty involved in finding the same frequency ratios. ■



Patch No. 1
Patch Name KING CRIMSON COURT

Key Mode	SPLIT
Sprit Point	D#4
Tone Balance	50
L-Key Shift	0
U-Key Shift	0
L-Fine Tune	0
U-Fine Tune	0
Bender Range	04
After(Pitch Bender)	04
Portamento Time	0
Portamento Mode	UL
Hold Mode	L
Output Mode	03
Reverb Type	01
Reverb Balance	43
Total Volume	100
Chase Mode	ULU
Chase Level	44
Chase Time	45

Used Tone
Upper ROBERT FRIPP
Lower TONY LEVIN

"DREAMDUST," a patch published in the October '87 issue, had an error in the values for DCA 2 Step 3. The correct values are Rate=50, Level=1 (not Rate=40, Level=1). Apologies to the creator, David Hare, and to any readers who used it. ■

PATCH WARE

IF YOU'RE LOOKING for sounds, check out the latest offerings from these companies!

A new cartridge, the **Prism Voice Cart** from Hitech Innovations, has been released for the ESQ1 with either 80 or 160 voices. The patches on this cartridge include a "User Switch," which allows the user to make adjustments to the patch through preset modifications (eg. modulators may be changed from "Off" to "LFO1" or "ENV2"). Each User Switch is documented and comments are offered on the various changes which can be made. The introductory price is $49.95. If you'd like more information, contact *Hitech Innovations, Inc., Sound Division, PO Box 16-1289, Sacramento, CA 95816.*

Microtel is offering a memory storage device which can be used on the Yamaha DX7, DX5 and DX1 keyboards and on the RX11 drum machine. Sold under the name 4 **Bank DX RAM Cartridge**, it stores 128 voices in a steel case designed with a beveled card edge to prevent damage to the instrument. Also included are 32 voices with each cartridge. For more information, contact *Microtel, Inc, 6864 W. 153rd Street, Overland Park, KS 66223. Tel: (913) 681-8700.*

A ROM cartridge for the DX7/IID/IIFD/S, which contains 64 original voices called **The Best of USA** is available from Key Clique. For information, contact your local music dealer or *Key Clique, Inc, 3960 Laurel Canyon Blvd, Suite #374, Studio City, CA 91604.*

Commodore 64/128 and Atari ST users may be interested in the **Sonic Horizon Sound Programs** available from MIDImouse Music. Each volume contains from 20-96 sounds, and several volumes are available for some synths. Sound Collections are available for the Casio CZ101, 1000, 3000, 5000, and CZ1; the Ensoniq ESQ1; Yamaha FB01, TX81Z, DX100, DX27, DX21, DX7, TX7, TX816; Roland Alpha Juno 1 & 2; Oberheim Matrix 6/6R and 12/Xpander; Korg Poly 800; Sequential SixTrak; Roland TR505 and TR707 . . . and more! For more info, contact *MIDImouse Music, Box 272, Rhododendron, OR 97049. Tel: (503) 622-5451.*

K-MUSE has expanded their library to include sounds for the E-Mu Emulator II, the Emax and SP12, and the Sequential Prophet 2000/2002, and have added additional disks for the Ensoniq Mirage Library. Each set contains six disks of an individual instrument group (eg. Dyno Drums, Killer Comps, Select Strings) and sell for $99 per set. Also available are disks for the Oberheim DPX1 (including The Burmer Collection created by Richard Burmer). For further information, contact *K-MUSE, Inc, SCS Division, 13341 Saticoy, N. Hollywood, CA 91605. Tel: (818) 764-7555.* ■

PROGRAMMING

CREATIN

Contriving curious recording environments and engineering strange treatments of sound are techniques that have made Daniel Lanois' productions of U2, Peter Gabriel and Brian Eno best-sellers. *Interview by Paul Tingen.*

STUDIO

ACCUSING A PRODUCER of making a mess would not ordinarily be a good way to begin an interview. But in the case of Canadian Daniel Lanois, it seems singularly appropriate. His basic working method comes down to creating as big a mess as possible, and then accepting the challenges and limitations this presents him and the artist. And there have been a number of *them* eager to experience this "art from choas" approach, ranging from Brian Eno and Jon Hassell to Peter Gabriel and U2.

Lanois doesn't seem to object to this assessment of his work; in fact, he takes it as a compliment.

"It's not far from the truth," he concedes. "What we're talking about here is creating spontaneity and performance in whatever way one can. At the end of the day, a good performance will override any production idea or any sonic idea you can have. If you've got a vocal that's strong and transports you as a listener, you're not going to worry what kind of EQ you've got on this drum or what sound you've got from that guitar. The delivery will override whatever small changes you can make in a sound.

