MUSIC
TECHNOLOGY

RUSH
Tough Talk

BRIAN ENO
Techno-Philosopher

PETER ERSKINE
A Step Ahead

COMPUTER NOTES
New Computer Music Section

SAMPLER QUALITY
Every Little Bit Counts

ON STAGE
Pink Floyd

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COMPUTER NOTES

THE INFLUENCE THAT computers have had on the music-making process over the last few years is phenomenal. Composition, performance, arranging, transcription, and sound creation and manipulation have all somehow been affected by them, and we can expect even more changes to occur in the none-too-distant future. Computers are showing up in more and more professional studios, in more and more bedroom studios and even in more and more live shows (and not just big name acts, either; club bands and "weekend warriors" are also starting to use computers live).

Along with their growth in importance has come a growth in the need for information about computers and music-related applications. Computer literacy has nearly come to be expected for musicians. But even if the desire is there to learn as much about computers and music software as one can, the actual information often is not. Well, magazines like MT are in the information business, hence we see it as our job to inform you of the latest developments in areas of interest related to our main topic: music and technology. Computer Notes, the new computer music section we're introducing with this issue, was conceived with exactly these concerns in mind.

Computer Notes will consist of reviews of music software, articles on computer music applications, general information computer articles, news on computer developments which could have an effect on music applications and other topics related to the connection between computers and music. Take a look at what's in there this month and I think you'll get an idea of what we're trying to do. In addition to our regular software reviews (to which have been added the smaller Micro Reviews), there's computer news, a general article on hard disks, and a report on a conference that looked at possible applications of artificial intelligence to music.

If you have any ideas or topics you'd like to see covered, let us know; we're very interested in putting in what you want to see. Also, though we won't devote an entire issue to a single computer, we will occasionally do computer-specific articles that can help you get more out of the computer you work with or own. Computers aren't always the friendliest beasts, but once you learn to tame yours, you should be able to coax it into helping you in some way or other with your creative process. If you haven't made the plunge into computers yet – you're certainly not alone – then the information may help you decide if a computer is something you're going to want to investigate.

We can't provide all the information you'll ever need to know about computers, nor do we intend to become a computer magazine. We do think, though, that information about computers not only fits into a music magazine, but actually should be there. And it should be there monthly, which is why Computer Notes will be a regular section within these pages.

To give the new section the appropriate hi-tech look, we've enlisted the help of computer graphics artist Colin Cantwell. In addition to the work he's done on demonstrations for this rapidly growing field, Cantwell's other resume credits include working on the design of many of the space ships in Star Wars. He's a MIDI freak as well, and we're extremely happy to have his contributions.

The music-making process has been and will continue to be greatly influenced by computers and the creative and workhorse power they can provide. Imaginative applications of computers and musical instruments have begun to produce exciting new kinds of music that could not have been produced prior to their use. And that's what the marriage of music and technology is all about. ■

Bob O'Donnell

NO PATIENCE REQUIRED

Patience is not necessarily a desirable trait for a musician. When musical ideas are running through your head you need equipment that won't slow down the creative process. Make you wait. While you risk losing a great idea. Or the feel. Or the moment.

Instead your equipment must perform, document, and help you produce results. As fast as you can work. As fast as you can create.

Alesis didn't invent the drum machine and MIDI sequencer/recorder. We *reinvented* them. We think a drum machine should sound *exactly* like real drums. And a MIDI sequencer should be a powerful, flexible computer, yet work as simply as a multi track recorder.

With the **Alesis HR-16 High Sample Rate 16 Bit Digital Drum Machine** and the **MMT-8 Multi Track MIDI Recorder** you can work like a musician. Think like a musician. Create like a musician. And sound like a great engineer. They'll let you exercise your musical muscle, instead of your patience.

The drum machine and sequencer have arrived. The real thing. At last.

ALESIS
STUDIO ELECTRONICS

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WE LOVE FREEBIES

Next time you're near a Dynacord dealer, stop in and ask for your free color poster – which is even more striking at its original size! The picture was originally published in

a German audio/video journal and snatched up by Dynacord for reproduction and distribution.

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

This cool poster can be yours, free.

UNRAVELLING THE 2290

Owners of TC Electronic TC2290 sampler/multi-effects units may be interested in the new video owner's manual which promises to open the doors to all the hidden secrets of the complex

unit. The 50-minute video utilizes a timecode, allowing the user to quickly find the appropriate information as specified in the table of contents.

MORE FROM TC Electronic, 120 County Road, Tenafly, NJ 07670. Tel: (201) 384-4221

BE A STAR

Mede Publications has announced availability of a home study course which covers the ins and outs of the record industry. The course includes a 162-page book and ten 60-minute audio cassettes in a vinyl hardcover binder, along with six additional audio cassettes featuring interviews with industry leaders.

The course starts with discussions of the development of the record industry, looks at each field of the business and reveals employment possibilities in each area. The interviews are with record company execs, studio musicians, record producers, studio managers and engineers.

Price of the course is $179.99 plus shipping and handling.

MORE FROM Mede Publications Inc, 114-41 Queens Blvd, Suite 304-P, Forest Hills, NY 11375. Tel: (718) 217-1598

AAAHHH . . . A BABY

Roland is promoting its new PR100 sequencer as ideal for educational use.

Roland has introduced a new sequencer to their line, the PR100, featuring an internal memory of approximately 17,000 notes, two tracks, and a built-in disk drive for 2.8" Quick Disks. The ever-present alpha dial and an eight-character LCD display are touted as features which make the unit easy to use.

Editing and overdubbing capabilities are provided, allowing mixing and merging, block repeat for looping playback, punch in/out for error correction, and a built-in metronome to assist in rhythmic accuracy.

The suggested retail price of the PR100 is $695.

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

EIGHT TRACK AUTOMATION

JL Cooper Electronics has announced the development of Mix Mate, a low-cost automation package for the eight-track studio. Included in the 7" by 10" package is everything you need to automate your mixing including a SMPTE or FSK reader/striper; dbx VCAs; fader and mute controls; automation memory and control smarts.

Several synchronization choices may be

Magi and Sam have found a Mate at JL Cooper.

utilized, including SMPTE, MIDI Time Code, MIDI Clocks and Song Position Pointers, and FSK. Mix Mate may also be set to act as a slave device to software packages such as Digidesign's Q-Sheet. The optional Mix MatePlus allows an Atari ST computer to provide graphics display, expanded memory and disk storage.

The suggested retail price is $995; the plus option is an additional $245.

MORE FROM JL Cooper Electronics, 1931 Pontius Avenue, West Los Angeles, CA 90025. Tel: (213) 473-8771

MIDI YOUR RHODES

The Electric Piano MIDI System Kit could bring your old keyboard new life.

So you're looking for a MIDI Controller, but you don't want to get rid of your old favorite, huh? Maybe you can have both! Music Dealer Service (MDS) has announced the release of a series of Electric Piano MIDI System Kits designed to upgrade electric pianos which do not have a MIDI interface through the installation of a hardware mod.

The kits offer full velocity-sensitive MIDI control, keyboard controller software and sustain pedal implementation. Each has a dual multi-function footpedal controlling two independent active MIDI channels, with control over the key range of each channel, split point and transpose offset. In addition, the PFI0/15 kit allows for MIDI In and Thru.

Kits are currently available for the Yamaha CP70/70B/80, PFI0/15, Kawai EP308 and Fender Rhodes 73/88 pianos. The suggested price for a 73-key mod is $899; 88-key versions are $950.

MORE FROM MDS/Music Dealer Service, 4700 W. Fullerton, Chicago, IL 60639. Tel: (312) 282-8171

NEED A MENTOR?

The Mentor MIDI Network Controller, first introduced at the summer NAMM convention, is now available from Forte Music. The system serves as a link between performance devices, sound modules and accessories, providing two MIDI inputs and four MIDI outputs, plus auxiliary MIDI control input. Each MIDI slave device may be configured independently for program, volume, key range, transposition, delay and other parameters.

The system is composed of two units: a rack-mount processor (single rack space) and a small remote programmer/controller. In addition to the four programmable slide controls located on the remote, inputs are provided for footswitches, voltage/volume pedals and other controllers. Programmable filtering of all MIDI message types occurs simultaneously, and key inversion, velocity switching and key mapping are possible. A memory cartridge interface has also been added for storage of preset data.

The suggested retail price for the system is $1450.

MORE FROM Forte Music Inc, PO Box 6322, San Jose, CA 95150. Tel: (415) 965-8880

The Mentor system is designed as a central controller/programmer for any MIDI setup.

HARMONIOUS PROCESSING

The H3000 Ultra-Harmonizer Pitch Change and Effects Processor (whew!) from Eventide includes capabilities that go far beyond any previous Eventide Harmonizer model. The unit offers stereo pitch change, which includes proprietary stereo/mono compatibility, phase compensated de-glitch circuitry, as well as "diatonic" pitch change. This is a new effect which creates harmonies by analyzing the note and generating harmony in real time based on the key in which the tone is being played. Additional signal processing capabilities include reverb and effect programs and several gated and multitap programs.

The H3000 is MIDI programmable, featuring MIDI program change and real-time MIDI control of all parameters. Specs promise full 16-bit resolution at a 44.1kHz sampling rate, frequency response of 5Hz-20kHz ±1dB, and signal-to-noise of greater than 92dB, "A" weighted. A 40-character by two-line back-lit display is provided for reviewing parameters, and user-adjustable parameters are available via four soft keys. The famous Eventide Harmonizer "knob" is present, but is joined by a direct-entry keypad plus up/down buttons.

All this for $2395.

MORE FROM Eventide Inc, One Alsan Way, Little Ferry, NJ 07643. Tel: (201) 641-1200
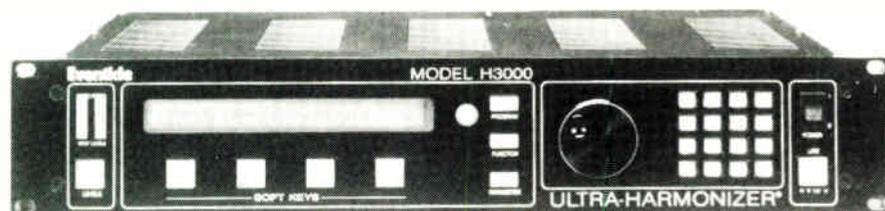
Eventide's H3000 Ultra-Harmonizer combines pitch change and signal processing with a complete MIDI implementation.

I WANT MY MTC!

MIDIMIX has announced availability of a new software update, 902.1, which allows MIDI Time Code to safely pass through MIDIMIX 8 and 9 merging systems. Special functions to eliminate All Notes Off or Data Slider info are available as 902.2 and 902.3. The new software is $30 plus $4 for shipping.

Also shipping now is the fourth generation of MIDIMIX 6 (a 1 in/5 out MIDI splitter), offering new connector spacing and more powerful circuitry. This unit is available for $35 plus $4 for shipping.

MORE FROM MIDIMIX, PO Box 161, Ashland, OR 97520. Tel: (503) 488-1023

LYNC YES, LYNX NO

MT would like to apologize to Lync Systems for misspelling their name in last month's interview with Tony Kaye of Yes. So no, the company is not spelt "Lynx" (which is a wildcat found throughout the N Hemisphere, having a short tail and tufted ears). Spelling and zoology class dismissed ... ∎

READERS' LETTERS

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

Dear Music Technology,

First of all, thanks for all the help you (and especially Bob O'Donnell) have been. I'm a sax player and electrical engineer and I own a MIDI recording studio. I'm also a bit persistent so I sometimes have trouble taking "no" as an answer. This led to my search for the ultimate generic librarian and MIDI development software for the IBM PC and compatibles.

Some of you out there may feel as I do – I often don't need anything fancy, just the ability to store banks of data/sound parameters/whatever. Why can't you build a MIDI string in a librarian program which permits you to force a data dump or load? The perfect example is my Korg DRV3000, which does not have any buttons on the front panel, and is theoretically capable of sending or receiving programs. I use it both for gigs (live settings), and recording studio work (custom setups).

According to Voyetra Technologies, whom I called for information, the reason why there have not been any programs which are truly able to up/download MIDI data from any device is that each device has its own idiosyncracies. I was told that it takes them as much as two days to make a device perform consistently. Club MIDI, however, is planning a true generic librarian in its newest version of PROLIB. It's due in a few months.

The MC500, by the way, is great for this purpose, because you can program system exclusive data into different tracks, experiment, and perform dumps with it. Some companies I talked to expressed reluctance to make their sequencer do this. "Marketing conflicts," they called it! Come on guys, give yourself a chance to capture some market share! Oh, well . . .

Phase II of The Search was for a MIDI programming assistant. I figured that I would write a (crude but effective) librarian program myself. Bob hooked me up with Altec, who says that their MIDI-Basic program should be ready soon, followed by MIDI-Pascal. Who says there's no Santa Claus?!

If I were a developer (hint, hint), I would write a "patch cord" program which prompted you for all the flavors of protocol. All you would have to do is read some manuals or just let it try combinations until it worked. First, you'd select the company's system exclusive code or write in the proper code if it wasn't in the look-up table. Then, you'd select protocol type (handshaking, no handshaking, etc), and whatever else is needed and see if it works. If not, try again! When it works, save that patch combination and name it.

Another nice MIDI application would be a software version of the Yamaha MEP4. It should shell around a sequencer program and filter/modify MIDI data passing through the MIDI interface. You could use this to split your keyboard, change controller data, mask data, etc. It would save another piece of studio hardware.

Finally, some comments. (What? Not done yet?) As a horn player, I don't like the Akai EWI. If it is truly aimed at wind players, let it play like a wind instrument. Otherwise don't call it that. It is really a totally new device, as the Chapman Stick is to a guitarist. To me, a controller should be very similar to something you already know. In spite of some necessary adjustments, pianists don't have to play scales at 40 bpm to get used to a MIDI keyboard. In my opinion, it also needs variable key touch sensitivity, so you can touch the keys and not trigger it, and a programmable "octave key response lag" so that when you jump octaves, it doesn't jump down additional octaves before you get to the octave key you want.

Lastly, a suggestion. Whether a horn player chooses Yamaha, Akai, or some other wind controller, there are some features that were totally missed. How about programmable fingerings or selectable ROMs, so a bassoonist or flute player doesn't have to change technique? Also, alternate fingerings could be programmed to create slight detunings or some other MIDI parameters. (Remember those old horn players getting wah-wahs on the sax through the use of alternate fingerings?) You could create a unique fingering, determined by the player, to put the axe into system exclusive mode. Then you could send any pre-programmed control message by hitting a particular key combination. It could be program change information, MIDI channel change information, device number change, whatever.

I really don't want to do this stuff after work. I just want to play music. That's why I offer these comments, for whatever they're worth. If someone uses any part of this, please remember me with a prototype to test!

**Ken Fink
Cranbury, NJ**

Thanks for the suggestions and ideas, Ken. I'm not sure that everyone would agree with all of them, but it's certainly a good way to get a dialog going. New product ideas and updates to existing products often come about because of these kinds of comments.

Dear Music Technology,

I am a poor guy who happens to be interested in the electronic music world. My setup consists of a Max from SCI (the first synth I ever purchased and, believe me, nobody warned me), a CPS101 Electric "Piano" from Casio with an unbearable sound, but the cheapest touch-sensitive MIDI keyboard you can find, an FB01, an EX800, a TR505 and a Commodore 64 computer. And if this isn't sounding so bad to you, wait 'til you hear them through a BX600 mixer from Boss and an old Fender Champ guitar amp. Creepy, isn't it?

Anyway, by selling my last underwear (I get around wrapped in a couple of soft keyboard carrying cases), I've got enough to buy an Amiga 2000, a Roland S50, a stereo amp, monitors, etc. By now I suspect you must wonder, why am I writing and telling you the story of my life? It's because I just finished reading the second part of the Technology article about the Fairlight CMI III and, believe me, it was a lasting experience.

It was something like when you buy Auto Trend magazine and you read about the specifications and performance quality of the new Ferrari or Porsche, even though you know

Wanna use all those mic inputs?

One of the oldest and largest specialists in computer and printer interfaces, Applied Creative Technology Inc. brings its analog and digital expertise to the recording and music industry with their new DB8 Console Interface.

If you have a mixing console with mic inputs, our DB8 will open up worlds of new potential for you. The DB8 converts high-impedance instrument or effects box outputs to low-impedance balanced mic-level signals, without coloration or loss of dynamics. And it gives you eight channels of this conversion in an economically priced rack-mounted unit.

You'll be surprised at how much you can do with it!

With instruments like keyboards, guitars, synthesizers, samplers and drum machines, you can feed each individual voice or sound output channel into the mic inputs of your console. In addition to faithfully maintaining the original sound quality of your instruments, the DB8 frees you to be more creative and get better first generation recordings. You'll be able to mix, EQ, add effects, and spatially manipulate each voice, instrument or sound during recording, just as if you were mixing a bunch of live mic's. Plus, since the DB8 converts your instrument signals into mic-type signals, the cables between the DB8 and your console can be as long as any mic cable, which means you can locate your instruments anywhere you want—again without signal degradation or any increase in noise.

With effects boxes like digital delays and signal processors, the DB8 lets you feed their output channels into your console's mic inputs, giving you the ability to individually EQ, pan, filter and further effect them. This beats the heck out of a console's simple "effects return" which gives you little or no control.

In one take you'll be able to independently mix, EQ and position all instrument sounds and the effects processor versions of these sounds. Imagine the creative power this will give you!

The DB8 is built using today's technology for today's instruments and consoles. Each low noise (S/N >93db) highly isolated channel (>100db) has a standard quarter-inch phone jack input and a 3-pin male XLR output. To further maintain the integrity of your original instrument sounds, its frequency response is razor flat from 10HZ to 25KHZ ±0.2db and its harmonic distortion is less than .008%. It's even protected against phantom powered mic inputs, and has an LED overload indicator for each channel in the unlikely event any input signal approaches its +10.5db max input level.

To quote one user: "It's a dream come true!".

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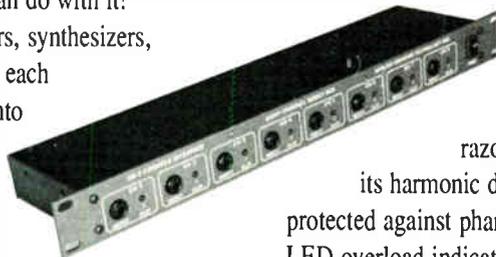
that you might never be able to have one. But dreaming gives meaning to life, right? I now want to see something like it about the Synclavier.

Keep the good work going and thanks for the good times.

Tom Karalias
Long Island, NY

Thanks for the feedback Tom. Just for your info, we did do a piece on the Synclavier way back in our inaugural issue. Unfortunately the actual magazine is sold out, but you can order a photocopy of the article by writing our subscription department.