"I find that the place where you're recording can also make a huge difference. Unusual locations have much to do with creating a recording environment. Certainly that's how I've worked with U2. We simply bring portable equipment into odd locations. Just the excitement of being in a new place and doing something slightly unorthodox helps to free your imagination. *The Unforgettable Fire* (U2's last LP) was recorded in Slane Castle in Dublin. The ballroom and the library had so much life and character, you walked in and hit a drum and said: 'My God, is this what we're going to get?' It's really inspiring."

In a similar vein, the latest U2 album, *The Joshua Tree*, which was produced with Brian Eno, was recorded at three different locations: the homes of guitarist The Edge and bass player Adam Clayton, and Ireland's Windmill Lane studio. Peter Gabriel's *So*, in which Lanois shared the production with Gabriel himself, was recorded entirely in the studio control room and in reverse order: guitars and keyboards first, rhythm tracks last. Producers often talk about "bending the rules," but Lanois seems to take greater liberties than anybody else. And indeed, his productions have an excitement to them which seems to be becoming more and more rare in today's pop music – one of the reasons Lanois currently finds himself in such demand.

THE FACT IS that Daniel Lanois likes to record things differently. Whether it's an unusual location, an unusual setup or unusual equipment, it all comes down to creating an environment which has the highest creative potential for the artist.

"What have you got to lose? You may get a brilliant track, and if you don't then you can always go back to a more standard approach. But it certainly always supplies a few nice surprises in the way of performance.

"I'd like to encourage other people to try it. For people who work with low budgets, renting a house and bringing in some gear is a cheap way to do it. It's in studios that you spend all the money."

Perhaps the best way to understand Lanois' approach is to take a closer look at his recording procedures. Preparations for *The Joshua Tree* began in May of last year . . .

"We had this pile of gear that we moved around with us," the producer explains, "essentially a whole studio. We'd bring the stuff into the house and build the whole thing up, with plexiglass doors and things. One of the main features was that we set up the band as if they were playing in a rehearsal environment or even a live setting.

"I think it works very well to do it that way. It pushes the band to work out their parts and get them working as in a live performance. You're not living in a land of promise, hoping that a track will come together with overdubs and manipulation."

Such a setup would be a nightmare for many producers – those that want as much control as possible over the different sections of the track, and hence prefer as much separation as possible between individual tracks. Lanois, however, doesn't share this philosophy, advocating performance as a more valid consideration.

"If a track is well performed you'll get a great mix out of it. It's as simple as that. Separation, or even the acoustics of a room, aren't that important. I've heard great recordings come out of little basement studios. In fact, one of the tracks on *The Joshua Tree*, 'In God's Country,' was recorded live in the basement of The Edge's house – which is not a particulary inspiring place. It's a kind of muggy little room where everything sounds dead. It worked because of the spontaneity and the lack of pressure at the time of recording. Now, you could say: 'You can't record down here. We need a proper room for the drums and we need this or that sort of mic.' But you could spend three days working out a foolproof plan and still not get a performance."

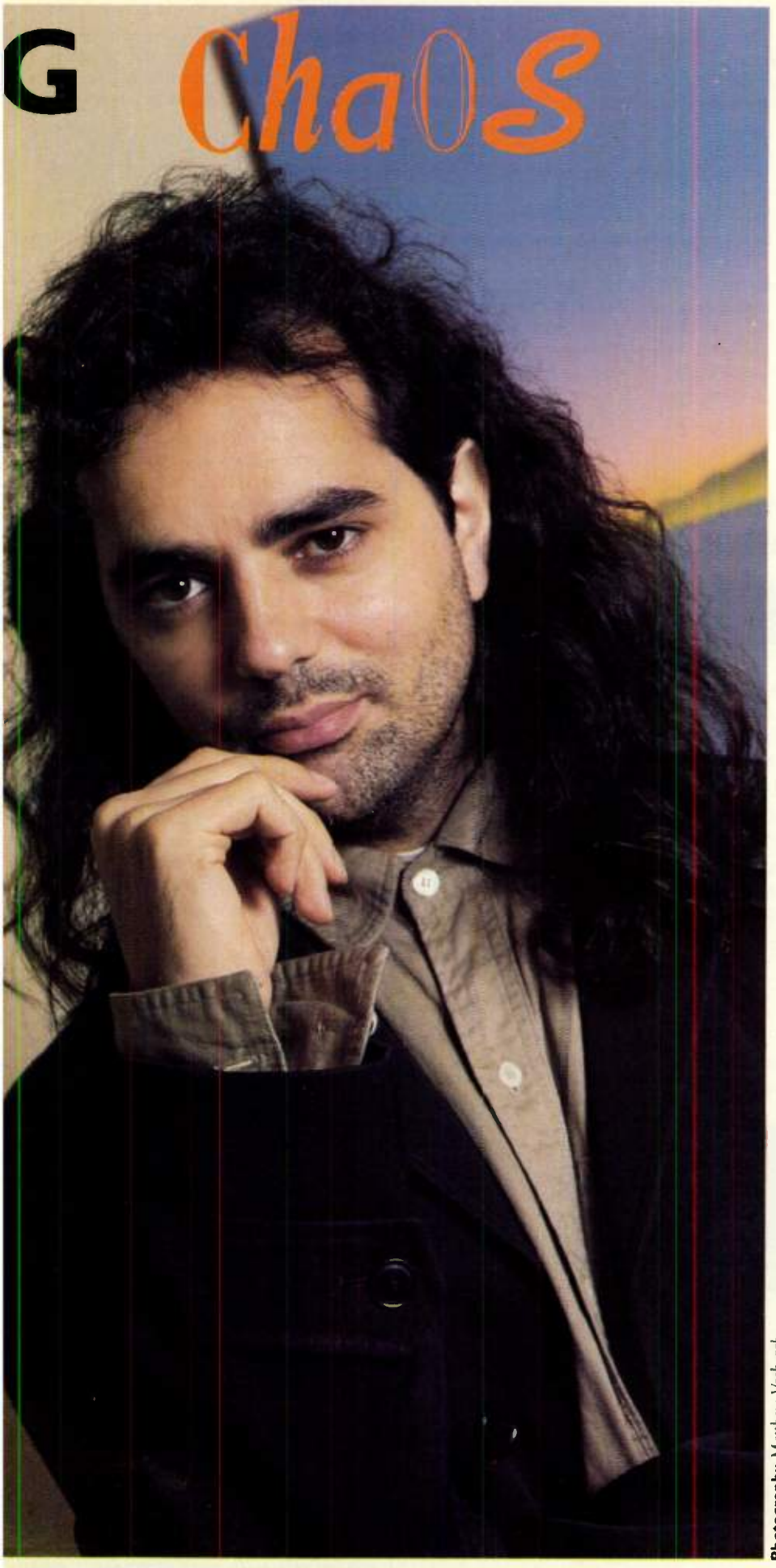
In the case of Gabriel's *So*, the album was sketched out between Gabriel, guitarist David Rhodes and Lanois: "We started off with vocal sketches, guitars and a rhythm box, putting it onto 24-track, until we had a good representation of the song. We had a lot of fun doing it: we called ourselves the three stooges and we'd wear hard hats, as if we were showing up for work on a building site. If times got tough, you'd look at your mate wearing a yellow hat and everything would be alright."

Entertaining games aside, even Lanois had his doubts about the recording of *So*. "The three of us would keep working on a song until the mood was completely right, and you couldn't kill it. Then we got the rhythm section in and they would overdub on what we had, using the rhythm box as a time reference, although most of the box was eliminated in the end. It's not a technique that I thought would be a good idea years ago, but it worked for that record, and I've heard other people do good work with it."

"A good performance will override any production idea you can have. If you've got a vocal that's strong and transports you as a listener, you're not going to worry what kind of EQ you've got on this drum or that guitar."

Another unusual technique used was to record everybody in the control room – including the drums. The producer presents his case:

"It's not very practical, but it is a lot more fun. The communication is a lot better and also I think that it's a great concept for musicians to hear themselves over the main monitoring system. I do vocals almost always in the control room. I can adjust the mix at the same time and get a direct response from him or her."



Photography *Matthew Vosburgh*

Lanois decided to switch from a Studer A80 24-track to a Mitsubishi 32-track halfway during the sessions. This was due to lack of tracks and Lanois' anxiety to avoid using synchronizers. It wasn't without problems of a different kind though, as he explains.

"I felt like taking all the A80 information and dumping it onto the 32-track, doing some pre-mixes and getting on with our lives. But when we transferred things to the Mitsubishi, there were some huge surprises – like the bass almost disappearing entirely. So we just dumped some guide tracks and used it for vocal overdubs, for which it is excellent. In the end we did a synchronized mix of the A80 and Mitsubishi material."

With this we hit upon Lanois' preference for old equipment over new.

"I appreciate the simplicity of old equipment. It's reliable, I feel I can trust it."

Putting his money where his mouth is, the producer is currently putting together a studio with brother, Bob. For this they are buying up 'vintage' equipment. Lanois elaborates.

"We're just accumulating gear at the moment, which is a lot of fun. We got hold of some old valve mics, like The Neuman U47 and the Sony C500. Of course we will be using the Studer A80 as our multitrack machine. It looks amazing too, so it's fun to be in the same room as that tape recorder. We're also getting an API console, which is still one of the punchiest sounding consoles ever, especially for bass."

But surely, a nice looking tape recorder and a good bass performance from a mixing console are hardly reason enough to start buying up as much obsolete equipment as possible? Lanois admits that there is more to it than that, although he starts off with one of his favorite topics: the quality of bass sounds.