Dear Music Technology,

I'm a bit confused by your Patchwork section. I'd like to try out some of the ESQ sounds, but don't know which preset from which to start editing. A step by step explanation would help me, and probably other readers too who aren't seasoned programmers. Thanks for any help you can offer. I love the magazine and am even considering subscribing to Home & Studio Recording. Keep it up.

Jeff Diorio
Kansas City, MO

Sorry about the confusion Jeff, but we've been working under the presumption that most everyone knew how to take advantage of the patches included in the column; I guess we were wrong. Anyway, it doesn't matter what patch you start from when you enter the parameters, you will always end up with the same result. The patch charts cover every parameter available on the respective instruments, so if you plug them all in, then they will all be different from the patch you start with (unless by chance they happen to be the same). The only problem that could occur is if you didn't enter all the listed parameters, then you could end up with something different, but if you enter all of them that will not occur.

If you have a patch on the synth you want to plug the new program into, that you think may be similar to the Patchwork sound (based on its description), then it may be easier to start with that – some synths have a basic or initialized patch for programming purposes – but the final result won't be any different. It's purely a matter of convenience. Hope that clears up the confusion.

Dear Music Technology,

Hey Chris (Many), get with the program! Not everything in this wide world of music needs to be MIDI capable. In fact, when it comes to high-quality music typesetting,

Oberon (which you reviewed in the November issue) lets you work with very few limitations. As a musician/composer I need high-quality music scores for the studio, and the other products you discussed are sorely lacking when it comes to conveying music concepts on paper. When you hand a piece of music to a musician to read and the music looks amateurish you can bet that it will come out sounding amateurish. Music copying is an art form that, by the sound of your article, has either died out or is no longer needed. I guess it's obvious that I disagree with your review of the Oberon Music Typesetting software.

Oberon offers a software package that works on any IBM environment, has strong WYSIWYG features, and outputs to HP laserprinters. The HP laserprinter is used because it has good memory capacity as a standard feature and does not require the redundant and relatively slow processing language of Postscript. The advantage of using fonts in a text mode, as you should know, is one of conserving available memory and quality by vectorizing fonts. This method also greatly enhances the accuracy of on-screen proofing before final output. Music is a creative process that needs space in order to fully be expressed. A music notation system needs the same ability for the creative musician/composer to convey accurate performance instructions, whether it be a simple lead sheet or a modern/avant-garde score with special symbols created by the composer. (Oberon has the ability to supply custom fonts.)

It would well be worth Music Technology's time to take a further look at Oberon and put it through its paces. Then I feel you will see why those of us who are using this software package are really happy with it.

Craig Stewart
Boulder, CO

Chris Many responds: I appreciate Mr Stewart's comments and am glad he's happy with the Oberon system. My opinions of this system are simply that: my opinions. The features that Oberon omits from their product, such as MIDI compatibility and dot matrix output, merely limit the available market. This is particularly true because there are several solid software programs that provide those functions, in addition to the high-quality Laser printer output professionals desire. The fact is, a music typesetting product based on text editing that leaves out such basic features as Cut and Paste or Block Move can only be summed up in one word; incomplete. And for $700 I'd expect to get a full-featured whiz bang piece of software for my PC, not a partial port over from an existing mini-computer application that didn't take into account the technological advances made in the field of music over the last decade. ■

360 Systems
Pro Midi Bass

If you're looking for sampled bass sounds but don't want to deal with the hassles of a sampler, this new "bass-in-a-box" may be for you. *Review by Matt Isaacson.*

LONG BEFORE SAMPLERS became plentiful and relatively inexpensive (well, two or three years is a long time nowadays), ROM-based drum machines came on the scene and made their presence felt. With real drum sounds encoded into their chips, they deftly swept the existing syntho-tronic beat boxes aside into a low-budget specialty niche. As time has passed, manufacturers have got better at providing the in-demand sounds and features while holding down or dropping prices. Consequently, drum machines, in spite of generally being a notch or two below samplers in terms of potential flexibility and sound quality, continue to represent good value for many users, and new models are regularly being introduced to a receptive market. In fact, given the sort of features which many recent models possess, there are no doubt many users who would find drum machines preferable to the more sophisticated alternatives, regardless of price.

Conceived with a similar purpose – if a somewhat less broad appeal – is the Professional Midi Bass from 360 Systems. It was developed as a successor to the original Midi Bass in an effort to address a large number of issues raised by users of

that model and to provide a more widely useful and professional product. Was the effort worth it? Ultimately, only *you* can decide for yourself, but I will say that the Pro Midi Bass goes far enough beyond its predecessor that a direct comparison would not be worth the space it would take up.

What It Be

THE PRO MIDI Bass can basically be described by the needlessly flippant phrase "Box o' Bass." It gives you real bass sounds via samples, of a full range of acoustic and electric bass notes – none of this overused DX7 stuff or mushy analog synth patchwork (although a very understated Minimoog-type sound has been included, lest you hanker after the synth which the Pro Midi Bass may have unseated). You don't have to wait for these sounds. They live in ROM chips inside the box, in finest drum machine style, and the Pro Midi Bass is ready to jam by the time your finger has left the power switch. Those who are not impressed by this point may still be swayed by considering that the waiting time applies not only to initial power-up but also to preset changes – *all* available sounds are on-line at once and immediately accessible via program changes.

The package is simple, small and clean – a 1U rack-mount box. You won't hurt yourself lifting it or wear yourself out deciding where it should go. A power switch, volume knob, 2×16 LCD and a handful of programming switches and LEDs are on the front panel. The rear panel is even simpler: MIDI In, Out and Thru jacks, and audio out (plus the obligatory LCD viewing angle adjustment, which probably should have been up front). Upon power-up the Pro Midi Bass informs you via the LCD as to the revision level of the installed software as well as the number of bass sounds you have. This is done because the number is not strictly fixed – very ample room has been left for expansion of the internal sounds. The stock unit comes with eight sounds, which use up less than half of the available ROM sockets inside.

The Pro Midi Bass is a monophonic playback system – it is capable of generating only one sound at a time. This is not to say that the Pro Midi Bass only lets you *get at* one sound at a time. In fact, the basic structure which makes up a preset holds places for four sounds. Each preset (there are thirty, by the way, in the latest software) lets you define lower and upper zones which map out areas of a MIDI keyboard over which the instrument will ▶

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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!

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- 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.

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Pan ID: MIDICONCEPTS

respond. These zones can be anywhere from one note to over three octaves wide and may adjoin one another or be in completely separate parts of the keyboard. As a result, it's easy to put the sounds where you want them, even if the keyboard you're playing from is not geared for that sort of splitting. A transpose value can be set separately for each zone, allowing, among other possibilities, the same set of pitches to appear in two different parts of the keyboard at once.

Within each zone, two sounds are selected from the available set – one is a given threshold, and the other is the "normal" sound which plays at all lower velocities. A very obvious use for this normal/accent feature is to use a plucked or picked sound normally, with a slap or popped sound on the accents.

Bass-ic Editing

A HANDFUL OF other controls which fall within the preset category are available and are separately adjustable for each of the four sounds in a preset. Volume sets the loudness of each sound relative to the others. Decay and release control the dying away of the sound after it is triggered – decay while its key is held down, and release after the key is released – over a range from instantaneous to very long term. A low-pass filter lets you cut out some of the high end of a sound to keep it from taking over. The filter has no envelope and is not affected by note velocity, but the cutoff does track note

consider buying two of these, I might start looking around for a cheap sampler instead).

The instrument also provides some options with respect to how it will decide which note to play when presented with more than one playable note at a time. The basic mode is last-note priority, in which the latest arrival always gets preference. This is the mode of choice for players who occasionally hit two keys when trying to hit only one, as it tends to make such mistakes less noticeable. Low-note priority gives preference to the lowest note when more than one note is played. Using this mode, the Pro Midi Bass can be directly layered with another keyboard sound on which some combination of bassline, chords and melody is being played. It'll pick out and play the lowest note only, allowing for more flexibility than with a hard split point. High-note priority is similar, but chases the highest note being played at any time. Some players may be disappointed to learn that the voice cascade and note-priority settings are global rather than being part of each preset.

Other non-preset controls include enable/disable of MIDI patch change, main volume and pitch-bend response (plus or minus three semitones), MIDI channel setting, master fine tuning, and keyboard transpose interval. This last one specifies a transpose amount which operates on both zones of whatever preset is called up. It works in conjunction with a transpose button on the front panel which allows you to instantly switch the transpose on or off

sound: namely that the loops are quite short, typically only one cycle of the sound. In some cases the sound almost appears to detune slightly upon entering the loop, although in just as many other cases this effect is not very noticeable. Where offensive, it can be got around only by releasing the note before it goes into the loop. Another problem which appears in a widely varying degree is that of abrupt changes in the tone of a sound at the multisample splice points. In fairness, this problem afflicts all samplers to some degree when used for multisample playback, because it's pretty hard to record two different things and end up with samples that sound the same. For that matter, playing the same note on two different strings of a bass guitar will give you two different sounds, which is precisely the problem (or indicates that this shouldn't be viewed as a problem . . .).

One Standup Bass sample was noticeably noisy, and fret buzz was audible in one of the Funk Thumb samples. Also, in general, small clicking sounds sometimes accompanied voice stealing. These are things which might complicate a critical recording session, but are not likely to get in the way during performance, unless you are heavily given to bass solos. Meeting the Pro Midi Bass on its own terms will enable you to get good results with it. 360 Systems are reportedly at work on an alternate sounds library, including double-length sounds aimed at minimizing the loop problem by simply providing more playback time before reaching the loop. Based on the example sound provided for review (Picked Roundwound J-Bass), they're making big strides in all areas of sound quality.

The Scoop

BY NOW IT may be apparent that in many respects the Pro Midi Bass has playback capabilities very similar to those of a sampler loaded with bass sounds. Why not just buy a sampler instead, and be able to sample your own sounds, etc, etc? It's pretty simple. The Pro Midi Bass is not for everybody. If a sampler is what you really want, don't buy the Pro Midi Bass. If truly ultimate sound quality and realism at any cost are what you're after, consider hiring a bass player. But, if what you want is flexible, reasonable-quality, no-fuss sampled bass playback at a reasonable cost – particularly, if you already have a sampler and don't wish to constantly devote a substantial portion of its relatively expensive voice and memory resources to a use which doesn't really require them – the Pro Midi Bass may be just the thing for you. ∎

Samples *"The Pro Midi Bass can basically be described as a 'Box o' Bass.' It gives you real bass sounds via samples, of a full range of acoustic and electric bass notes."*

number. Velocity control for the "normal" sound is an on/off switch – velocity either does or does not affect the loudness of a note. (An actual adjustment of velocity sensitivity would have been helpful in cases where there seemed to be too much velocity response.) Velocity control for the accent sound takes the form of a threshold velocity at which playback switches from the normal to accent sound.

In spite of all these nice features, you can still only play one note at a time. For those nice octave and parallel-fifth effects, there is the second-voice cascade feature. When enabled, the Pro Midi Bass will respond only to alternating incoming MIDI notes – intervening notes are echoed out to the MIDI out jack, where, budgets willing, a second Pro Midi Bass is waiting to pick up the overflow (you were wondering why a rack-mount box has a MIDI output, weren't you?) This can also produce some interesting effects if the overflow module is something other than a Pro Midi Bass. (Frankly, though, by the time I had to

(eg. for a quick change of octave in any preset). There's also a memory-protect control and a test note function which allows you to play sounds from the front panel – a helpful tool when a bad MIDI connection is suspected.

Da Sounds

WHAT ABOUT THE sounds themselves? The stock selection of sounds includes: Fingered Steinberger, Fingered and Picked Flatwound P-Bass, Rickenbacker, Funk Thumb, Funk Pop, Standup Pizzicato and Minimoog Square Wave. All of them are multisampled to allow them to cover a keyboard range of three octaves or more, and all but Funk Pop are looped to allow long decay times. The names describe the sounds pretty well, and the basic character of each sound is also pretty good.

On a more microscopic level, however, the quality of the sounds is a bit spotty. One problem is a consequence of the limited recording time available for each

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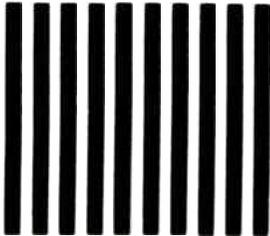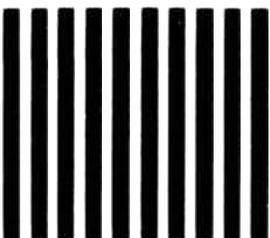
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Primarily known for his work with the jazz groups Weather Report and Steps Ahead, drummer, composer and dedicated Mac-aholic Peter Erskine has immersed himself in modern music technology. *Interview by Bob O'Donnell.*

PROGRESSIVE MODERN JAZZ is ripe for sonic explorations with new technology. The kind of music you hear on recordings from labels like ECM and Passport Jazz has an adventurous, thoughtful spirit that, though difficult to describe, is particularly appropriate for creative uses of synths and MIDI controllers. And not surprisingly, some of the most interesting and successful applications of modern musical instruments have come from the marriage of MIDI technology and jazz. Guitar synthesists Pat Metheny and John Abercrombie, keyboardists Joe Zawinul and Lyle Mays, wind synthesist Michael Brecker and a few other great jazz players have produced imaginative, expressive music that goes a long way toward exemplifying the *potential* of the new instruments.

Another player who deserves inclusion in that elite group is drummer/percussionist Peter Erskine. Like most of the others, Erskine first gained notoriety for his playing ability on the acoustic version of his main instrument, but over the last few years he's earned recognition for his work with electronic drums and drum machines. He's also a synthesist and composer and has begun to produce and record his own compositions with the help of his trusty Macintosh and his well-equipped home MIDI studio. One of his most recent projects was the score of incidental music for a successful production of Shakespeare's *A Midsummer Night's Dream.*

Erskine's second solo album, *Transition,* available on Passport Jazz records and Denon CDs, combines all the elements of his musical personality. It features straight-

ahead jazz tunes which highlight his acoustic playing – including his aggressive, signature snare work – and a tasteful electronic drum solo. He also uses sequences which either the band (John Abercrombie, guitar synth; Marc Johnson, bass; Joe Lovano, sax; Bob Mintzer, sax; Kenny Werner, keyboards; Don Grolnick, keyboards; and Peter Gordon, french horn) or he alone, plays along with. In fact, a very impressive "out" sequence with lush synth textures that was composed by one of Erskine's collaborators, Vince Mendoza, is one of the album's highlights.

In addition to his own record, Erskine has lent his in-demand timekeeping abilities and distinctive snare work to records by Weather Report, Steps Ahead, Bass Desires, John Abercrombie, Mike Stern and Gary Peacock, to mention a few. Interestingly

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enough, the same snare work which helped him establish his reputation as an acoustic player also provided him with an opportunity to get directly involved with technology. Specifically, the story involves the Oberheim DMX digital drum machine.

"When I was in Weather Report we had used the Linn LM1 drum machine on one record and the word we heard was that Oberheim was working on a drum machine of their own. Well, Tom Oberheim used to come to Weather Report shows and I had met him, and a short while later a keyboard tech for Joe [Zawinul, synthesist and leader of the band] who had a connection with Oberheim called me up and I went into the studio and recorded about three sets worth of drums for them. Most of the sounds on the original DMX, including the snare, were mine."

But the question is, how did a drummer who grew up listening to Art Blakey, Elvin Jones and other straight-ahead drummers *first* get into technology?

"I've always been intrigued by sound," Erskine replies, "and when I was a young musician my father encouraged me to study keyboard. I liked the piano but I was more intrigued by the Hammond organ in our house. I used to love to play with the drawbars and change harmonics and all that stuff. Then my folks got me a Wurlitzer organ, which was not a synthesizer, but they did have a book that showed you how to get the sounds of particular instruments and I enjoyed creating new sounds."

A natural outgrowth of Erskine's interest in sounds and rhythm was the decision to become a drummer/percussionist. After finishing school, he toured with the Stan Kenton band and then trumpeter Maynard Ferguson. A few months after the late Jaco Pastorius saw one of the Ferguson band's shows, Erskine got a call from Jaco and synth master Joe Zawinul about joining Weather Report. "And so there I was, standing next to *the man*," Erskine recalls, "with his ARP 2600s, the Oberheim modular system, a Prophet 5 and a few other strange little ARP synths. I remember the big news at the time I joined was that Joe had just gotten the Prophet and that just set him free.

"I guess I've always had an ear out for the hi-tech stuff," Erskine sums up, "and I've been fortunate enough to be working in musical situations with musicians who use it themselves or who have music that encourage that kind of experimentation."

Speaking of which, Erskine experimented a great deal himself, trying a number of different electronic drum/percussion gadgets to create new sounds for himself. He began his search early on. "Before I joined Weather Report I was a frequent visitor to NAMM shows and I remember when a few electronic things like the Synares came out. I was playing around with one of them and I found it very easy to come up with nice, fat, low frequency sounds – it sounded kind

of like hollowed log drums – so that was my first involvement with drum electronics."

The real future for Erskine, however, lay in electronic drum kits and drum machines. "We used the LM1 right after it came out on a tune called 'When It Was Now,' and I'm pretty sure it was the first instrumental, at least with a band that had any kind of improvisatory thing, using a drum computer. It was really exciting programming it," Erskine recalls, "I got a crash course from Joe's keyboard tech, and we loaded up the tune and played along with it in one take. It was really fun.

"Then I left Weather Report, moved to New York and started playing with Steps Ahead," he continues, "and all of a sudden

the word was out that Bill Bruford was playing this new instrument. So I went to a concert to see him and the clarity of his sound was incredible. It turns out it was the original Simmons SDS5. They looked really painful to play, though, and it turns out that, for durability's sake, they were using the same tremendously hard plastic that the British police used for their riot shields. So if you got really excited and wanted to play something loud you could really hurt yourself. But the *sound* knocked me out, so I went into a kind of 'wait and see' mode."

In the meantime, Erskine also became the operator of Steps' Oberheim System. "I really liked the way that system (the OBX synth, DMX drum machine and DSX sequencer) operated and the fact that the ►

World Radio History

► DSX sequencer offered a tapeless form of recording was amazing. Of course, editing and all those kinds of things were tricky, but when you haven't experienced the beyond, you can be pretty knocked out by what that system could do.

"Then I finally decided to take the Simmons plunge. I did a NAMM show for them and got to use an SDS7. I remember I wasn't exactly sure of how the thing worked, so I just set up some ostinatos on their sequencer, but I loved the sound possibilities, the multi-note samples. I found most of the sounds to be grainy and noisy, however, so I went looking for a 5. I ended up doing a lot of phone work with drum technician Vince Gutman, and he put together a system with footswitching and triggering capability for me. I had it split between the 5 and 7 because I liked the digital attack capability of the 7, but I also liked the thick analog stuff from the 5. It's a tremendous sounding instrument, especially in the lower frequencies."