"I'm very interested in developing and rediscovering the kind of pulse and drive that used to be present in the bottom of old recordings. That must have had a lot to do with the gear which was around at the time. I don't notice that much difference in quality between records now and the good records that were made even in the '50s. It's not as if we are wildly advanced now and it was archaic then.

"Some of the vocal sounds from the '50s are the best vocal sounds ever. When you listen to those Elvis Presley recordings, the vocals are right in your face. They're warm and they're powerful and you think, 'They did this 30 years ago, what are we doing inventing new microphones?'"

IT WAS LANOIS' individuality that brought Brian Eno to him in the early '80s. Eno, tired of the cost and lack of character of New York studios, was impressed by some demos he'd heard recorded in Grant Avenue studio in Ontario. ►

► He thought he'd give the studio a try, and found Lanois and his brother Bob behind the faders.

It was his connection with Eno that was to bring Daniel Lanois into the international limelight, but he'd already made a name for himself in Canada. Born in 1951 of French-Canadian parents, the young Lanois set out to make a career in music. By his mid-teens he was playing guitar professionally in a variety of rhythm 'n' blues bands and dance bands, touring Canada, and occasionally backing strippers.

STUDIO

Lanois studied a little music, took some individual guitar lessons, became proficient on the pedal steel guitar and worked with a lot of country and western artists. In 1970, he started a studio in the basement of the family house with his brother, pulling a lot of work from various roots music – C&W, gospel, R&B – and also from advertising jingles for the local radio station. In 1980, the studio expanded to 24-track, which was the beginning of Grant Avenue studios.

"You could say: 'You can't record here, we need a proper room for the drums and we need this mic'. You could spend three days working out a foolproof plan and still not get a performance."

The first major project Lanois and Eno shared was the latter's collaboration with Harold Budd on *Plateaux of Mirror*. Eno's *On Land* album and the *Apollo* project (on which Lanois also was credited as co-composer) followed. Lanois went on to produce *Dream Theory In Malaya*, by Jon Hassell, and *Voices* by Roger Eno. In 1984, Brian Eno invited Lanois to co-produce U2's *The Unforgettable Fire*, the success of which led to their production of *The Joshua Tree* two years later and to Lanois making his name internationally as a producer. How does the partnership work?

"Brian tends to work with a greater overview of the project and lets me get on with recording vocals and so on. He doesn't have a lot of patience for homing in on sounds, moving things around, taking a drum performance at one end of a recording and moving it to the other. He likes to be presented with a tape that's in good order, and then he will get on with his treatments. He tends to look at the big picture while I get on with the chores."

Lanois regards his "feeling" for music as the most important element in his working relationships.

"I think that the attraction with artists like Gabriel and U2 is that they're thinkers as well as players," explains the producer. "In my experience, intellectual people need soulful people around them to bring out the performance in what they do. Although my intellect is alive, I operate more from my instinct. I can recognize a good performance, it's one of the things that I do well under pressure. I know when there are great musical moments going by and I can help people to capture them and get them onto tape. It's a good marriage."

Another of Lanois' specialties – and one he has in common with Eno – is treatment of sound. He has used these techniques most extensively on the ambient albums of Brian and Roger Eno and on Gabriel's *Birdy*.

"It's almost another side of what I do. I keep treatments available on the console all the time. There's like a bank of 12 or 16 channels that are designated to treatments, and that's all they do. At any given point I can send an instrument or a vocal to these treatments and get a quick impression of what is working and what isn't. This often takes plaoe in the absence of the band, once there's something on the tape to work with. It's like fun time – try this, try that, modifying what's already there. It's like tinting an existing photograph or having a photograph and increasing the contrast. In a lot of cases I have given new life to a track with these treatments."

One such track is 'In God's Country' off *The Joshua Tree*.

"The guitar now has a beautiful shimmer which has a lot to do with the mood of the track. What was a fairly straightforward rock track is now undermined by a mood of unrest; not all is well. It supplied Bono with new inspiration. It gave him a clue to

modify his lyrics and give the track a greater dimension."

Although not overtly recognizable, there are a lot of these treatments on *The Joshua Tree*, as Lanois reveals.

"It seems straightforward but there's a lot of subtle manipulation there. It's not like a boring documentation of a dry room. You want to give more dimensions to a record so that it has a lasting appeal.

"At Windmill Lane they have a big warehouse, so we put a PA there, put the drums through it and re-miked them to get some punch. There's a staggering difference, just piping the instruments to the back of this room added a whole other dimension. Almost like the difference between Dling a synth and putting it through a Mesa Boogie guitar amp. It gave us a result which digital reverb couldn't give us."

MOVING ON TO synthesizers, Lanois alights upon another aspect of *The Joshua Tree*: Brian Eno's DX7 programming.