Erskine's next major gig was a reunion with Zawinul and Co. for the Weather Update tour. Along with the new band came a new set of drums, the Yamaha Electronic Percussion System, which helped him overcome some sound problems that he was facing. "I scrapped my triggering because I was so intrigued by the sounds the Yamaha FM system was producing that I wanted to be able to access those sounds without hearing a tom-tom as well. At the same time I was realizing that whenever we got a great snare sound, as soon as we opened up the tom mics we lost what the snare drum sound was, so I said, 'Why not go for a direct tom-tom sound?' I still need a real bass drum and snare drum to feel like a drummer," Erskine admits, "because there's just so much in my vocabulary of playing that needs those two."

Erskine was impressed by the unique sounds of the system, and its MIDI implementation met with his and Zawinul's complete approval. "Zawinul was intrigued by the heart of the system, the PMC1 Percussion-to-MIDI Converter. Its MIDI capability is rich and deep with things like Dynamic Note Shifting, which allows you to crossfade between different notes on the same pad depending on how hard you hit it, so you can play melodies. I remember Joe said, 'I don't know how you drummers have gone so long without this.'"

After a brief stint with the Korg DDD1, Erskine also changed his drum machine preference to Yamaha. "I did a bunch of product demonstrations and clinics for Yamaha in Japan and I got to spend some time at their R&D center, which was really exciting. That's also where I really discovered what the PMC1 could do and found out about the RX5. The RX5 sounds are real clean and not only are they great to begin with, but they also lend themselves well to effects. The thing is confounding at

first, but once you get into it and figure out the architecture, then all of the buttons have a rich meaning."

DISCUSSING DRUM MACHINES with a drummer who's known for playing music with a great deal of freedom may seem a bit odd, but Erskine does not feel threatened by the machines. He does admit that many people, particularly some in the recording industry, underestimated the impact that they would have, but he feels that the instruments can serve an extremely useful role in music. And because of his interesting views on the notion of "time," he doesn't think that variable tempo devices like the Human Clock or Time Commander are always necessary when you play along with drum machines (although he does think they could be useful to create click tracks for recording and post-production applications).

"I understand they work very well," says Erskine, referring to the devices, "but to be frank, the idea of using them live doesn't appeal to me because I like to play with a clock. See, a constant clock, like that provided by a drum machine, is like a good drummer. A good drummer is rock solid, and playing with a clock gives you the chance to breathe with it because the reference point always stays the same.

"Now time is an absolute," he continues. "You always hear guys say 'ahead of the beat' and 'behind the beat' and that really confuses a lot of drummers. It's not a

pattern on a drum machine is restricting. "You can either be locked in your room or you can let your imagination go," Erskine explains. "We all have a tendency to rush and all these other time tendencies and that's the human thing. I can do plenty with a backbeat that's steady on two and four. If you want to be able to speed up a little bit and still have the drum machine give you a backbeat, then that's fine. But when I think of a backbeat I generally think of things that are in the pocket. You want that stuff to sit right there and with a drum machine you can do that. Metheny's band does it marvelously," Erskine adds as an example, "because there's a percussionist up there and unless you really watch and see that he isn't playing a tambourine or whatever, you'd never know there was a machine."

Erskine also has unique opinions about and approaches to programming drum machines, which he normally does from a keyboard. "For realistic drum programs, I do it in a linear, through composed kind of way. I go left to right, from start to finish and, if I'm working with a sequencer, I generally put my drum parts on last. Being a drummer I like to see what the tune is doing once, and then I just put a drum part on it.

"I'll make a pass with bass drum and snare and then I'll put things on top. I selectively quantize certain parts, but not others. What this does is get you away from repetitive patterns. As a drummer I've never thought in terms of four- or eight-bar patterns, so why should I when I program? I

"When your sax player sounds like vibes and your vibes player sounds like a guitar and your guitar player sounds like a sax, you feel like saying, 'OK, why don't we all get back in our seats'."

conscious decision you make, you just do what feels good, but there's an absolute, there's a center to the beat, just like there's a center to pitch. If you want to deviate from that, then that creates tension and music's moving forward is based on tension and release. For example, if a soloist is playing along with a clock and he just goes 'da-da-da-da-da-da-da-da,'" Erskine sings very evenly, "then it's no good. But if he does something like 'doo-by-ya-ba-da-be-do-be,'" he sings with a great deal of freedom and swing but still against an equally rigid clock, "then it's great. Now if somebody wants to start changing the tempo and saying, 'Oh, we want that freedom,' I think it's just because the guys can't cut playing with a machine. I mean that's my point of view. If you keep the clock, that gives you the chance to create subtleties which give the music a sense of forward motion. You can get the 'rubber-banding' type of effect."

Similarly, though initial impressions may lead you to believe the contrary, Erskine does not feel that playing along with a

think of it from beginning to end, as an unbroken thread, because that's what timekeeping is all about."

Not everyone Erskine comes into contact with agrees with his approach. "I was in the studio laying down some drum machine parts and the engineer turned around and said, 'These drums aren't all hitting together. Did you quantize this?' and I just gave him a shooing motion and said, 'Just turn around and mix; don't worry about it.' I like things to be a little sloppy," Erskine explains, "and the fact that things don't repeat is real good in my mind. I mean basically, I'm just improvising every time I put down a drum part."

IN ADDITION TO his work with electronic percussion devices, Erskine has jumped headfirst into the synth and MIDI fray. His home studio setup consists of a Casio FZ1 and CZ101, a Roland Super JX, a Korg DW8000 and Poly 800, a recently MIDI'd Oberheim OBXa, a Yamaha DX100, the RX5 and the

PMC1 and TX816 rack. "Another instrument that I really love is the Roland MKS20, which my associate Vince Mendoza has and I occasionally borrow," Erskine adds. "It sounds great and is really useful."

Erskine ties the system together with his beloved Macintosh. At the time of the interview he had been using Southworth's Total Music for sequencing, but he's considering switching over to Mark of the Unicorn's Performer. He also uses Opcode's Librarian program for his TX.

The combination of the Mac and his synths have had a dramatic effect on Erskine's musical life. He's grateful for all the capabilities that the technology has given him and he's basically caught the MIDI bug in a serious way. He also has some humorous anecdotes about how MIDI has affected a lot of other musicians. "Bill Bruford has a great line, the punch line of the joke about bands used to always be 'Four musicians and a drummer,' but now it's 'Five guys standing around reading manuals.' And John Abercrombie made the joke that we should start a DA, Digitals Anonymous, and have meetings where people could stand up and say, 'MIDI has made my life completely unmanageable.'

"It's just teeming with so many possibilities," enthuses Erskine, "and you just want to jump right in, but then you take on all this technical responsibility because the stuff's gotta work and if you're going to use it you've got to know what you're doing."

Erskine also points out that though the potential of the technology is great, so is the chance for overkill. "In Steps things got a little goofy because you could get to a point where you didn't know who was playing what. I mean obviously, the improvisational ideas of one guy are different from another, but when your sax player sounds like vibes and your vibes player sounds like a guitar and your guitar player sounds like a sax, you feel like saying, 'OK, why don't we all get back in our seats again.' I think it's always good to be able to take a step back when you're dealing with these new instruments and see what you're trying to accomplish in a given piece of music."

Despite his interest in time and timekeeping, the new musical direction Erskine would like to take plays with the notion of time and takes advantage of the sound creating possibilities of his instruments. "What I want to do next with sequencers is create music with harmonic changes and development that doesn't fall on bar lines. I want it to have an amorphous kind of cloud quality and I want to imbue it with the innate, creative intelligence that the best improvised music has to offer, which to me, is jazz."

Erskine is quick to point out that he's not referring to New Age music. "I resent listening to New Age because a lot of

people who play it, their touch on the instrument really sucks. And to my ears, most 'New Age players' can't improvise their way out of a wet paper bag. Plus, most of what I've heard has such a predictable harmonic sense that it's very

"I think what these instruments and what technology gives all of us is a chance at achieving our artistic ideas. They're like the keys to the Magic Kingdom."

unappealing. What I want to do is create music that really expects or demands a lot from the listener but is eminently listenable and enjoyable at the same time."

A tough task perhaps, but one that Erskine feels he can achieve, particularly with the help of his electronic tools. "I'm

not a very good keyboard player, but I'll tell you, something happens when it's me and my Mac and I get alone and work. I'm really surprised by what comes out and I think that's what these instruments and what technology gives all of us. I mean some people out there have not had much formal training on their instrument but they have an artistic idea and these instruments give us a shot at doing the best thing and achieving those ideas. As far as I'm concerned," Erskine chuckles, "they're like the keys to the Magic Kingdom." ∎

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J·A·P·A·N
showing out

The music trade show held recently in Tokyo offered a glimpse into what many Japanese manufacturers have in store for the future and a hint of what may be introduced at the NAMM convention.

MUSIC & SOUND EXPO '87, otherwise known as the Tokyo Music Fair, was held at Tokyo's Museum of Science Technology (Kagaku Gijutsu Kan) and at the nearby Hotel Grand Palace between October 15 and 18. The fair is held every two years in Tokyo and unlike trade shows in this country, it is open to the public. During the weekend of its run, the show site was full of general end-user attendees, many of whom were teenagers anxious to see the newest gear.

This year's fair hosted about 110 exhibitors, some from countries outside of Japan. Despite the large number, however, there weren't that many new products that were actually introduced. A number of companies did offer very interesting demo exhibits of products that may never be introduced, though (and some of them *may* see the light of day outside of the R&D lab). Regardless of which, they give a very good insight into what direction or directions these companies may be heading in.

What follows is a brief description of some of the more noteworthy products that were either introduced or simply displayed at the show. Keep in mind that just because a Japanese company introduces a product to the Japanese market doesn't mean it'll be available in the US; but there's certainly a good chance that it will be.

Korg 707: This is a performance synth that comes in four color variations and a stylish rounded form. It has only 49 keys, but being light in weight and compact in size, it is best suited for use as an over-the-shoulder keyboard. The keyboard is sensitive to velocity and aftertouch controls, and the pitch-bender and control wheel are placed at the best location for use as a "shoulder" keyboard. The tone generator inside the 707 is the same as the DS8 and ROM cards for the DS8 are compatible. Each voice features two digital oscillators and each oscillator has a Timbre EG with seven parameters, and an Amplifier EG with six parameters. Voices also have their own six-parameter Pitch EG, so as to be able to create complex sounds.

Several modes are available for performing: 1) Single mode, which is eight-note polyphonic with one voice (or patch); 2) Double Mode, which uses two different voices on a split keyboard; and 3) Layer Mode, which is two voices combined in unison. When used with a sequencer, it is possible to play eight parts with different sounds, which is 4) Multi Mode. The synth uses FM technology, like the DS8, so it is suitable for producing metallic sounds; however, if you use Layer Mode, you can create very thick and heavy sounds. The instrument has a large memory: with a RAM or ROM card, 200 programs and 20 combinations are immediately available. (Without a card, 10 programs and 10 combinations are possible.) The 707 has two audio outputs, so it is good for both live performance and recording applications.

Roland S550: Roland unveiled a module version of the S50 with the same general sampling specs. The S550 is also equipped with a mouse and can be controlled remotely by the RC100 remote controller. The memory capacity is double that of the S50; it has 1.5Mbyte of internal RAM. Accordingly, the number of sounds and patches have also increased: two groups of 32 sounds and 16 patches are now possible. In fact, put simply, the S550 is the equivalent of two S50s in a box. Consequently, it is possible to obtain detailed multi-sampling and high grade sounds. As for peripheral memory equipment, Roland also displayed a prototype of a hard-disk drive. With it, you can store the equivalent of ten floppy disks on one hard disk. Also, the loading time is greatly shortened, which is beneficial for live performance.

Another big modification to the sampler is that there are eight outputs instead of four. You can assign different MIDI channels to each, so you can use it as the equivalent of eight samplers.

Functionally, the S550 uses Version 2.0 software of the S50's operating system. The pre-trigger and previous sampling functions are included in that, so user sampling is easy to do. Of course, you can also use the existing sound library for the S50.

Korg X2000 I, X2000 II: (Demo ▶

Exhibits). The 2000 I is a keyboard that has preset sampled sounds à la the Kurzweil, but cannot actually sample. The 2000 II is a keyboard that has sampling capabilities as well. They are both more developed versions of the DSSI and they both also have sound sources besides sampled sounds. The plan is to equip them with many more functions but the details are unknown at this moment. They may be released in 1988.

Yamaha V2: The V2 debuted here as a new 4-operator/8-algorithm FM synth. In the US it will be called the DXII. It is eight-note polyphonic, and its 61-key keyboard has aftertouch and velocity control. Like the TX81Z, with which it is compatible, waveforms beside a sine wave can be used as waves for operators. As a result, even though it's a 4-operator model, a variety of sounds can be created. Frequency Fine, Fixmode, EG Reverb, Transposed Delay, Pan, and Chord Set are all parameters available for sound-making functions. It

also has micro tuning, which became popular with the DX7II instruments. One of the unique functions of the V2 is called Quick Edit. This allows you to change nuances of a sound's attack, release, brilliance, and volume with one touch. It is a convenience not only for beginners, but for all programmers because it can be used to change details of the preset sounds as well.

One hundred and twenty-eight new preset sounds were created just for the V2. The data for them is also compatible with the TX81Z. A RAM4 cartridge can also be used for storing more programs. As for performance functions, there are detune parameters, note shift between each sound and Alternate, with which you can produce different sounds at each successive key on. You can split the keyboard in eight sections, create partial layers, or play eight sounds together in unison. Because each of the eight sounds can receive on different MIDI channels, it can be used effectively as a sound source for a sequencer.

Casio VZI: (Demo Exhibit). Casio's new digital synth, the VZI, was just used for demonstrations at the fair. No detailed specs are available now, but this model uses the company's newly-developed LSI 2, which is totally different from both analog and digital synthesis. You will make sounds with it by compiling modules. The following brief specs are current as of now, but there may be some changes when this model is actually released: 16-voice polyphony; 61 keys; maximum of 256 onboard sounds (64 presets, 64 internal, 128 in cartridge). One of the instrument's features is called Combination Mode, where you can combine up to four sounds freely and create higher grade sounds. It's like an expanded version of Split or Layer functions. The overall sounds are very thick with Analog Brass and metallic percussive sounds being particularly impressive. Three wheels are available, one of which is dedicated to pitch-bend while the other

two are assignable. The price and release date details are unknown.

Kawai R50e: The R50e is the same rhythm machine as the previously released R50 but with a brand new set of sounds. The sounds are all 12-bit PCM sampled sounds, sampled at a rate of 32kHz. Like the original R50, 24 preset sounds are included, but they are more up to date than the ones in the older model. In addition to regular rhythm sounds there are some powerful bass sounds (including thumbing) and an orchestra/brass hit sample.

Functionally, the R50e is the same as the R50. You can program the panning, tuning and effect on each sound. The three available effects are flanger, short delay and gate and their inclusion gives the preset sounds more variety. You can store up to 100 patterns, combining 50 preset and 50 programmable ones, and up to 10 songs,

each of which can have a maximum of 225 parts.

Yamaha RX7: A new addition to the RX series is like a younger brother of the RX5, but with 100 preset sounds. They include not only percussion or drum sounds but also bass, guitar, DX sounds, sound effects, voice and reversed sounds. In a sense, these other sounds make the RX7 more than a rhythm machine. The sounds are all 12-bit PCM samples and you can change the volume, pitch, and decay of each of them. They even respond to pitch-bend. You can create a melodic phrase by changing the tuning note by note on guitar or bass sounds.

Both step-time and real-time recording are available. If you use a MIDI keyboard, it is possible to have real-time input with velocity. Also, because it has eight-note polyphony per sound, you could play a guitar chord. It responds to MIDI song position pointer data, so it can be a powerful instrument for synchronized recording applications.

Korg SQD8: This is a compact eight-track MIDI sequencer equipped with a drive for Quick Disks. Real- and step-time input are both possible as is quantizing of the real-time input (the minimum quarter-note speed is 48BPM). For step time, you input a note duration from the sequencer and a pitch from a MIDI keyboard, which makes it easier for beginners. Like a tape recorder, it has REW and FF buttons, which also make its operation easier. You

can see what you are currently doing on the big display. The ease of tempo and volume data programming is also a big advantage. When you mix the data on several tracks onto one track, if you move a data slider while listening to the playback, you can also mix in changes of tempo and volume data, which allows you to easily program ritards or fade ins and outs. The SQD8 is full of editing functions: erasing per measure, transposing +/− one octave, and repeating. You can also use it as a data

filer by utilizing the QD drive. You can store a maximum of 16,400 notes of sequence data and synth sound data.

Yamaha TXIP: The TXIP is a digital piano sound module using AWM (Advanced Wave Memory) sounds. It has a full 88-note range, so it is compatible with 88-key controllers and it is 16-note polyphonic. A total of five kinds of sounds are available on the instrument: two pianos, one electric piano, harpsichord and vibraphone. It also comes with a note limit function, a unique arpeggio effect and a stereo chorus. The TXIP is a welcome addition for those who want to reduce the size of a live keyboard system.

Yamaha TX16W: Yamaha's new digital sampler is subtitled Digital Wave Filtering Sampler. It features 12-bit linear stereo sampling, 16-note polyphony, a maximum sampling rate of 50kHz, and eight individual outs. 1.5Mbytes of internal memory is standard; however, you can insert a maximum of three 1.5Mbyte RAM

boards, making the total memory capacity up to 6Mbytes. The sampling rates are 50kHz, 33kHz and 16kHz; and sampling time on each sample is limited to 5240msecs, 7930msecs, and 16370msecs respectively. The sampling time within the limit can be used freely, so there is no waste in memory. The stereo sampling rate is fixed at 33kHz, and a maximum of 7930msecs is possible.

You conduct the sampling process by deciding the sampling rate and sampling time. Then, the sample is stored in a buffer. Sixteen of these buffers are available, so you should be able to sample as many times as you need, until you are satisfied. Next, you do trimming and looping in the buffer. For looping, if you designate a point, the machine will automatically search for

the best point around there. Of course, there are also functions such as crossfade looping, reverse and mix. It is also equipped with new functions such as reload where, if you failed an edit, you could go back to the unedited sound; and normalize, which increases the level of quiet samples. The samples completed in the buffer are called Waves, and 64 of those can be placed in Internal Wave Memory. You can then create a voice (sound) by combining the necessary waves from the memory. The TX16W boots up its operating system software from the system disk when it is turned on, so it is easy to upgrade to a new software version. It uses 3.5" disks.

Korg MIDI Controlled Mixer: (Demo Exhibit). This prototype unit is a programmable mixer with eight ins and four outs. It can memorize not only volume levels but input EQ, fade time, group setting and effects rates.

TEAC: Their new products include the MTS30 MIDI Tape Synchronizer; the RS2000 Digital Reverb, with which you can create a variety of reverb effects using pre-EQ, pre-delay, and decay parameters based on eight reverb modes; the DL2000 Comp/Limiter; and the GA2020 Noise Gate. The latter two are stereo units.