"Brian has some very good sounds in his machine. He spent about a year just working on sounds. On top of that, we put all the DX7 sounds through a Mesa Boogie, *including* the sequences."

And that typifies Lanois' unorthodox approach to recording: the DX7's crystal clarity demands it be DI'd for it to be properly reproduced, so he feeds it through an amplifier designed for the distorted excess of the rock guitar.

"I highly recommend people to at least try playing a synth through amps because you get these peculiar bits of personality. A 12" speaker is not a full range speaker, so ►

▶ you get a denser sound with a little bit more poke and sounding a little more organic. It's because certain frequencies sound louder than others. Also, putting a graphic equalizer between the synth and the amp and trying different mics and different rooms may give interesting results.

"We put the drum sequence on 'With or Without You' through the amp because it sounds more like people playing in a room, rather than a machine. There was no contest between DIing it and putting it through the amp."

The areas of sequencing and computer-based equipment is clearly not a topic to which Lanois warms easily, at least not while speaking about it in general terms.

"They're just part of a toolbox, aren't they? The sequencers, the samplers, the drum boxes. They should be used when they apply. It's a matter of serving the song. For example, if you're looking for a mood of discipline or speed, then a machine can offer you that much easier than live playing. That's why we used a sequencer on the beginning of 'With or Without You,' we wanted a feeling of discipline. And then when the drums kick In halfway, they mean something.

"I find that when musicians and composers are genuinely excited about their tools, good results will come of it. Kraftwerk is a good example of a group that uses machines well. There's a stiffness there, but that stiffness is part of the mood that they're trying to get across. Yet it still sounds organic at the end of the day. I don't know how they do it exactly. It probably has to do with the tastefulness of the operators and that they're not relying on those machines to do everything for them."

"I can recognize a good performance, it's one of the things that I do well. I know when there are great musical moments going by and I can help people to capture them and get them onto tape."

When asked about his favorite synths, Lanois' initial response comes as no surprise to connoisseurs of his work.

"The Yamaha CS80 is one of my old favorites. It was one of the first polyphonic synths and it's a fantastic thing. To this day it's got some of the most amazing sounds. I'm continuously impressed with it.

"We used it on So, although the Prophet 5 was the main synth there. Peter has one of the best sounding ones I've ever come across. I also like the little Korg Poly 800II. It's light, you can carry it on a plane and it has some nice sounds. For bass sounds I like the Fairlight. A sampled bass sound has a personality of its own, no matter what you do with it, it's organic and punchy."

Although he presently has no commitments ("I'm taking a break after six years of continuous work"), Lanois states his plans for the future as "carrying on making records that in 10 or 20 years will stand out as classics." A Lanois solo album also sounds like a possibility.

"I still write material, but when you're so busy working with other people, the tendency is to leave your own stuff until later. I have quite a few unfinished compositions on the backburner. It's mostly instrumental music, although it could become songs. An instrumental album would be easy for me to do now, but I need to do some soul searching and decide whether to incorporate vocals or not."

As the interview slowly winds down, one last question suddenly gets Lanois fired up again: Who is the person who has influenced him most?

"My mother. She helped me to get started on my studio when no-one else would. And a strong family bond has been the single most important support. I feel very grateful to have that in this complicated world.

"As a little kid I remember a lot of violin playing and tap-dancing going on during family gatherings. When you're young that makes a big impact. It can really drive you to love music and in my case I took it up and made it my life. During my mid-teens I decided: this is it, this is what I want to do."

Lucky man. ■

SELL IT FOR NOTHING

WITH A FREE CLASSIFIED AD IN MUSIC TECHNOLOGY

MUSIC TECHNOLOGY'S free classified section is designed to help private sellers find buyers and bands find other musicians. If you buy and sell musical equipment or sounds as part of a small business, please contact the Advertising Department to place a regular classified ad.

KEYBOARDS

CASIO CZ3000 8-voice polyphonic synth; multi-timbral, patch layering, keyboard splits, stereo chorus, dust cover, EC, $450 obo. Frank, Tel: (804) 440-7377.

KORG POLYSIX with MIDI retrofit, great synth with anvil road case, $595. Tel: (201) 334-3350.

OBERHEIM OB8 polysynth, split keyboard, MIDI multi-layered sounds, cassette library, excellent cond, $995. Tel: (201) 386-0758.

MOOG LIBERATION synth, $400 obo. Jeff Roper, Tel: (619) 375-8384.

MOOG MODEL D synth; this is the original, no modifications, perfect cond, $450. Dave, Tel: (703) 569-2006, leave message.

PROPHET 2000 with memory expansion, Sound Designer 2000 for Macintosh, Opcode MIDI Mac interface, discs, never roaded, $2000. Tel: (612) 379-0614.

RMI ELECTRIC PIANO, sustain and volume pedals, portable w/legs, excellent cond. Tel: (919) 489-8430.

SERGE MODULAR SYSTEM, quality patching synthesizer, $1800, all offers considered. Tel: (213) 457-8390.

VINTAGE PROPHET 5, classic analog sound, Rev 1, good cond, $400. Robert, Tel: (415) 854-5222.

YAMAHA CP35 ELECTRIC PIANO, 73 keys, cover, like-new stand, pedal included, excellent cond; was $1800, now $550. Tel: (314) 594-3706.

YAMAHA CP70B, w/road cases, excellent, $1700 obo; Moog Opus 3 w/cases, $300 obo; Prodigy w/cases, $250 obo; Tapco keyboard mixer, $75. Tel: (206) 641-9135.

YAMAHA DX7, $1275; Kawai K3, $775. Neither used on road; in their original shipping cartons. Tel: (314) 468-4447.

SEQUENCERS

YAMAHA QX21 SEQUENCER, mint cond, $275 obo. Jarrett, Tel: (415) 221-9761.

DRUMS

E-MU SP12, mint cond, with turbo and Commodore 1541 disk drive and sound library, $1750 obo. Tel: (617) 449-6488.

KORG DDD1, new cond, must sell, $500 obo; Korg MR16 MIDI Rhythm Slave, $75. Tel: (919) 726-4052.

KORG KPR77 programmable drum machine, mint cond, $125. Tel: (212) 365-6988.

OBERHEIM DMX, mint cond, $350. Tel: (518) 482-9156.

REMO ROTO-TOMS, 7 drums w/stands and mounting hardware, sizes: 6", 8", 10", 12", 14", 16", 18". All in excellent cond, $500 obo. Jarrett, Tel: (415) 221-9761.

ROLAND DDR30 MIDI drum brain, rack-mounted, 48 different drum sounds, $650. Tel: (213) 276-9286.

SAMPLING DRUM MACHINE, E-mu SP12 turbo w/disk drive and anvil case, immaculate cond, $2000. Dozens of samples available. Kurt, Tel: (818) 782-4522.

SCI DRUMTRAKS with custom chips, tunable MIDI, excellent cond, $200. Tel: (415) 948-7523.

YAMAHA RX15 DRUM machine with updated internal ROM, excellent cond, $300. David, Tel: (818) 700-0890.

COMPUTING

SONUS SUPER SEQUENCER for Commodore 128, like new, $135; Commodore 64/128 MIDI interface, $55. Tel: (616) 452-1616.

SYNTECH STUDIO 1 sequencer and hardware interface for Commodore 64, $135. Tel: (518) 482-9156.

RECORDING

AMPEX MM1000 16-TRACK tape machine, remote control, updated, easy punch-in feature from remote, 15/30 ips, excellent cond, $6500. Tel: (516) 589-2169.

MINT TEAC 80-8 with dbx, $1800. Ron, Tel: (619) 294-4022.

RAMSA WR8112 CONSOLE, 12×4×2, direct outs on all channels, stereo and mono effects sends and returns, asking $1300. Steve Bozeman, PO Box 97, San Antonio, FL 34266.

ROCKTRON NOISE REDUCTION, 8-channels encode/decode, like DBX1, almost new, very quiet, $500 obo. Chuck, Tel: (813) 868-7508 or 845-3040.

TASCAM MINISTUDIO PORTA-ONE, 4-track, excellent cond, only one owner, great start for home recording, $325. David, Tel: (818) 700-0890.

TASCAM 246, hardly used, makes great demos, six channels in, many extras, $800. Tel: (317) 482-7601 or 482-7104.

PERSONNEL

EXPERIENCED KEYBOARD TECH, roadie, soundman looking for professional touring band. Rick Huber, Tel: (818) 997-8691.

KEYBOARDIST / ARRANGER / PRODUCER, top equipment, MIDI studio, looking to tour/record with pro signed project only. Ron Waters, Tel: (818) 344-2713.

PRODUCER (ALBUM CREDITS) with 8-track digital MIDI studio seeks projects for recording, free consultation. Tel: (201) 334-3350.

PROFESSIONAL VOCALIST (album credits) available for studio work (jingles, pop, R&B, etc). Terry, Tel: (201) 334-3350.

VOCALIST / multi-instrumentalist / samplist, seeks percussionist with hi-tech sample gear to complete project. Influences Tones on Tail, Japan. Contact Christopher immediately, Tel: (415) 474-8897.

MISC

ALTECH POWER SPEAKER, built-in 100W amp, $250. Bob, Tel: (415) 854-5222.

BYTER POLYSYNC UNIT syncs all standard synths, 12/24, 48/96 pulse, only $150. Tel: (201) 386-0758.