Toa MR8T: Toa introduced an eight-track cassette multi-recorder, the MR8T, with a tape speed of 9.5cm/sec. It is equipped with dbx noise reduction and eight input ports, and can simultaneously record on all eight tracks. In addition to the stereo outputs there are eight individual outs which can be used in conjunction with an external mixer. It is also equipped with sync in/out, which enables the machine to synchronize with a sequencer.

Korg's new rack series: The KEC42 is a multi-effector which contains four equalizer and two compander units. The equalizer is made up of three wide-range parametric components and one high/low shelving type. The KMX122 is a 12-channel line mixer which is best suited for use as a keyboard mixer. Finally, the DTI PRO is a digital tuner and the KMP6 is a MIDI patch bay having six ins and eight thrus. ■

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Yamaha RX7

Drum Machine

Yamaha's newest beat box is an interesting hybrid which offers both a lot more and a fair amount less than its predecessors.

Review by Howard Massey.

YAMAHA'S LATEST ENTRY in the mid-priced drum machine sweepstakes seems to have all the right numerological attributes: another "X," another "7." Will these bring to Yamaha the same kind of fame and fortune another "X7" instrument did some years ago? Tune in to the last paragraph to find out . . .

The general philosophy behind the RX7 seems to be that it is meant to "fill the gap" between the extremely low-priced, no-frills RX17 and the relatively high-priced and thoroughly full-featured RX5. To a certain degree, it succeeds in this quest, but, as we shall see shortly, Yamaha has, as usual, added so many new features – not to mention great sounds – to the newer instrument that, despite the major limitation of having only stereo outputs, the RX7 seems to almost surpass the RX5 in many areas. In other areas, though, it's clearly outperformed.

The Sound, The Look

BUT FIRST THINGS first – let's get straight to the main question: what does it sound like? Well, let's put it this way: the RX7 offers no less than 100 12-bit PCM pre-sampled sounds to work with. Of these, there are maybe 50 sounds that will be familiar to RX5/RX17 users (including electric bass, DX7 clavinet, marimba, brass and orchestra, and several reversed drum and cymbal sounds, because the RX7 doesn't have the "reverse" feature that the RX5 does). Then there are maybe another ten that are – well, shall we say "expendable"? (these include such gems as vocal "ha!'s," "uh!'s," and "get funky!'s") – and probably another 40 or so that are nothing short of terrific.

This last group includes a large number of well-recorded ambient snare drums, bass drums, and toms (even tom flams), along with a crystal clear bell tree, a beautiful steel drum, power and muted electric guitar notes and chords, and, for all you Kraftwerk fans out there, a number of cheesy-but-wonderful "rhythm ace" samples, camera shutter clicks, car door slams, and a very funny Euro-rock "bon" vocal sample. If you're going to use the RX7 just for standard drum kit sounds, you have lots of choices: nine different bass drum samples, ten different snare drum samples, two rim shots, seventeen (!) toms, four cymbals, and four hi-hats.

As the accompanying photograph shows, the RX7 looks very much like a scaled-down RX5, with the same 20 dedicated command keys, 10-key keypad, and 24 instrument keys. While the instrument keys themselves are not velocity-sensitive, the RX7 voices do respond to incoming MIDI velocity data. There's also a backlit two-line 32-character LCD, along with a series of status lights (for things like pattern/song mode, internal/external clock, etc). Of course, there are the usual stop/continue, start, and accent keys, along with four front-panel sliders for overall volume, click volume, tempo adjustment, and data entry.

On the rear panel, you'll find a single cartridge slot (for standard Yamaha RAM4 cartridges only; the RX7 doesn't accept ROM "waveform" cartridges), left/mono and right line level outputs (standard ¼" jacks, needless to say), a headphone output, and a separate click output. There's also a footswitch input, MIDI In, Out, and Thru, and a cassette interface for data storage and retrieval. No input is available

for external FSK clock signal, because the RX7 can only be externally synchronized to MIDI timing signals.

The Operation

FOR THOSE OF you used to programming Yamaha drum machines or sequencers, getting around on the RX7 is simple. For those of you who aren't, you'll have to learn to orient your approach to the typical Yamaha protocols: you select a main function, and then select one of several sub-functions, called "jobs" (these are all labeled clearly on the front panel, so you don't need to consult the manual constantly). At this point, data can usually be entered with either the "yes-no"/ "increment-decrement" keys or with the data entry slider, though occasionally, you will instead be asked to press the "enter" switch (which doubles as the start switch). On displays which show more than one parameter, you move the cursor around with the accent keys.

If this sounds a bit convoluted – well, it can be the first few times around. But after a surprisingly short time, most users find themselves adapting easily. Besides, once you learn the "Yamaha way" of programming these instruments, you'll be able to get around pretty much all the RX and QX instruments, because there is a certain amount of homogeneity among all their products.

Like the RX5, the RX7 allows you to program in either real time or via step entry. Events can be quantized to a

volume change commands can also be entered here, allowing you to create accelerandi, ritardandi, fade-ins, and fade-outs within your song. Various points can be specified within the song with user-named "markers," and a clever search function allows you to quickly locate any area of the song by marker name or by step number. Having completed all your song manipulations, you can then store up to 20 songs in memory, and can even create up to three "song chains," allowing you to play various songs in succession, nonstop.

Simple programming operations allow you to assign any one of the hundred ROM voices to any of the instrument keys, or to more than one instrument key. Once you've decided on the voice assignments you want for a particular application, you can then store it in any one of five user-created "drum sets". Five factory-created sets are also available in ROM. Individual voices can be extensively edited – you can, for example, adjust the pitch of any sample up and down over a ridiculous five-octave range, or you can use something called "multi-voice" mode to put a single voice over the entire top row of twelve keys and then assign a different pitch, level, and/or pan position to each key. Because there are only stereo outputs, there are extensive panning controls built into the instrument, and voices can even have changing pan positions at different repetitions within a single pattern! This allows you to set up very complex stereo images, with sounds changing position constantly – if that's the

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Editing *"Patterns can be edited extensively, with each individual event within a pattern completely adjustable in terms of level, pitch, decay, or pan position."*

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resolution of within a 32nd-note triplet, or, with quantization "off," within a 96th of a beat. You can store up to 100 patterns, with each being anywhere from 1-99 bars in length (though, obviously, you'd run out of memory long before you were able to store anything close to 100 patterns of 99 bars each). All the standard time signatures are supported, and quite a few weird ones as well (99/32, anyone?). Patterns can be edited extensively, with each individual event within a pattern completely adjustable in terms of level, pitch, decay, or pan position. A helpful "compare" mode, along with an edit recall command, make life really simple here.

Once you've got the pattern doing what you want it to, you then link patterns together into songs. Each RX7 song can contain up to 999 patterns, played in any order you like. In song edit mode, you can insert, copy, or delete patterns at will, or you can enter in repeat signs (any number of patterns can be repeated up to 99 times, and you can even "nest" repeat signs on up to ten levels). Tempo change and/or

kind of thing you're into (and I, for one, certainly am).

The RX7 also offers up to 16-note polyphony (depending upon specific voice assignments), as well as dynamic allocation, a feature not often found on drum machines. It allows you to play a sound repeatedly without incurring the abrupt cutoff you would get if that same voice were sent monophonically to an audio output. You can actually assign up to eight-note polyphony for any one voice, so that you can have that sound appear up to eight times in rapid succession – and hear it cycle through its complete envelope eight times. Thus, only playing a ninth note would cause the abrupt ending to the sound that you typically experience with a static allocation. Of course, if you really do want those "choked" sounds, you still have the option of assigning a single-note polyphony to any voice, or of using the "damp" feature, which allows you to cut off any voice in real time during the programming of a pattern.

Speaking of envelopes, the RX7 doesn't really provide much of an envelope at all ▶

World Radio History

for individual voices, just a variable decay time. On the other hand, there is an onboard LFO for each voice (an LFO on a drum machine? I love it!) – useful more for the longer samples than for the percussive sounds. The LFO signal can be used for pitch or amplitude modulation of the sample, or both, with variable depth controls. What's more, the voices on the RX7 can also respond to external MIDI pitch-bend data, as well as the onboard pitch envelope supplied for each voice.

The MIDI

THE MIDI IMPLEMENTATION of the RX7 is really quite extensive. In addition to being able to respond to pitch-bend data, it also works with external MIDI clock, song

making the RX5 much more suitable for sequencing operations than is the RX7. Ah, well, you can't have everything (where would you put it all?).

Having only stereo outputs (as opposed to individual outputs) clearly limits the amount of outboard signal processing you can do (since it will affect all voices panned to that output). However, Yamaha Giveth and Yamaha Taketh Away, so in partial compensation, the RX7 provides something called "effect" for each voice. Essentially, this is a built-in programmable digital delay (more specifically, it's a MIDI delay, meaning that it causes no signal degradation). Controls here allow you to specify the number of repeats (up to four), the delay time between repeats (between

machine. These can be loaded into RAM from the internal ROM at any time. The "Doo Wop" demo, in particular, wins the MT Award for the Funniest Drum Machine Demo of the Century, hands down. Even if you never buy an RX7, make a point of listening to this one at your nearest music store – you'll enjoy every minute of it.

The Conclusion

SO WHAT CONCLUSIONS can be reached? Clearly, the RX7 offers a number of unique and interesting features that, coupled with the overall quality of the supplied voices and moderate price, make this a very attractive instrument. In fact, the RX7 is certainly one of the best stereo output drum machines currently available. On the other hand, having only stereo outputs is a major drawback especially in professional or studio applications.

Unfortunately, this also leads to a couple of annoying design features – for example, because you no longer have a physical mixer onboard to balance the individual voices, you need to go to voice edit mode to adjust their levels. The problem is, you can't get into that mode while a pattern or song is playing back, so you can't do a real-time mix. Similarly, you can't adjust pan positions while a pattern or song is playing back. Limitations like this can really get frustrating, especially if you're used to having the luxury of individual voice outputs, where both level and pan are typically adjusted in real time at the mixing board.

Panning *"Because there are only stereo outputs, there are extensive panning controls . . . and voices can even have changing pan positions at different repetitions within a single pattern."*

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select and song position pointer, and can perform bulk dumps of not only voice and sequence data, but "setup" data (things like the MIDI configuration) as well. Each individual sample can transmit on any MIDI channel – and this is *very* useful if you're using the RX7 as a quasi-sequencer.

What's more, there's an ingenious system utilized here that allows you to custom-configure the MIDI reception of the instrument. It works like this: any incoming MIDI note number data coming in over a specific channel can either control the pitch of RX7 voices assigned to that channel, or those note numbers can control the individual RX7 voices according to their note number assignments, or reception can be turned off altogether. Any or all channels can be set for pitch control, but only one can be assigned for voice control. What this means is that, in the former instance, any RX7 voice can be played with varying pitch over a five-octave range (and with eight-note polyphony as well) from an external controller or sequencer. In the latter instance, different keys played on a keyboard controller, or different note numbers issued by a sequencer, will play different RX7 voices according to their programmed assignments. Very nice!

On the down side, it's worth pointing out that two special MIDI features of the RX5 are not available here. The first of these allowed for EG bias control of the amplitude and envelope characteristics of a voice. This made it entirely possible, for example, to expressively "play" the RX5 voices from a wind controller like Yamaha's WX7. The second RX5 MIDI feature that is missing here is the ability to selectively transmit MIDI note-off commands at specific time intervals following a voice's note-on. In other words, you could effectively assign a gate time for each voice,

10 and 500 milliseconds), the pitch difference between each repeat and the original sound (up to plus or minus six octaves!), the level of each repeat, and, perhaps most impressively, the pan position of each repeat. For those of you out there who are stereo fanatics like I am, this is a truly wonderful feature that allows you to really "image" each sound – and each sound can have entirely different effects settings – in really interesting ways.

These pre-programmed "effects" can be brought in and out in real time with a dedicated effects on-off switch, and the use of this switch during real-time recording is memorized, too. Thus, voices played with the effect on during pattern recording will be played back that way regardless of the current status of the on-off switch during playback.

As we mentioned earlier, there is only a single footswitch input on the rear panel of the instrument, but there's practically nothing that single footswitch can't do, thanks to a virtually open-ended footswitch assignment function. This allows you to quite literally link the footswitch to any single front panel · switch, without exception. In other words, you can do the boring, predictable thing and assign it to either the start or stop/continue switch, or you can get a little more adventurous. You can, for example, assign it to any instrument key, so that the voice assigned to that key will sound each time you step on the switch. Or you can assign it to the "damp" control, or the "effect" on-off switch, or the real-time or single step write switches. You can even assign it to the voice assignment job that assigns you to the voice assignment job that assigns it to the voice assignment job that . . . (talk about a hall of mirrors!)

Finally, the RX7 comes with eight demo songs that show off the capabilities of the

Another major limitation is the fact that the RX7 will not accept ROM cartridges, meaning that unless Yamaha or a third party manufacturer gets into the sound chip business (and Yamaha has never done this in the past), the 100 voices you get when you buy this machine will be the same 100 voices that will still be in the machine when either it or you eventually leaves this mortal plane. Moreover, there's a real lack of RAM space in the internal memory for the storage of voice data – an edited voice can only be saved over the original voice. True, you can always get the original unedited voice back with a simple initialization command, but the point is that you can never have the original and the edited versions available to you simultaneously.

So, is it the brass ring for the RX7 or is it the tin one? Our conclusion, sadly, is inconclusive. For the price, there's no question that the RX7 delivers excellent sound and numerous exciting features. But it does present several major limitations, so you'll need to deliberate carefully before letting those moths fly out of your wallet. ∎

World Radio History

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The new Ensoniq EPS Performance Sampler and SQ-80 Cross Wave Synthesizer with Poly-Key™ pressure sensitivity

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Playing an instrument with Poly-Key pressure is a musical treat. Each individual note responds vividly to your touch. You can control the modulation of pitch, vibrato, brightness or loudness—even the mix between two different sounds—all by varying the pressure of individual keys. So, instead of just playing a chord, you can command an entire string section. Or give horns real individuality. Or play dozens of other expressive effects you never could before.

The Ensoniq EPS — The only sampler that can play and load at the same time

It's hard to be expressive when your keyboard is silent, so the EPS lets you load sounds from the disk *while you are playing*. No other sampler—regardless of price—has this important performance feature.

Another new means of expression—Instant Patch Select—lets you choose alternate wavesamples or programs instantly in real time. With two patch select buttons located near the pitch wheel, you can instantly add expressive variations to a sound as the spirit of the moment moves you.

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In addition to expressiveness, your instrument needs sounds that can cut through a stage full of amplified instruments. The Ensoniq SQ-80 Cross Wave Synthesizer cuts like a sharp knife.

Cross wave technology involves grafting the transient attack characteristics of one sound onto the beginning of another. The SQ-80 has a total of 75 sampled and synthesized waves on board, including multi-sampled bow, pick, breath and hammer attack transients, as well as inharmonic loops and sampled and synthesized sustain waves. So you can create thousands of sounds that not only cut, but sing and soar as well.

There's also an 8-track MIDI sequencer and built-in 880K disk drive for program, sequence and MIDI system exclusive storage. Each disk can store up to 1728 different programs and 10 full sequencer or MIDI system exclusive blocks. With one disk, you can be set up and ready to play before the guitar player tunes up.

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Disbelief turned into amazement.

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Now I can get my live mixes perfect, with volumes, panning, eq, and effects all changing for each channel during the song. I can even have the entire system off in between songs, leaving just my microphone on, because that's what I programmed it to do at home!

If you're using MIDI and beginning to purchase a number of sound modules, the SPM8:2 is a must. You'll find it in your local music store—probably in the drum department."

Bo Tomlyn is an LA based studio synthesist, currently performing a series of educational seminars in music stores across the country. Bo's seminars cover all aspects of programming, interfacing and performing with today's hi-tech music equipment. Contact your local music store for details.

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Bo Tomlyn

On Stage
PINK FLOYD

Photography Rosemarie Rounseville

Pink Floyd is back on tour and our report takes you past the lasers, lights and smoke to explain how the show's music comes to life. *Text by Rick Huber.*

QUITE THE SPECTACLE, it was. Pink Floyd has always been known for elaborate shows with great sound and breathtaking lights, and the current tour is no exception. The departure of Roger Waters has not stopped the rest of the boys – David Gilmour, Rick Wright and Nick Mason have put together a first-class show.

The stage is really clean, with no visible amps or monitors. Trap doors are located in four places for custom designed lighting "droids" to pop up. At the back of the stage is the round screen that has been with Floyd for years, updated with varilights around the edge. Varilights are also hung on tracks and suspended over the stage, moving in on different players for different tunes. Computer-controlled lasers (which are run

by software designed by music software company Hybrid Arts) are also shot everywhere with meticulously positioned mirrors throughout the arena. The sound system is again in the impressive "quintaphonic" setup. The Flying Pig makes an appearance, but this time he's a boar sporting fangs and male attributes due to a rights battle with Waters.

In the past, Pink Floyd has used tapes extensively to recreate the album sounds. For this tour, they wanted to play live as much as possible. Solution: MIDI. Samples were made from the masters available and played by Jon Carin, the support keyboard player, leaving only a couple of sound effects on tape.

MIDI is also very evident on the new material. Sequences are used on the new

tunes (a click track comes from an RXII) with changes done live by Jon on Roland MC500 sequencers on cue from the band or film. Several of the tunes have extended solo or free jam sections to add to the live feel. At one point, the PA is almost shut off while Gilmour, Mason and bassist Guy Pratt play off each other.

Double drumming has always been a favorite of mine, and Nick Mason on a medium-sized Pearl kit and Garry Wallis in a cage of acoustic drums, cymbals, and Simmons pads (and lots of little things to beat on!) is a sight to see. Mason triggers several Wendel Jrs, but basically plays a straight acoustic kit with a steady strong beat. Wallis, on the other hand, is very electronic, triggering Simmons and samples.

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The current touring band also features Tim Renwick on guitar, who provided some nice soloing and strong rhythm work for Gilmour's very impressive playing, and saxophonist Scott Page, whose screaming solos threatened to steal the show. And of course what would Pink Floyd music be without some gutsy, soulful, female backup singers? Margret Taylor, Rachel Ford and Durga McBroom fulfill that role on this tour.