ELECTRO-VOICE S18-3 SPEAKER cabinet, $600; Roland CPM120 8-channel mixer with 120W amp, $600. All EC. Tel: (919) 489-8430.

MC500 OWNERS: 600+ cover songs for exchange, oldies, R&B, dance. Parker, Tel: (213) 468-6080.

PROPHET 5 REV 3.2, Pro-One, Akai AX80, Roland TR707 drum machine, other items. Tel: (512) 899-4888, evenings.

PROPHET 2000 and Mirage owners, let's swap samples. John Villec, 800 D Street #7, Arcata, CA 95521. Tel: (707) 826-7506.

SEQUENTIAL SIXTRAK OWNERS, looking to trade patches. Greg Kyryluk, 6526 SW 20 Street, Miami, FL 33155. Tel: (305) 261-2558.

T&D AUDIO CUSTOM keyboard cabinet, 4-way, pro specs and components with aluminum road case, asking $1600. Tel: (201) 371-0403.

WANTED

CZ LIBRARIAN PROGRAM for Yamaha CX5M computer. If you have or can write such a program, Tel: (312) 452-1684.

SAMPLES (FILES) in Macintosh Digidesign Sound Designer format from any sampler, purchase or trade. Clay, Tel: (502) 896-6581, collect.

USED CASIO CZ101; also looking for used Simmons drums. Paris Toon, Tel: (617) 784-8658, leave message.

ROLAND TR707 DRUM machine, Roland MTR100 digital cassette recorder. Both must be in excellent cond. Tel: (919) 489-8430. ■

MT FREE CLASSIFIED ORDER FORM

FILL IN THE FORM to a maximum of 18 words (one in each box), and send it to: **MUSIC TECHNOLOGY, 7361 Topanga Canyon Blvd., Canoga Park, CA 91303.** Please print clearly in BLOCK CAPITALS. Your ad will be printed in the next available issue.

Name .....

Address .....

.....

City .....

State .....

Zip .....

Please include this ad in the ..... *section.*

CLASSIFIED

Advertisers Index

THE RIGHT STUFF

Kawai's R-50 has the right sounds and the right features. Check it out:

POLYPHONY

Polyphonic note assignment lets a single instrument sound more than one note simultaneously. With polyphony, a crash cymbal can continue to ring even after the same cymbal is struck again–even at a different pitch. Or build a kit with 10 different sized toms or a scale of tuned Agogo Bells.

PITCH AND PAN FOR EACH NOTE

Every note in a pattern is stored with its own tuning and pan for ultimate control over your sounds.

ON-BOARD EFFECTS

Flange, Delay, and Gate effects are built in and also programmable for every note.

PAD PROGRAMMABILITY

12 Drum Pad programs allow assignment of any sound to any pad, each with its own settings for volume, accent, pitch, pan and effects. Each program can then be recalled with the touch of a button or footswitch.

MEGA-MIDI

The velocity-sensitive R-50 is the perfect slave for MIDI keyboards and sequencers. Each MIDI note can be assigned to any drum, with unique pitch, pan, and effects settings. MIDI Song Pointer enables synchronization with SMPTE controllers. And eight extra note triggers enable the R-50 to play external MIDI devices as well.

KAWAI ALTERNATE SOUND CHIPS KEEP YOUR OPTIONS OPEN

From ten snare drums, to bass guitars, orchestra hits and brushes, the wide range of sounds available keep the R-50 at the top of its class.

TAPE INTERFACE & MIDI DATA DUMP

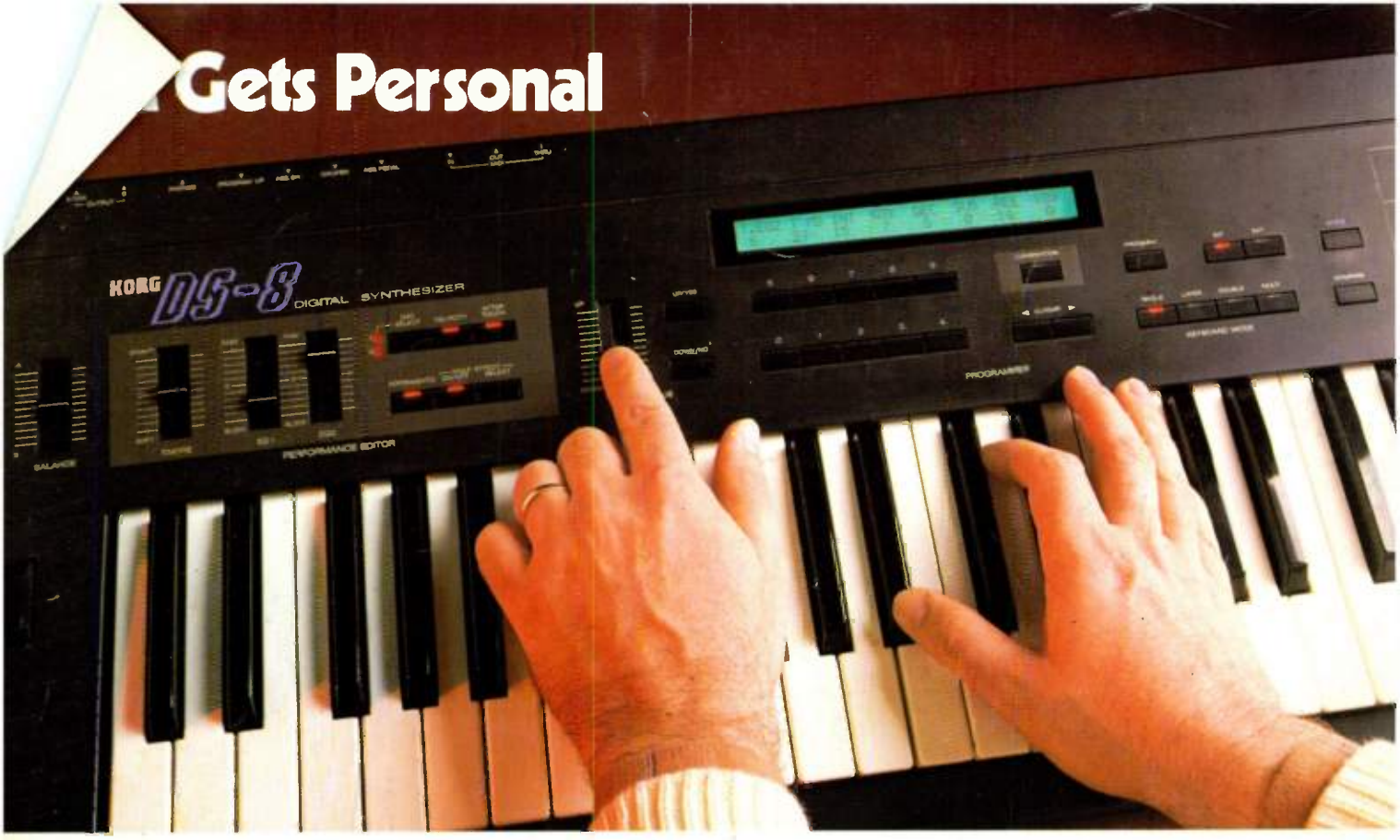
Store patterns on computer or use the inexpensive tape interface.

Kawai's small R-50 with the big sound. Before you decide on a drum machine, give the R-50 a good listen at your authorized Kawai EMI Division dealer.

KAWAI

KAWAI AMERICA CORPORATION
EMI DIVISION
2055 East University Drive • P.O. Box 9045
Compton, California 90224-9045

Gets Personal



Used to be, anyone could get complex, wide-ranging FM digital sounds — if you didn't mind sounding exactly like everyone else. Excessive complexity made FM synthesis the private property of specialist programmers.

Now Korg engineers have simplified FM. The new DS-8 has eight two-oscillator FM digital voices — and a full set of fast, intuitively familiar analog-style controls. No more searching through endless program cartridges, no more settling for "close enough." Get the sound right without getting tangled up in operators, algorithms and techno-secrets.

The DS-8 has responsive real-time performance editing to bring out the individuality of your playing. Combination programs that give you instant access to splits and layers. And full multichannel, multitimbral operation to give you a virtual multitrack studio. Assign any number of the DS-8's eight voices to different programs and MIDI channels. Sequence your bass line, piano comp and horn or string lines, all playing at once.

Take control of the DS-8 at your Korg dealer. You'll find that it gives FM synthesis a whole new personality — yours.

DS-8 Features

Versatile, cost-effective design. Eight voices with two FM oscillators each. Programmable multi-effect digital delay. New multi-stage Timbre and Amplitude envelope generators. Four waveform options plus cross modulation. 100 program memory (200/400 with optional RAM/ROM cards). Full MIDI capabilities with programmable multi-timbral voice assignments. Splits, layers and multi-patch combinations including MIDI channel, digital effect and output panning.

Weighted action, 61 velocity- and pressure-sensitive keys. Nine function real time Performance Editor section: EG and timbre controls for each oscillator; Mix/Balance for layers or splits; Oscillator select; Velocity & Aftertouch on/off; Effects select. Two assignable footswitch/pedal jacks. Programmable joystick.

For a free catalog of Korg products, send your name and address, plus $1.00 for postage and handling to: Korg USA, 89 Frost St., Westbury, NY 11590, or to Korg USA West, 7886 Deering Ave., Canoga Park, CA 91304
Exclusively distributed in Canada by: Erikson Music, 378 Isabey Street, St. Laurent, Quebec H4T 1W1

KORG®