All in all, an impressive sight and sound. ■

■ JON CARIN runs the bridge of the Starship Floyd from his perch behind Rick Wright. Jon controls all sequences from two MC500s, chosen because of their ability to be run quickly from visual cues off the films (some computer sequencers have to think before starting, thus creating a delay). All of Jon's gear is linked together in a custom setup put together by his tech, Andy Leadbetter, with a Sycologic MIDI switcher. A Kurzweil MIDIBoard acts as a master keyboard, driving five racks of synth modules and effects, including a Roland MKS50, an MKS70, an MKS80 Super Jupiter, a D550, an ESQM, two Emax HDs, a TC2290, a REV7 and an SPX90. A DX7, a DX7IIFD and a Roland D50 round out the other available keyboards and a Roland TR727 is available for latin percussion sounds. Two DMP7s and an MV802 are used to mix everything together, sending only a stereo mix to the house. Jon is also a programmer and likes live control of his sounds – programmers for the D50, Super Jupiter, and MKS70 are on sliding drawers to allow changes during the performance. The band chose the Emax rack module with hard disks because they were the only rack-mount sampler with three-second loads. But Jon is looking for better sound quality and was planning on picking up a Kurzweil 250 with a Mac before the next leg of the tour.
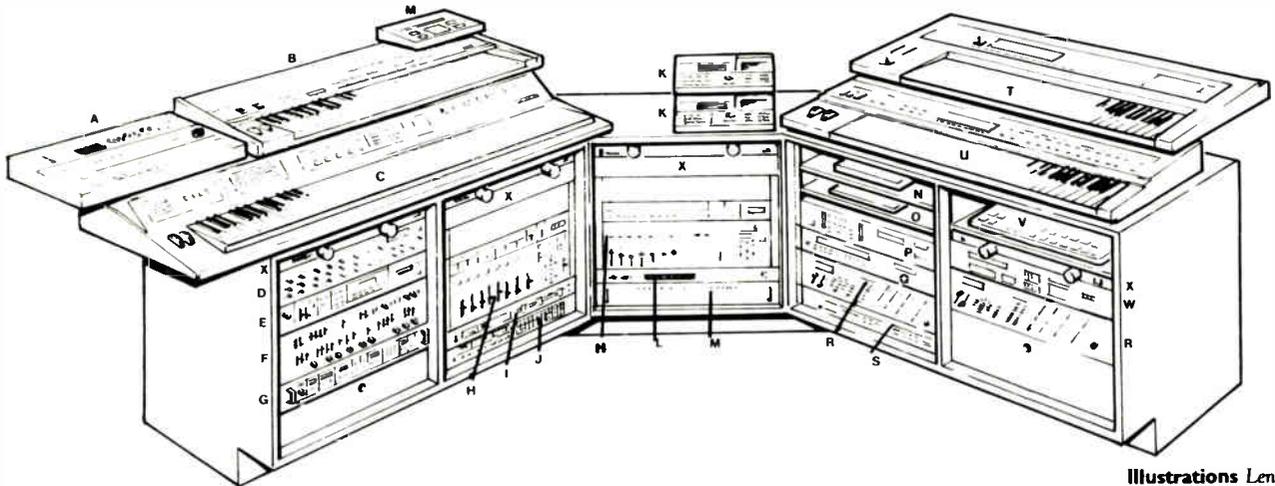
Illustrations *Len Huxter*

Key to Instruments
◀ John Carin

A	Roland TR727
B	Yamaha DX7
C	Kurzweil MIDIBoard
D	Yamaha MV802 mixer
E	Roland MKS80 module
F	Super Jupiter Programmer
G	TC Electronic 2290
H	Yamaha DMP7
I	Yamaha SPX90
J	Yamaha REV7
K	Roland MC500
L	Quark MIDISwitch
M	Sycologic Switcher
N	Roland PG800 programmer
O	Roland PG1000 programmer
P	Ensoniq ESQM
Q	Roland MKS70 module
R	E-mu Emax module
S	Roland MKS50 module
T	Roland D50
U	Yamaha DX7IIFD
V	Yamaha RXII
W	Roland D550
X	Furman Power Conditioner module
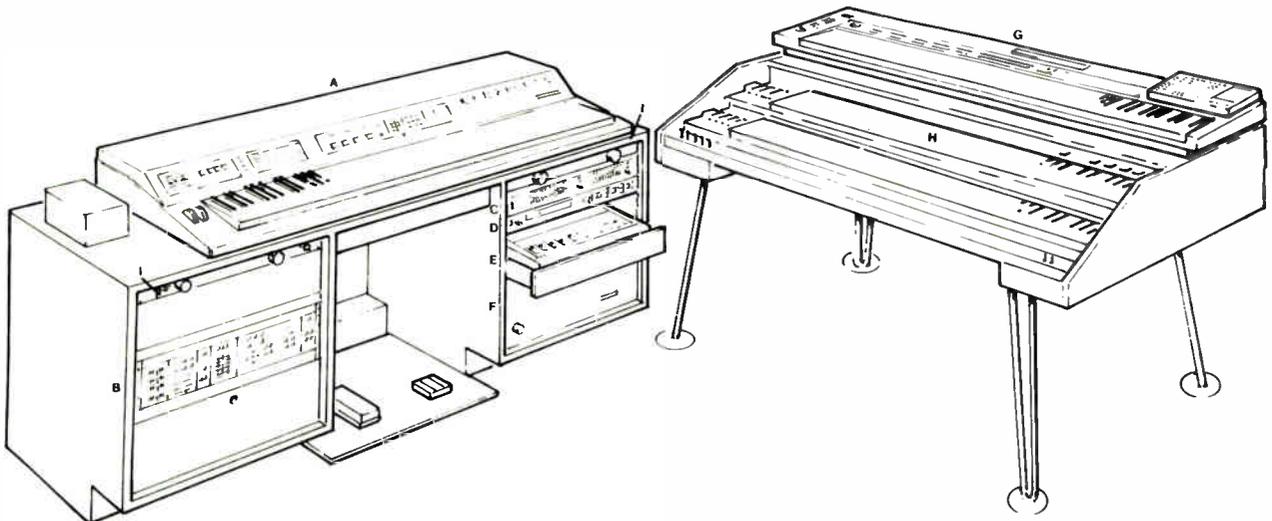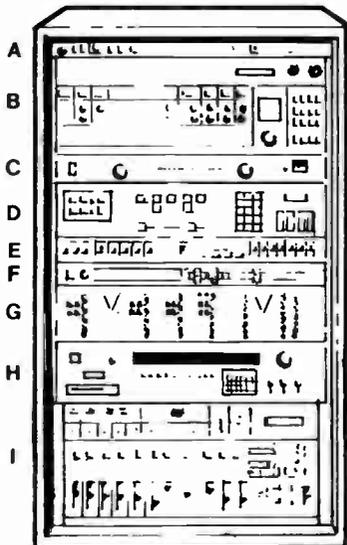
▲ Garry Wallis
A dbx 463X
B Simmons SDS7
C Furman Power Conditioner module
D Simmons MTM
E Simmons SPM8:2
F Yamaha SPX90
G Simmons SDS5
H Akai S900
I Yamaha DMP7

■ GARRY WALLIS' cage is outfitted with an impressive rack for his triggered sounds. Several Simmons pads and triggers on acoustic drums are sent through a Simmons SDS5, SDS7, MTM and an Akai S900 sampler. These are mixed through a Simmons SPM8:2 mixer and a DMP7. Wallis produced some incredible sounds with this rig during the course of the show, including some low frequency things that literally shook the entire arena. Tough stuff.

◄ Rick Wright
A Kurzweil MIDIBoard
B Kurzweil 250 module
C DOD EQ
D Yamaha SPX90
E Yamaha DMP7
F Hammond B3 Power Stabilizer
G Roland JX10 Super Jupiter and programmer
H Hammond B3
I Furman Power Conditioner module

■ RICK WRIGHT'S setup is a bit simpler, with a Kurzweil MIDIBoard driving a Kurzweil 250 rack. A Hammond B3 with a Leslie backstage, and a Roland Super JX w/programmer. All this is fed through a DOD graphic EQ and an SPX90 into a DMP7.

INSIDE ENVELOPES

Though they often seem a bit mysterious, envelope generators offer the key to programming creative, expressive sounds on any synth. We take a close look at the various types of envelopes and explain how you can use them to your advantage. *Text by Lorenz Rychner.*

PITCH, TIMBRE, AND loudness are the three basic elements of any sound. You know that. But if you try to synthesize a sound by setting a pitch, creating a timbre and setting a volume level, you'll quickly realize that there's something terribly amiss. Synthesized sounds require one more all-important element to make them interesting to listen to: changes over time. Static sounds are incredibly boring, but once you give a sound some movement in any or all of these three basic areas, then you've started your job as a synthesist.

Synths offer a number of different devices to create changes over time, but the most important of them by far are the envelope generators (EGs). You use EGs to effect these changes by defining envelope "shapes" which automatically raise and lower the levels of parameters associated with pitch, timbre and loudness. EGs allow you to have a great deal of control over these parameters and without them, the programmer couldn't do his or her job. The "better" the synth, in fact, the more features it should have that provide such control. And you don't have to spend megabucks to be in charge. There's been a revolution in this area, mostly due to new developments in the design of EGs on some *very* affordable synths. Among other factors, your choice of synthesizer should even be influenced by the number of EGs that an instrument has, as well as their configuration, and the possibilities of their assignments.

To give you an idea of the importance of envelope generators, let's take a look at a typical programming session and see how EGs fit into the total picture.

Programming a Sound

LET'S SYNTHESIZE A low-pitched brass instrument, using a sawtooth wave and three basic ADSR (Attack/Decay/Sustain/Release) EGs. You need to give each note three shapes, using an EG assigned to the amplifier for loudness, another EG at the filter for color (overtones), and a third at the oscillators for pitch.

The *loudness* is programmed at the VCA, or its eqivalent. Low brass seems to swell up in loudness at the beginning of

every note, but this is deceptive. The swell is a matter of color, to be adjusted at the filter. The attack speed of the loudness is very fast. You just don't hear much until the filter has caught up. The loudness then peaks and quickly diminishes by a fair amount. As long as the player's breath lasts, the loudness can be sustained unchanged at this diminished level. As the player stops the note, the loudness falls to silence in a very short time (but not instantly). For an ADSR, this suggests a VCA shape as shown in Figure I.
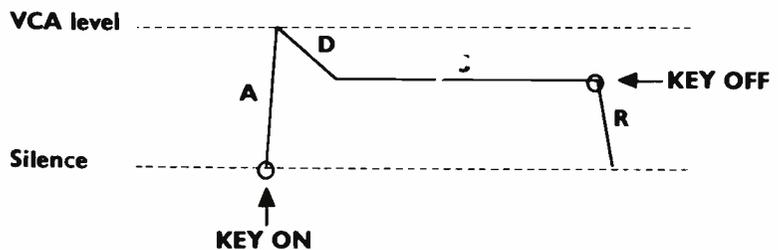
Figure I.

A = Attack rate of speed, D = Decay rate of speed, S = Sustain level, R = Release rate of speed. Note that the maximum loudness is not programmed from within the ADSR, it's a given that the loudness always peaks according to a separate parameter called VCA level or something similar. Note also that A, D, and R are only rates, like miles per hour, not fixed time values. How long a trip actually takes depends both on the rate and on the distance.

Low brass builds its *color* up from nothing. The first instance of sound has no overtones, which means it looks like a sine wave. At the peak of the attack there's an overshoot that settles down to a steady color during sustain. At the end of every note the color diminishes rapidly by losing

overtones. You must set a low cutoff in the filter to reflect that lack of overtones at the beginning of notes, and you must give the EG a lot of power to shift that cutoff to a higher level for the overshoot. Because an ADSR always returns to the starting point, the overtones are automatically taken out at the end of notes when the cutoff is lowered to its starting position. (See Figure 2.)

Note that the shape is almost identical to that of the amplifier's EG. The attack speed, though, is slightly slower. Also, you have a choice for the initial cutoff – the filter can be as open or as closed as you want it to be at the beginning (and therefore at the end) of every note. For brass, this has to be low.

Brass instruments also overshoot the pitch at the attack peak. Exactly how much depends on the musical style and on the dynamics. (This is not to say that brass players can't play in tune – please, no angry letters). The amount of pitch movement must be small, and it has to happen very fast, returning quickly to a zero level, which is the true pitch of the key you're playing. The trick is to adjust the attack so that it's fast enough to sound realistic, but not faster than the filter, to get the pitch movement to coincide with the color overshoot. (See Figure 3.)▶
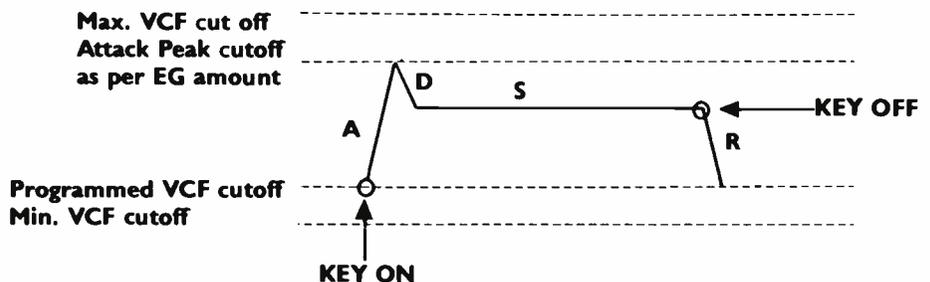
Figure 2.
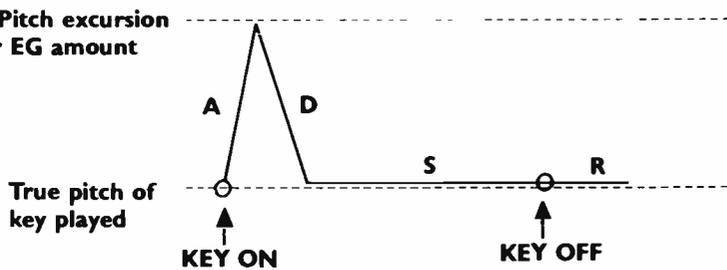
Figure 3.

▶ As you've seen, the ADSR deals with three rates of speed (Attack, Decay, Release), and one level (Sustain). But musical sounds deal with two more levels: the level before and after you play a note, and the maximum level reached during a note. Be aware of the parameters and "givens" that deal with these two levels. At the VCA, or amplifier, the level before and after a note is not adjustable, it's always silence. You set the maximum with amplifier level. At the VCF, or filter, you set the level before and after a note with filter cutoff frequency, and the maximum is the result of the filter EG amount (also called depth level or intensity). At the VCO/DCO, or oscillator, the level before and after a note is the true pitch of the key you play, and the maximum is set with the oscillator's EG amount/depth level/ intensity. (Don't forget, your synth may have different names for all of these parameters.)

Other Envelope Shapes

UP UNTIL NOW, I've been describing basic ADSR envelopes. A number of manufacturers have come up with variations to the envelope theme, however, and many of these provide even more control over the sound. On many Korg instruments, for example, you find ADBSSR EGs. The 'B' stands for *Breakpoint*. It's a level that can be set anywhere between zero and maximum. You set the speed at which it is approached with the Decay parameter. As soon as the signal reaches this Breakpoint level, it moves on towards the Sustain level, and it does that at the speed that you program for *Slope* – which is actually the first 'S' in the name.

The term *transient* is used for the type of level that the Breakpoint represents because the signal doesn't wait there (*trans* is latin for through). In our brass example, we can use this to make a double attack, by setting a low Breakpoint with a fast Slope in either the VCF or the VCA EG. This gives the impression of a slapback echo before the sustaining part of every note. Here's the new VCA shape (see Figure 4).

On recent Korg instruments the value for Decay can take on a new function. If the Breakpoint is set to maximum and Sustain to less than maximum, Decay can act as a timing value for a temporary peak

hold. High values of Decay freeze the signal at the attack peak level before the Slope value lets the signal descend to the Sustain level. The other "givens" we defined above for the ADSR are also valid for the ADBSSR EGs. With the next kind of EG that's going to be addressed, however, this changes.
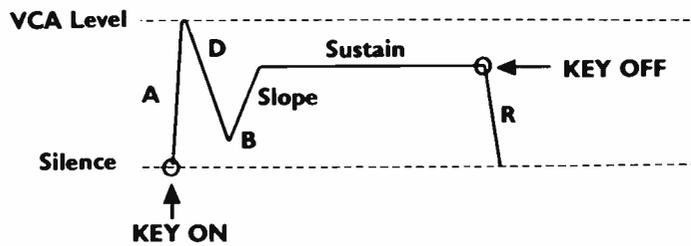
Figure 4.

Rate and Level envelopes provide an even greater amount of control and permit you to overcome one major limitation of ADSR-type envelopes. So far an important "given" has been that the first level is always the maximum. Later levels (such as Breakpoint and Sustain) could only be the same or less than the level reached during the attack. This relatively new design of EGs, however, lets you program individual levels right from the start. The Roland Alpha Juno, for example, lets you program two transient levels and one sustain level and each level can be approached at its own speed. Roland uses the term Time for their rates, but the values you set are not milliseconds; the actual time still depends on how far apart you set two levels. So with a rate and level envelope our brass sound's amplitude changes could now look like the envelope in Figure 5.

Note that there's no Level 4 available. That's still a "given," preset to zero. The Alpha Juno also has only one EG, so the loudness, color and pitch changes will have to take on the same shape if the EG modifies them all.
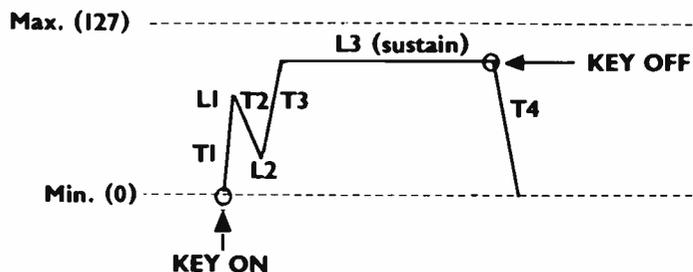
Yamaha was the first to let you program the level after Release (and therefore the level before the next note). On their DX7 instruments this is Level 4. With it, you can set a value that causes the sound to maintain a certain loudness until you pull the plug. While this sounds stupid, it's actually valuable because of the different functions that the oscillators ("operators" in FM terminology) have when acting as carriers or as modulators. Check out the factory Harpsichord sounds where Modulators shoot up to a high Level 4 to give the second pluck on key release that's so typical for that instrument. Adjustable Level 4 values are also valuable in the Pitch EG where the pitch can travel high or low after release.

Casio CZ instruments let you program seven levels and eight rates (the eighth level is preset to zero), and you pick where the sustain happens. This means that you can have several individually timed ups and downs after release, before the signal goes to zero at level 8. The instruments also have separate EGs for pitch, color and loudness. To give you an extreme idea of what you can do with these kind of envelopes, plug in and take a quick listen (if you can) to the CZ sound detailed in Figure 6. It takes about 30 seconds for the pitch, color and loudness to all settle in at their respective sustain levels, and there are still a few surprises after you release the key. Notice that it doesn't even use both oscillators nor all the EG stages.

What if you want a sound to be delayed every time you press a key? You can't just slow down the VCA attack, because that produces a swell-up. On instruments with multi-stage EGs you can program a very low L1 with a slow R1, followed by a fast R2 to a high L2, hoping that this will cause a delay. Check out the (old) DX7 factory sound 'Watergarden', where this trick (L1=0) confuses the DX brain, with amusing results. Oberheim instruments

Figure 5.

PARAMETER

LINE SELECT	MODULATION		DETUNE				VIBRATO				OCTAVE	
	RING	NOISE	+/-	OCTAVE	NOTE	FINE	WAVE	DELAY	RATE	DEPTH	+/-	RANGE
1+1	OFF	OFF	↓	0	0	25	4	99	94	39	-	1
(1.2+2.1+1)	(ON/OFF)		(+/-)	(0-3)	(0-11)	(0-60)	(1-4)	(0-99)	(0-99)	(0-99)	(+/-)	(0-1)

1

DCO 1

WAVE FORM	
FIRST	SECOND
1	0
(1-8)	(0-8)

E N V (PITCH)								
STEP	1	2	3	4	5	6	7	8
RATE	18	40	41	64	12				(0-99)
LEVEL	73	14	66	66	0				(0-99)
SUS/END				SUS	END			

DCW 1

KEY FOLLOW	
9	(0-9)

E N V (WAVE)								
STEP	1	2	3	4	5	6	7	8
RATE	30	23	30	23	37	30	52	0	(0-99)
LEVEL	99	19	99	64	99	38	93	0	(0-99)
SUS/END				SUS				END

DCA 1

KEY FOLLOW	
0	(0-9)

E N V (AMP)								
STEP	1	2	3	4	5	6	7	8
RATE	84	27	32	30	19	47	34	29	(0-99)
LEVEL	99	34	99	72	95	38	99	0	(0-99)
SUS/END			SUS					END

Figure 6.

actually have a parameter for this. Their EGs are often referred to as DADSRs – the first D stands for "Delay" – and with them you can create an envelope in which the speed of the Attack is fast, but it doesn't kick in until *after* the programmed delay time.

Other Devices

OBERHEIM INSTRUMENTS ALSO have a number of other features with which you can further manipulate envelope shapes or perform "envelope-like" modulations. *Lag* generators can change the transition of a signal from level to level, smoothing out sharp transitions in an envelope. *Ramp* generators affect the front end of sound events, similar to the Attack portion of an EG. The word "ramp" describes the shape of the change in level it creates. Finally, *Gate* – which some other manufacturers' synths have as well – is another alternative shape most often available instead of the VCA EG. It causes an instantaneous attack speed to full level, full sustain level, and instantaneous release back to zero level. It basically looks like one half of a square wave's cycle, or a trigger pulse.

Dedicated vs. Assignable EGs

NOW THAT YOU'VE seen *how* envelopes work, it's time to look at what they're working *on*. As mentioned above, EGs can be used to control all three basic elements of sound, but not all synths offer this kind of flexibility. Some instruments only have one or two EGs, and on instruments which have more, some

permit you to assign any EG to any portion of the sound chain and others have specific envelopes dedicated to each of various sound generating functions. In other words, some instruments have a filter envelope which can only be used to adjust the filter's cutoff frequency and others have things like envelopes 1-4, any of which can be used to control the filter or the amplifier or both. Let's look at some popular synths to see exactly how their envelopes are configured.

Roland's Juno 106 has a single EG – an ADSR – that can be given two jobs at once: VCF cutoff and VCA level. The VCA can also be switched to Gate, which is the only way to get two different contours for timbre and loudness. This approach was taken a step further in the Alpha Juno I. It still only has one EG, but it's no longer a simple ADSR (see Figure 5). It can be used to shape pitch, timbre, and loudness – the first two with individually selectable polarity (that is, in either a positive or negative direction). Each of these three jobs can be set to respond to keyboard velocity – Roland calls it Dynamics – and aftertouch, where harder playing is required to reach the programmed levels of pitch, timbre and loudness. The loudness contour can also be set to VCA Gate, with or without keyboard sensitivity affecting the loudness level.

Kawai's K3 and Korg's DW8000 have two dedicated EGs – ADSR on the K3, ADBSSR on the DW8000 – for VCA and VCF, with velocity and aftertouch affecting their levels. Only the DW8000 lets you

invert the VCF EG's polarity. The DW also has "Autobend," which can be set to produce a single pitch movement at the start of every note played, by approaching the pitch of the key from either above or below at a programmable speed. The effect is basically the equivalent of having a ramp generator affect the pitch of an oscillator.

Users of Roland JP and JX instruments have two partly assignable EGs (ADSR) that can do up to five jobs: VCA (fixed for EG2), VCF, OSC pitch (1&2), and OSC mix. Gate can be used at the VCA instead of EG2. If an EG is given more than one job, all the events governed by that EG have the same shape, but not necessarily the same intensity, because each job can be given a separate Envelope Amount.

Korg's Poly 800 has three dedicated ADBSSR EGs. Each of the two oscillators has its own loudness EG. The third EG controls the VCF and, if called up, the loudness contour of the Noise. Korg's DS8 goes a significant step further with five dedicated EGs. Each of the two independent oscillators has two ADSRs for the VCA and VCF, both sensitive to velocity and aftertouch. The fifth EG is an ADBSSR in control of the pitch of the (by now combined) oscillators.

Kawai's new sinewave-additive wunder-kind, the K5, abounds with fourteen dedicated multi-stage EGs. As you build your waveforms in each of the two sound sources (digital oscillator banks), you can assign harmonics singly or in clusters to four EGs. Each sound source then has its own separate EGs for pitch, timbre and loudness.

If all this EG control isn't enough for you, dig deeper and buy yourself an Oberheim Xpander. It has six voices, and each voice can have five DADSR EGs, for a total of 30 assignable EGs – not to mention Lag and Ramp Generators for additional control.

The Final Word

GET TO KNOW your synth(s) intimately. Be sure to know the EGs, their assignments, and memorize ballpark values for rates and levels. What's the maximum pitch change via pitch EG? How long does it take to get the loudness from zero to the maximum at the slowest value? Find the filter cutoff value below which there's no more sound. Set the filter envelope amount to zero and play while raising the filter cutoff from zero until you start hearing a very dull sound. Memorize that value. It may be as high as 45 on a 0-99 range, or as high as 4 on a 1-10 slider. This type of specific knowledge is invaluable if you want to be able to successfully program your instruments.

Space doesn't permit me to go into more EG tricks today, but don't let that stop you from exploring. Have fun! ∎

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PERFORM
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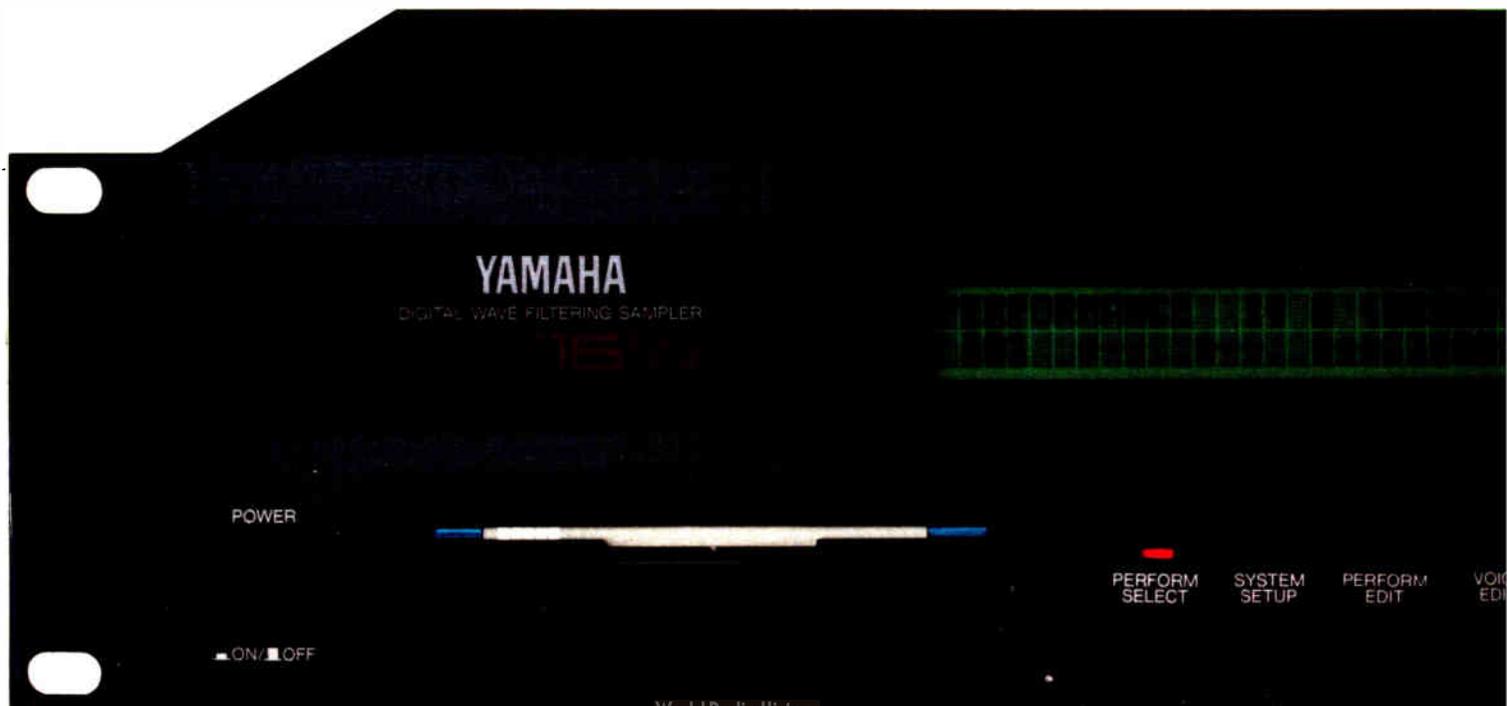
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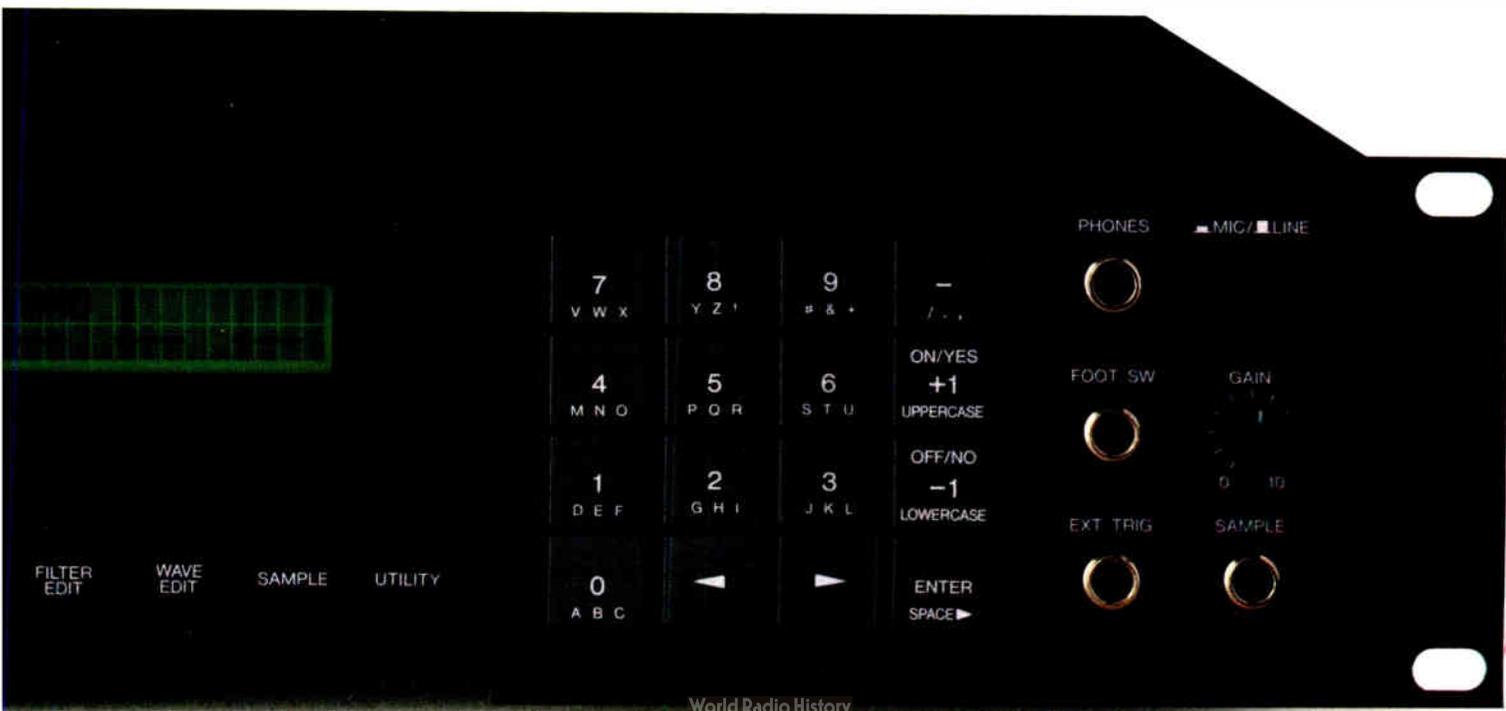
Ensoniq SQ80
Cross Wave Synthesizer

Building on the concept of the ESQI, the company's latest synth adds new features which are specifically intended for performance. *Review by Howard Massey.*

H, NOSTALGIA. BACK in the paleolithic pre-MIDI analog days, it seemed like there were dozens of new synths with the "Made in the USA" sticker being released every month. Nowadays, the release of a new synthesizer by an American manufacturer is big news. Well, the SQ80 has recently been unveiled and, with its near $2000 price sticker, Ensoniq is apparently prepared to do battle with the DX7IIs and D50s of the world. How will it fare? Only time will tell, but for now let's take an in-depth look (and listen!) at this latest entry in the Synth Wars.

Overview

ENSONIQ DESCRIBES THE SQ80 as being a "performance"-oriented instrument, and, indeed, a number of its features

are geared towards use in a live context. Perhaps its most exciting feature is its implementation of polyphonic key pressure (or aftertouch, if you prefer). This is a feature rarely found on synths in this price range and it hasn't, in fact, been seen since the Late Great Sequential T8 (that'll even rhyme if you rap it right, bro). What this means is that you can play a chord and, by pressing down on individual keys within that chord, you can have the individual notes altered in differing degrees. Here, the key pressure can be used to alter the pitch, volume, filter cutoff frequency, or to route an LFO signal to any destination. Ensoniq has designed a new type of patented 61-note keyboard to deliver this feature, and its feel is significantly different from any other synth keyboard you've encountered before.

I personally found it to be a bit clunky

and it certainly takes quite a bit of getting used to, even though its sensitivity can be adjusted. My advice here would be to definitely try the keyboard before you buy, bearing in mind that even though the SQ80 can receive and respond to polyphonic key pressure via MIDI, few if any other MIDI controllers have the capability to transmit it. (Except, of course, for the company's new EPS Sampler which has the same feature.) In other words, if you don't care for the keyboard action, you're probably stuck with either using it anyway or foregoing the poly key pressure feature altogether.

The front panel is fairly sleek and surprisingly uncluttered, despite its 51 switches, two sliders, and two-line, eighty-character fluorescent display. Except for the colors, it's basically the same as the ESQI. In fact, the entire instrument is

basically a souped-up ESQI. There's also a cartridge slot available and a built-in 3 ½" disk drive for the storage of onboard and external MIDI SysEx data (into a sizeable 64K buffer). No real-time access of data from disk is available; like the Yamaha DX7IIFD, the drive is used for storage purposes only.

Hidden away behind all the switches and displays is an eight-track sequencer, similar in many respects to the one on the ESQI. But here, of course, sequence data can quickly and easily be saved to and loaded from disk. The rear panel offers stereo audio outputs (the left output doubles as a headphone jack, which I think is a pretty poor design); MIDI In, Out, and Thru; a control voltage input (boy, I haven't seen one of those for a long time!); a sustain pedal input; a sequencer start/stop footswitch input; and tape in and out for sync and data storage. (Though why you'd ever want to dump to tape with an onboard disk drive and cartridge slot is baffling. I guess you could use it to load data tapes for the ESQI, which will work with the SQ80.)

Voice Architecture

THE SQ80 IS described as being a "cross wave" synthesizer, but this description of the process used is actually a bit unclear. Like the ESQI (with which it is upwardly compatible; you can play back ESQI-created voices and sequences on the SQ80), the basic voice architecture is eight-voice multitimbral, with three digital oscillators per voice, plus a single four-pole lowpass filter, along with three LFOs and four envelope generators per voice. Each digital oscillator offers a choice of 75 different waveshapes, many of which originated as samples of the attacks of assorted string bows, mallets, scrapes, plucks, boings, bongs, and bells. If this sounds a lot like the basic philosophy behind the D50, you're right, though I found the samples used by the SQ80 to be far less pleasing to the ear than those offered by the D50. Digital representations of standard analog waveshapes are also available, as well as a few looped samples, and several samples of drum sounds. Apart from a ring modulator-like amplitude modulation feature (which you probably won't use very much in creating most musical patches), the overall system here is clearly standard subtractive synthesis with digital wavetable oscillators – hardly a technological breakthrough.

Having said that, there are a number of unusual and helpful features present here. The SQ80 operates in either whole, layered, or split mode, or you can even split and layer your voices. The maximum total polyphony is eight voices, though this is reduced to four in layered mode. Beyond this, the instrument utilizes dynamic voice allocation, so that any number of voices (up to the maximum of either eight or four) can come from either patch. Another very nice by-product of this feature is the way that you can access different voices in succession from a bank of ten (there are four banks of ten voices each in both internal and cartridge memory, and the cartridge holds two "master banks" – doubling its capacity). If you select a voice with a long release time, for example, and play a note and then select and play a different voice from the bank menu, the first voice will continue fading away. Unlike most other synths, the process of calling up a new patch will not cause a currently sounding voice to abruptly cut off. I could see this as being extremely helpful in live performance.

Programming

BASIC PROGRAMMING operations are identical to the ESQI – if you're familiar with that – and are very similar to those offered by the Oberheim Xpander and Matrix instruments – if you know them. A number of "soft" switches surround the display, which shows the various editable parameters available on the page you have selected (you choose from among the different pages with dedicated switches on the right-hand side of the instrument). The chosen parameter is underlined, and you can then increment-decrement the values ▶

R
E
V
I
E
W

► or use the data entry slider for rapid changes. When a "yes-no" answer is required, this will be shown in the display and you simply reply by pressing the associated soft switch. In general, I found programming the SQ80 to be a simple and fast operation; the instrument provides a very "user-friendly" human interface, particularly if you're already familiar with general subtractive synthesis routings and/or the Oberheim way of doing things.

Like the Xpander and Matrix instruments, multiple modulations are available for most parameters (including stereo panning), with no less than fifteen modulation sources available for each of the dozen or so destinations. Several of these sources (like the LFOs) have modulation inputs as well, so you can chain together several devices to set up interesting and complex effects. The SQ80 envelope generators are also fairly unique, with four levels and constant time changes between them. Thus, the rate of change will speed up or slow down according to the difference between two adjacent levels, so as to always arrive at a set time. I found it annoying that you can't just directly enter in time values here; instead, you need to enter in a number from 0 to 63, and then look up the associated absolute time value in a chart in the manual. On the other hand, any of the level values can be either positive or negative; this means that you can, for example, use a single envelope to both raise *and* lower the filter cutoff frequency from its set value. This really makes a lot more sense than the more usual envelope inversion routine offered by most other synths. The envelopes can also be rate scaled or linked to note velocity with either linear or exponential response (unlike the straight linear response on the ESQI). A provision is also available for a kind of "second release" for a simulated reverb effect. It's cheating, but it's reasonably effective. It's also worth pointing out that, unlike the D50, for example, the SQ80 offers no onboard signal processing beyond this simulated reverb.

The SQ80 does have polyphonic key pressure, standard pitch-bend and modulation wheels, a sustain footswitch, and a CV input for a pedal that can control volume or modulation, but for an instrument that touts itself as being performance-oriented this really isn't that much. You can quickly access numerous voices and you can easily pre-program in various modulation routings but there aren't as many real-time controls as say, on the DX7II instruments.

The Sequencer

THE "OTHER HALF" of the SQ80 is its onboard eight-track sequencer. In fact, well over half of the 206-page owner's manual is devoted to its operation – which should tell you one of two things: either

it's incredibly full-featured or it's just plain confusing. Well, the bad news is that the sequencer really isn't all that full-featured. What you have here is a plain vanilla sequencer with all the usual append, insert, delete, quantize, and event edit functions. Tracks can be recorded in real or step time and can be copied, merged, or deleted. Each polyphonic track can transmit and receive over a different MIDI channel or can drive a different SQ80 synth voice. This is where the multitimbral capabilities of the SQ80 come through, and its dynamic voice allocation scheme makes this work really well. Real-time punch-in is supported, though punch-out isn't; you'll always need to record right through to the end of the track.

One nice feature here is the ability to initiate the punch-in point directly from a key on, so that the erasing of previous data and recording of new data doesn't occur until you actually start playing. This effectively ties in the punch process to your own musical sensibilities – a nice idea. Another very handy option allows you to preview an overdub or a quantization process before irrevocably losing the original track. The presence of a track edit buffer ensures that you can always recall the original take if you decide that the new one isn't any improvement – or that you can restore the original, unquantized track if the quantization ruined it. I could see this as being very helpful in many situations. But the actual layout of the sequencing operations is really not as straightforward as it could be – even if you've had extensive experience with dedicated or onboard sequencers. I found the entire process not nearly as user-friendly as is the process of voice creation and editing.

The MIDI implementation of the SQ80 is excellent, with Poly, Omni, Mono, and something called "Multi" mode (used for MIDI-channel-to-track-assignments in the sequencer) all supported. Additionally, you can selectively enable the reception of various kinds of MIDI data, and there's even an overflow mode that allows you to slave multiple SQ80s together for sixteen or more voices. Global controllers can be assigned while in mono mode, making this instrument suitable for use with a guitar controller, for example, and both MIDI clock (as well as standard tape sync), song position pointer, and song select are recognized by the sequencer.

Ergonomically, the SQ80 is well-designed and certainly appears to be road-worthy, although my limited time with the instrument can't confirm this. But you have to be suspicious of an instrument which offers a soft reset routine (somewhat unfortunately called "re-initialization") which wipes the entire contents of the internal memory (which normally is battery-backed). Sure, good backing-up habits are important, and the presence of

an on-board disk drive makes things even easier, but it really seems like overkill to have to always wipe the internal memory whenever a reset is required.

The Sounds

WHAT DOES THE SQ80 sound like, you ask? This is clearly a subjective area, but let's plunge right in and get it over with. It basically sounds almost exactly like the ESQI, but considering the competition it's facing in this price range, that's not good enough.

To be brutally frank, I found the actual sound of the instrument to be disappointing. There is little clarity in the low end, which is either non-existent or muddy, and the top end is gravelly and somewhat irritating at times. The raw materials – the digital wavesamples themselves – are largely undistinguished, and the more standard analog-like waveshapes lack any real punch or definition. The onboard drum sounds are also nothing to write home about. It also seems strange that Ensoniq developed a new high-quality sound chip but only used it in their new sampler and not the SQ80.

Conclusions

THE BOTTOM LINE is that in the Ensoniq SQ80, you have an instrument which offers many interesting features and is quite easy to program, but has a somewhat questionable sound. The poly key pressure is a nice feature if you like the feel of the keyboard and even if you don't like the sound of the instrument you could use the SQ80 as a master controller for your system. In fact, doing so would allow you to take advantage of the on-board sequencer – though it is a bit limited – and the storage capabilities of the disk drive.

But if you're purely interested in sound quality, then the SQ80 can't match what the competition has. And in my opinion, the three most important factors to consider when buying a synth are: first, sound; second, sound; and third, sound. The most phenomenally feature-laden instrument won't be worth a lot to you in the long run if it doesn't sound great.

The SQ80 offers a multitude of new features, many of which are useful and creatively stimulating. And if you're interested in an all-in-one system with disk-drive storage, then it may be perfect for you. But in the end, it seems as though there's room for a lot of improvement in the sonic department, so my advice is simple: listen before you buy. ∎

FIRE IN THE HOLD

With thirteen years and twelve studio albums, Rush is in the midst of yet another highly successful national tour. Drummer/lyricist Neil Peart and guitarist/inventor Alex Lifeson reveal the technology behind their setups and their shows. *Interview by Deborah Parisi.*

HOLD YOUR FIRE. It takes a really good poet to give new meaning to an old saying. Although poetry – or, to avoid the austere gazes of the literary critics, *popular* poetry – relies on images that we can all relate to and understand, using a cliché to convey a

central message is usually disastrous. It smacks of sell-out, burn-out and blow-out. Not a good idea.

But few critics are accusing Rush of any of the -outs these days. Their musical growth has progressed from screeching heavy metal through classification as "a

poor man's Yes" to a brand of technologically-assisted rock which is truly their own. Live performances have become increasingly polished and (not unexpectedly) glittery, with the production quality of their albums following suit. And the lyrics of Neil Peart have gained new

insight, moving away from fantasy, futurism and science fiction to explore the primitive, the primal.

"I guess it happened by accidental design," Peart says of the poetic theme of *Hold Your Fire* (Polygram). "I realized that without really consciously thinking about it, I was into the world of instinct and the world of temperament, and subconscious kinds of things – Jungian things, certainly. And the only thing I could do was to go with it. It seemed to be where my muses wanted to take me. So the rest of the album indeed did follow that, and I broke

sampling had come of age and it was a tool that I really wanted to have and could no longer resist, I went to Jim Burgess at Saved by Technology [a Canadian company specializing in MIDI equipment and system applications] and said, 'Here's what I want to do, and here's what I don't want to do.' And he recommended a setup and worked with me a lot to get the gear and the library of samples from my older records. It's good to have someone like that to steer you in the right way."

Burgess "steered" Peart into a system which incorporates samplers, MIDI

with a custom-made A/B switcher from Hi-Tech Music Systems so that patchbay #2 could take over if patchbay #1 went down.

"The other important part of my system is a KAT controller," says Peart, "and I use that for all keyboard percussion parts plus another triggering source, because it has its own Akai. So sometimes when all the other pads are full, I'll put the sound I need on the keyboard so I can just whack it and the sound comes out. In the past I used to have a glockenspiel over there, and a set of the little tuned cymbals that are called crotales (a Turkish bell cymbal), but those kind of things are enormously difficult to mic, and for the amount of space they take up they only offer you one sound.

"On one song called 'Mission'," he explains, "there's a marimba solo. On the record I used a syncopated snare drum matching in unison with the marimba part, and Geddy plays a bass part also in unison with that. So what I did for the tour was assign the sounds of both the marimba and the snare drum for that particular song, so that I can have both of them as I play the keyboard part. Within the same unit I have other straight marimba parts and glockenspiel parts and various other sounds, so it's been very valuable. I wasn't sure at first if I would use it live or how successful and reliable it would be. But since it has been good, now it has my faith.

"That's essentially it," he says, "from pads and triggers into a MIDI controller, and of course the KAT is a MIDI controller by itself. I use a couple of Alesis Midiverbs just to make these electronic sounds sound more like my acoustic drums do in the environment." ▶

Peart *"Sampling sequences lets us make our sound as big as possible and lets us reproduce the records as closely as possible. It's still a tremendous challenge in trying to make it work."*

it down into various subthemes beneath that common umbrella of instinct and temperament."

Although new songs like 'Force Ten' and 'Second Nature' display a social consciousness reminiscent of the Vietnam era, Peart doesn't see a general trend of '80s lyricists in that direction. "Every once in a while I get hopeful, and I think I see that trend. And then it goes, and a whole bunch of pap comes out, and people are just as happy with that. It became its own fashion, you know, throughout the whole Live Aid thing. The whole industry was like standing up on this pillar and saying, 'Look at me, look at me! I'm generous!' And it died . . . possibly because of the posturing that went with it and its essential hypocrisy. Like the punk movement, it had to collapse from within because you had a lot of people saying things that their day-to-day lives and the rest of their work just didn't back up. So I think that big wave – in spite of Roger Waters' hopefulness – I don't think the tide is turning. I just think it was like a tidal wave that came in and went out again.

"It is an old, old story, that goes back to politics, to baby kissing. It's just trying to make yourself look really good to the people down there and have them think, 'Gee, here's a regular guy. He's up there, but he's got a big heart.'"

Peart is just as candid in discussing his stance towards technology. "I'm not a pioneer by any means. I sort of take the Rolls Royce attitude of letting other people pioneer things and prove them and then adopt them – like Rolls Royce uses General Motors power steering because they make the best power steering. You don't have to pioneer if somebody else does it. You can still be just behind the leading edge but have the advantage of things that are reliable. And you avoid the trendy aspect of things like Syndrums, where in the early days every beer commercial had that sound on it, and you avoid having to wince about your past.

"When I finally figured that digital

controllers, and pads. "All the pads are assigned through a Yamaha MIDI controller, the PMC1, into an Akai sampler, and that's triggered by Simmons pads and also by a footpedal. My left foot bounces around from hi-hat to bass drum pedal to triggering pedal – it does a lot of work down there," he laughs. "I also have a small home-made trigger which we affectionately call 'Sydney,' which is virtually a miniature Simmons pad. It's octagonal, but it's only about 3" across, and it allows me to trigger sounds from my front kit without having to take up the space that any of the proprietary pads require. It's something that we came up with ourselves – shock-mounted and very sophisticated now, but strictly a unique item."

According to Burgess, designing the system that the band has on the road was quite a challenge. "One of the problems we had was that the band didn't want to be restricted by playing to a click-track, so a hardware sequencer wouldn't really work," he says. The alternative was to sample sequences that had been used in the studio so that band members could trigger two to four bars of repeating phrases with the tap of a foot or key.

Geddy Lee, the co-writer, bassist, and keyboard player, uses an astonishing number of keyboards and modules for his performance. On stage, he has a PPG, Prophet VS, D50 and a Yamaha KX76 that are being used as controllers. In addition, he uses Korg MIDI bass pedals and Taurus pedals to send program change information. In "the pit," he has seven S900s, two Roland D550s, one D50 and a DX7 with an E! mod, along with four of Intelligent Music's MIDI Mappers. Manning the pit is Jack Secret (an alias), whose job requires loading close to 100 disks into the various samplers at the appropriate moments. More than minor pressure, there; one wrong disk and a song could be ruined.

All of the controllers run into two JL Cooper MIDI patchbays, which are set up in a parallel configuration and modified

Peart is also an enthusiastic endorser of the new Zildjian cymbal mics. "I'm very happy with those. Again, as I described in respect of glockenspiel and crotales and wind chimes and that, all that stuff is so hard to capture in an arena. I was always haranguing our soundman with, 'Why can't I hear that little cymbal?' So now, combining those individual Zildjian mics plus the overheads, you get both. You get the overall picture of the air around the cymbals, but you also get a good individual capturing of that sound."

The Akai S900 has also become a trusted tool on the tour, bringing Peart's rhythms into the digital domain. "I have done quite a bit of sampling myself. In fact, on one of the songs – ironically I'm using Ludwig drums now – but I have samples of my previous Tama set. I took those samples and assigned them to the electronic drums for one of the older songs. So I'm playing an old song with an old drum sound. It's great.

"On one song on the album called 'Tai Shan,' I have an antique Chinese drum which is far too fragile and valuable to think about using live," he continues, "and I brought it into our rehearsal studio and sampled it. I have a number of antique, especially Oriental and African, musical instruments that the only way I can use them is to sample them. So it gives you all that freedom. That's what I like the most."

Using samplers in place of sequencers allows the band to have a greater degree of control over the triggering of sounds. "It's a blessing for a three-piece band and to me doesn't carry the same moral stigma as using tapes for backing vocals or rhythm

tracks," Peart says. "You have to work it out yourself, and program it yourself, and trigger it yourself. I find it less of a crutch and more of a challenge. When a loop comes in and out of a song, like on the song called 'Prime Mover,' right from the first beat I have to be so locked into the tempo that every time it comes along it fits. It lets us make our sound as big as possible and lets us reproduce the records as closely as possible, and is still a tremendous challenge in trying to make it work."

Peart is one of many who compares sampling other artists to stealing. "It's a very naughty moral question," he says, "and I'm engaged in a struggle with myself because in my drum solo I do use some samples that are like Count Basie Big Band shots. The controversy is strongest, of course, when you lift, say, a snare drum sound off a record and use it as a snare drum sound. A drummer learns over the course of many, many years how to tune that drum, and the engineer learns after many, many years how to record it to get that sound. The real pros . . . like Andy Richards who did keyboard work on our last two albums, he has classical things; but the way he uses them, they're all twisted and bent out of shape, and they bear no relation to their sources. So it does become a creative thing, and it's a long, long way from robbery.

"In a recording context," he continues, "I think it would give me more serious moral qualms. Live performance is frivolous enough that you don't need to take it that seriously. I allow myself the indulgence. Plus it's kind of a long-time ambition to play with Count Basie," he laughs, "so this lets me do it every night."

Peart has never used a drum machine for recording, but is thankful for its use as a songwriting tool. "Because of the fact that we work separately, where I'm working on lyrics and Geddy and Alex are working on music, it does all my hack work for me of keeping a beat while they work things out.

And plus," he adds, laughing (albeit a trifle rudely), "Alex has a very extraordinary sense of rhythm which doesn't bear any relation to anything a drummer would do, so I have actually gotten some good ideas from strange patterns that he's programmed into the drum machine. It's sort of a humorous source of inspiration.

"It's the same as the way Geddy uses the Macintosh and the music software for that," he explains, "because he doesn't consider himself a keyboard player. It's just part of the load that he bears to contribute to the overall texture of our band. And consequently, he can work out all the parts he wants to play and then just say, 'Hey, Mac, play these!' It's just like having another member to do it, but you don't have to have the interpersonal chemistry changes of having a fourth member in the band."

ALEX LIFESON FIRST got together with Geddy Lee in 1969, five years before Peart joined them to create Rush. Although he is well known for his slashing, screaming, heavy metallic guitar, recent albums have revealed a rhythmic and melodic sensibility which transcends the abilities of most of his peers. Even a casual stroll down the musical trail left by the band over the past 12 years displays his and Lee's growth as the writers for Rush's tunes.

Writing the music for an album typically takes about four months of intense work. On *Hold Your Fire*, they started working together at Elora Sound near Toronto, which provided a quiet rural spot on a farm – the studio is actually located in a barn. "We set up the Teac 388, a small eight-track unit I have, in the studio. We had all our gear set up there, but we picked a little corner, put some nice lighting in, set up the 388, and plugged our direct bass and

keyboards in," Lifeson explains. "A typical day would be up at around 10:30, breakfast together, discuss what was happening or whatever, and then Geddy and I would start working around noon in the studio. Neil would stay in the house working on lyrics. Then he would come over about 5:00 and we'd spend an hour discussing arrangements, lyrics, and Neil would give us a critique on the musical end of what we'd come up with. Then we'd break for dinner, and after dinner we'd work together as a band on the songs. It developed to the point where after about a month, we could start recording into the 24-track and then start refining it. Working like this, we're totally prepared before we even go into the studio. We don't have to spend additional

Roland DEP5 and a Yamaha SPX90 as outboard gear, and I'm using a Yamaha MIDI programmer, the MFC1. That lets me scroll through my programs, and all of my equipment is MIDI. That's especially handy with the SPX90 because you can type in the name of the song or whatever information you want in the display, so for the 20 songs that I do, I just type them in and scroll them through. That's really important these days, because it's very, very busy on stage for us.

"It's always been a rule with us that we wouldn't do anything in the studio that we couldn't do live," he says, "and that was fine in the past. But with *Power Windows* [Rush's last album], we thought we'd expand . . . to make the record for the

thing. Why don't I just run it in my catalog and maybe we can sell a few of them.' He sort of laughed about it, and I said, 'Sure, why not?' I think we sold three or four hundred of them," he laughs.

"We had to think of a name, so we decided to call it the Omega Concern," he continues. "'The last word in guitar stands.' Then I designed the Omega Lyric Stand, which lets you pin the lyrics up and turn all the lights off in the studio, still giving a nice warm light. It's a rear projection light, and all you see is the lyrics on this opaque screen, and it's nice and moody. But it's kind of a joke," he admits. Lowering his voice to simulate a DJ's, he recites, "'Whenever you need something that no one else has, Omega will be there.' Or 'At Omega, we're concerned.' Our motto is, 'We have what you need. You have what we want.'" Hey; when you're hot, you're hot.

The liner notes on *Hold Your Fire* offer a fairly standard list of thanks, but with the unexpected offerings of gratitude to Patsy Cline and "all cowboys everywhere." What? A country influence on a prog rock band? "We just go through one crazy phase to the next crazy phase," Lifeson says. "When we went into the studio, they had a satellite dish. And late afternoon, they had *The Big Valley*, *The Rifleman*, and *Bonanza* all in a row. You know – Miss Barbara Stanwyck. So somebody went out and bought a bunch of cowboy hats, and we'd all sit around," he says, going into a heavy western drawl, "talking like this. We decided when we went to England we'd take all this stuff with us. So there we were in England, in a 600-year-old house, and we'd walk through with spurs, satin cowboy shirts, cowboy hats, talking to the English housekeepers with 'Thank you very much for the vittles. We've got to be moving right along here.' And they thought we were crazy . . . this lasted for about three weeks. We'd play Patsy Cline every morning, and Ben Cartwright singing 'Ringo.' Do you remember that song?" Unfortunately, I do.

But all joking aside, Rush is intent on playing the kind of music they see as their own. Even if it means traveling to strange locations, learning new technologies or doing other sorts of strange things, the bottom line is holding the fire and staying true to their own musical ideals. It's an idea that they hope other young musicians will share. As Peart relates, "What matters to me is playing the music I like. So whether I make a living at it or whether I have to do another job to make my living, the point of honor is to keep the music good and pure. Straight from day one, young musicians have to make that decision in their own minds. And then they have to decide how hard they're willing to fight for it."

Rush has made the decision, fought the fight and won the battle. For them, the fire burns on. ∎

Lifeson *"It's always been a rule with us that we wouldn't do anything in the studio that we couldn't do live. Fortunately, technology is at a state right now where it's a lot simpler and much more convenient."*

concentration while we're recording on rewriting something. It's very precise."

From Canada the band moved through four studios: two in England, one at Montserat, and one in Paris. "We went to the Manor in Oxfordshire because we got good results there with the drums," Lifeson says. "It's a really good room, all stone and wood, and the ceiling is about 30' high in the drum room. There are movable panels on the ceiling that you can reposition to get it to sound like you want.

"After that," he continues, "we just went to places we wanted to go. We thought we'd take a chance, and it worked out great. The important thing is to keep yourself up and doing it in a different environment. The bottom line, if you're concerned about economics, is that it's not going to cost you any more to go into one studio for four months than it is to travel around to try other studios. And prices are usually a little bit less outside of North America – depending on the fluctuating exchange rates," he laughs. "England was a bargain because at the time the dollar was very strong, so the money spent on flying over there didn't work out to any more."

Recreating an album for live presentation is always a tremendous challenge, and Lifeson uses a wide array of gear to assist in guaranteeing a slick, professional show. "I just started using TC Electronics on this tour," he says, "and I have a couple of 2290s and 1210s. I really like them because you can really do a lot of things. You can adjust the panning any way you want, you can set the dynamic control, you can reverse it so that you only have repeats when you're playing. And signal-to-noise is very quiet. With all the additional software you can get for it, you can really utilize the sampler. It's really quite an amazing unit."

Lifeson continues through the list of gear being used on the tour. "I also have a

record's sake. That was fine in the studio, but then it comes time for you to tour and you go, 'What did I do?' Fortunately, technology is at a state right now where it's a lot simpler and much more convenient, I think. That's probably the biggest thing that the advanced technology has going for it . . . quality in production."

While Lifeson is obviously a champion of many of the benefits of technology, he is not nearly so warm on guitar synths. "I never really gave it much time, though," he admits. "I've tried a few out, but don't feel comfortable with it. I think you have to approach it as a different instrument – it's not a guitar, really, it's something else. I spent a few weeks with the Stepp, and I thought, 'Well, yeah, that's kind of neat,' but for the kind of money that it costs, I couldn't . . . I thought it was really crazy. But it requires a different style of playing and technique. I'd rather fool around with the guitar and effects.

"When you look at how they've developed keyboard synths," he says, "you get a better picture. Geddy's got this Yamaha – he started taking piano lessons this summer, so he lugs it around – but the touch on it is just like the piano. You know, like a grand piano would be. They've developed that, and it sounds fairly close to a real piano sound. They've developed it so that the touch is the same. But they haven't done that with guitar synths."

Lifeson is also a bit of an inventor ("An inventor by trade," Peart laughs. "You're talking to a musical scientist there."), but it happened almost by accident. "A few years ago I was using both electric and acoustic guitars on stage," he says, "but I wanted to make the change quick, and I couldn't find a decent guitar stand. So I designed one and had it built. A friend of mine owns a music store just outside of Washington DC, and he said, 'Hey, this is a pretty good

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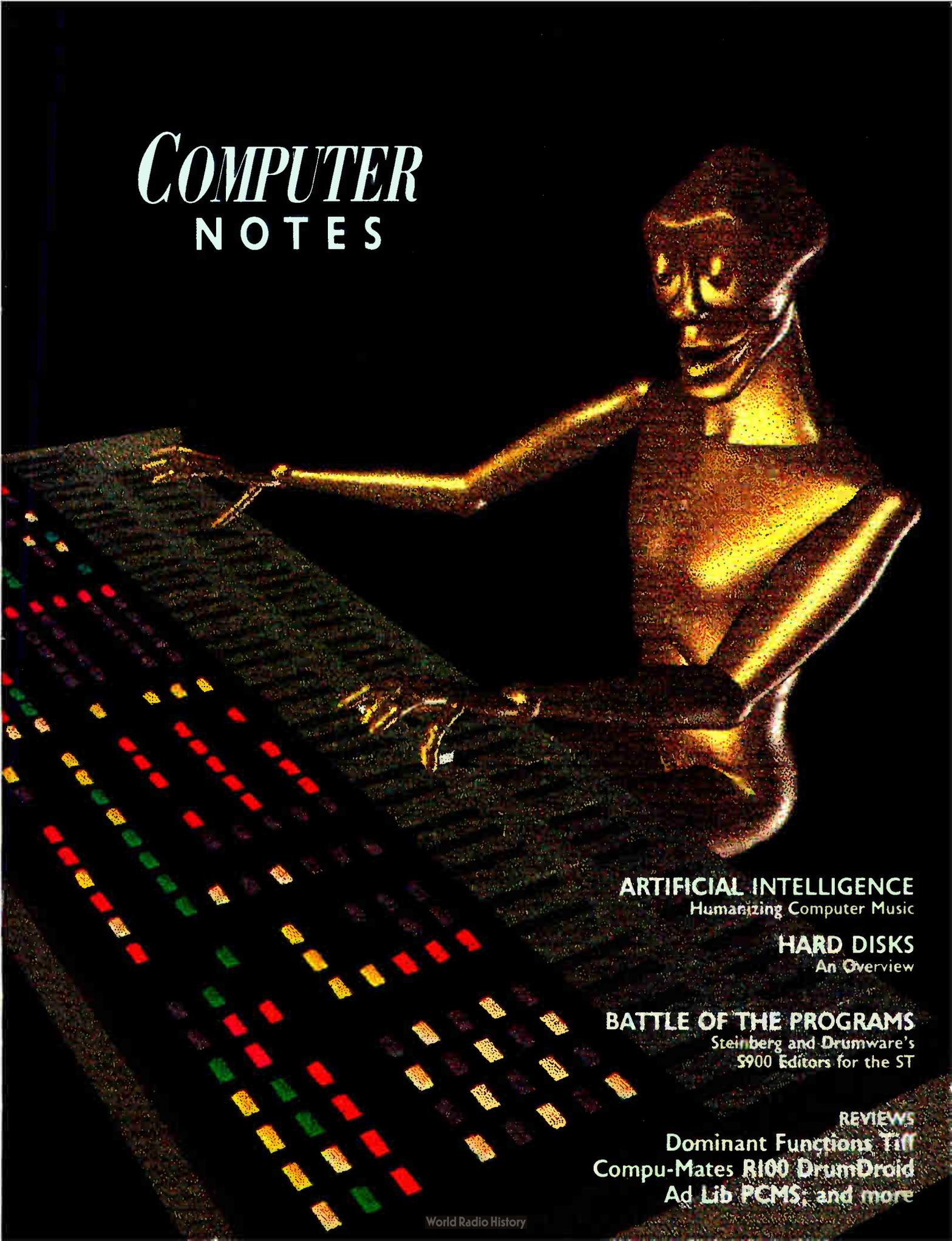
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HARD DISKS
An Overview

BATTLE OF THE PROGRAMS
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REVIEWS
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COMPUTER NEWSDESK

FOR YOUR MAC ONLY

It seems as though a month can't go by without at least one regional music equipment expo taking place somewhere. For the public, these shows are a welcome opportunity to check out a large cross-section of the current assortment of musical instruments by many manufacturers, with demonstrations by product specialists rather than by retail sales people. Still, it can be hard to take it all in with the time allotted. As the three-day NAMM shows are only open to manufacturers and music dealers, and as the range of instruments on display includes acoustic pianos, sax reeds and high-end digital recording systems, there must be a better way of getting *public* exposure for the latest computer-based musical tools.

Cue the **MacMusic Fest 1.0** held recently at Paramount Studios in Hollywood. The Fest,
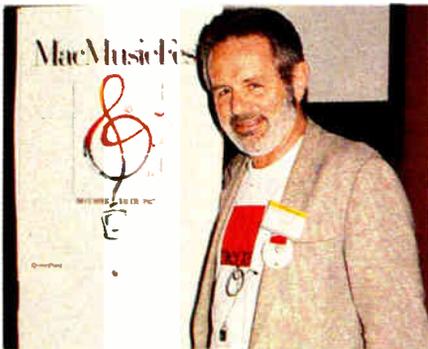
which took place on December 5, 1987, was an excellent opportunity for musicians and engineers to see, in one place, dozens of Macintosh-based products demonstrated by factory product specialists. Above all, it was the first show I've ever attended where every software package on display supported one specific family of computers, which made for a very well-focused show, indeed.

Among those manufacturers represented at the show were Digital Music Services with their DMP7 Pro support software for the Yamaha DMP7 (as were Yamaha themselves), OpCode, Digidesign, Optical Media, Beaverton Digital, and IMS (with the latest version of Dyaxis). Also making a debut was HMSL-Mills with the Heirarchical Music Specification Language, a sophisticated algorithmic music composition language which they had running on the Mac and the Amiga (curiously). JL Cooper showed off the new Mix Mate all-in-one eight-track mix automation package for under $1000. Kurzweil demonstrated the new 1000-series, while Fairlight and New England Digital each demonstrated the virtues of their respective professional music production systems. Several manufacturers of film scripting software packages had a chance to display their wares alongside music software, which added a distinct professional element to the show.
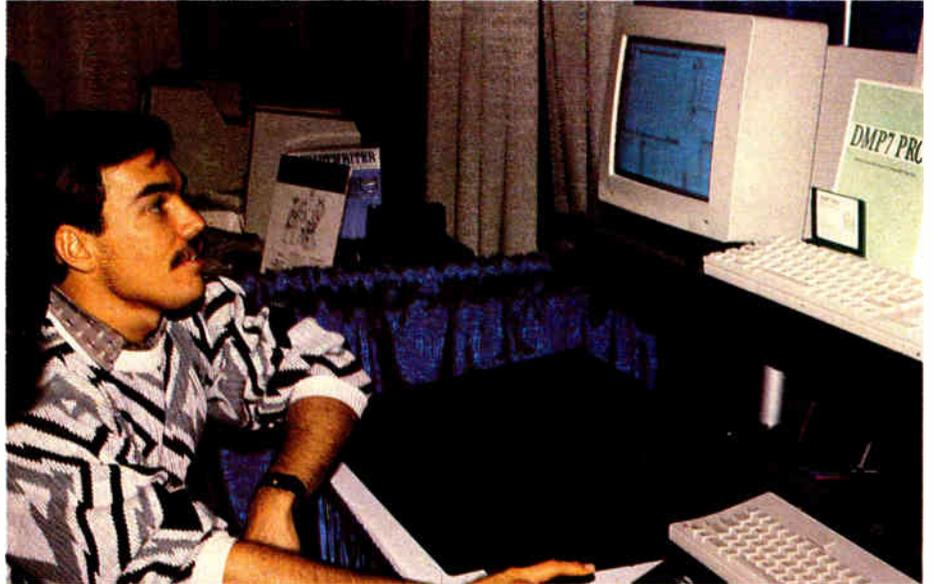
A new users group for professional composers, MEGA (Macintosh Entertainment Guild of America) made its debut with promises of a new computer network and forum for discussion. (If you're interested in finding out more information about MEGA, contact Filmsonix at (213) 653-0240.) It will be interesting to see how this one develops.

Apple, of course, was on hand to show off the virtues of the Macintosh family, Multifinder, and Hypercard. Throughout the day, Apple gave away all sorts of Macintosh doodads, visors, balloons and so on, culminating with the give-away of a hard disk-equipped Macintosh system. My main disappointment came from the lack of music software and musical instruments at the Apple booth. It seemed as though their exhibit would be the best opportunity for show-goers to get "the big picture." Still, it was a good start.

The exhibitor booths occupied Studios H, M and L, while the Paramount theater itself was employed for seminars given by experts on the ▶

Once upon a time, Southworth Music Systems invented the JamBox/4™. Never before had a synchronizer done so much and cost so little. Not only did it let you SMPTE lock your MIDI sequencer to tape, but it let you position anywhere on your tape and lock up in less than a second, and stay locked for hours. As if that weren't enough it came with a built-in MIDI merger that let you record four MIDI instruments at the same time. And it let you filter out the MIDI you didn't want.

Everyone said the JamBox would only work with a fancy professional tape deck. They were wrong. It worked great with any tape recorder, even a home VCR.

But, alas it only worked with the Apple Macintosh™. The people using Performer™, and Q-Sheet™, and Cue™ and MidiPaint™ were all happy. But the people using the IBM PC, Commodore, Atari and hardware sequencers were all sad because they still had to spend thousands of dollars for a sync box that couldn't do nearly as much as the JamBox.

This made the people at Southworth sad too. So they worked night and day in their workshop to find a way to make a new JamBox that could work with any computer or hardware sequencer. They invented the JamBox/2. And it cost even less than the JamBox/4.

Pretty soon everyone had a JamBox and they all lived happily ever after.
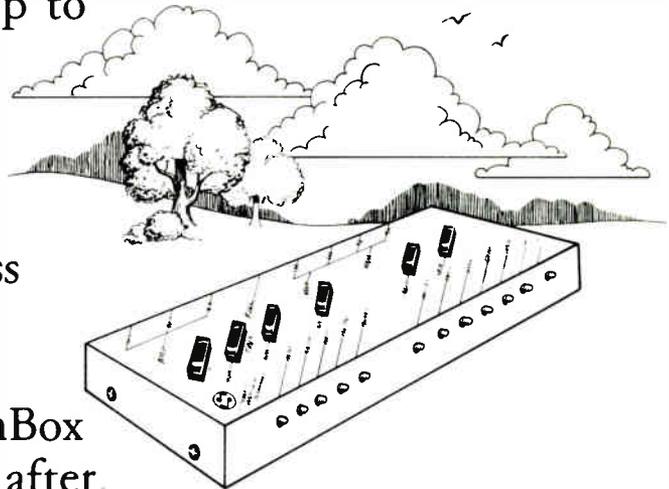
use of computers in professional music applications, and Apple employees, who outlined the company's growing interest in and commitment to the music and entertainment industry.

Being a Mac user myself, I was pleased by the direction of the show, though there was little new there that hadn't already been displayed at one trade show or another. But as a public show it hit the spot, and I hope that similar MacMusic Fests start up in other parts of the country. Perhaps more active involvement by local users groups could help make this happen. In the meantime, Filmsonix, the LA-based Apple VAR (Value Added retailer) that organized the Mac Fest gets big points for taking the plunge. *Rick Davies.*

GRAPHIC NOTES

A new music publishing program for the Macintosh is available from Australian-based Graphic Notes, called, surprisingly enough, Music Publisher. The program has been chosen by Apple for use at the Winter NAMM Show for demonstrations of the computer's music publishing capabilities.

Graphic Notes maintains that their package is fundamentally different from other music scoring and composition programs in that it takes a graphic arts approach to lay out a page of music. The program allows audible proofing of music scores through external speakers.

The program can accept music notation input from a MIDI source and can output to a range of Postscript-compatible laser printers. A music font is supplied to complement the package. It offers orchestral scoring capabilities as well as note and lyric printing.

The suggested retail price is $595.
MORE FROM Graphic Notes, 2-1645 East Cliff Drive, Suite 29, Santa Cruz, CA 95062. Tel: (408) 476-0147; FAX 408-462-6387

ONE STEP AT A TIME

If you've been yearning to try your hand at MIDI painting, but aren't willing to part with $500 for the opportunity, you might be excited by Southworth's One-Step MIDI Sequencer for around $400 less. The program uses palette selection for note quantization, note duration, and track selection, and can read and write MidiPaint format segments, MIDI File sequences or Total Music sequences.

Microscope and telescope tools as well as numbers typed on the keyboard let you look at the data with any of nine levels of magnification. The marquee allows a selection of notes to be enclosed, and a scroll tool moves the visible portion of the screen to look at additional notes. The wide brush lets you paint in a series of quantized notes in a single operation. The narrow brush lets you paint in notes with a one millisecond precision, quantize a single note, or change the velocity of a note. The question mark tool provides numeric editing of note or program change data.

One-Step also includes graphic controller
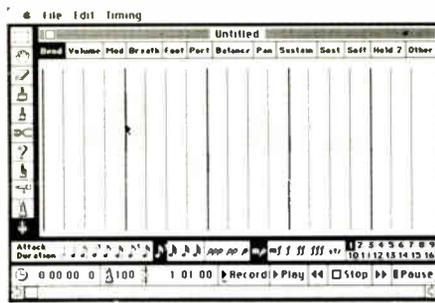
One-Step's graphic editing tools are modeled after the various Macintosh paint programs.
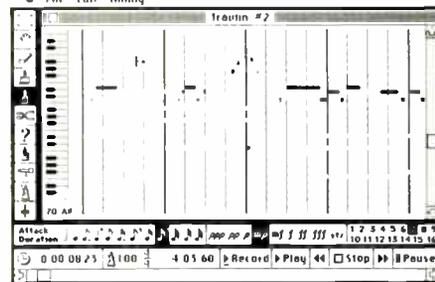
editing so that you can draw in the desired control curve with the mouse.

The best part? Suggested retail price is $69.
MORE FROM Southworth Music Systems Inc., 91 Ann Lee Road, Harvard, MA 01451. Tel: (617) 772-9471

THE DOCTOR IS IN

There are few sure things in the software game, but one of them is that Dr. T's will always be there with yet another program. Most recently, the Massachusetts company announced the availability of three new editors – all for different synths, and all for different computers!

First up is a Korg DS8 editor for the Atari ST by Caged Artist which includes complete graphic point and click editing and control of all instrument parameters. Second is a like-featured editor for the Casio CZ series and the IBM PC by John Lilly, which requires a mouse and a CGA or EGA monitor. The third is a Roland D50 editor for the Apple Macintosh by Caged Artist.

Also available is the KCS Level II with PVG and Master Editor for the Amiga (see December '87 *Newsdesk* for more complete information).

Nothing if not versatile, this Dr. T's.
MORE FROM Dr. T's Music Software, 220 Boylston Street, Suite 306, Chestnut Hill, MA 02167. Tel: (617) 244-6954

ST STORAGE SPACE

Atari Corporation has announced a new, dual-purpose CD player which can either be hooked up to an Atari ST or Mega computer and used as a CD-ROM drive or hooked up to your home stereo and used as a CD audio player. The 540-megabyte storage capacity of the CD-ROM portion equals around 1000 floppy disks of data – somewhere around 200,000 printed pages.

The unit is styled to match the popular computers, featuring a front-loading drawer enabling the unit to be stacked under the monitor. A separate remote control device

allows users to control audio functions from anywhere in the room.

The player is expected to be available in the second quarter of 1988 at a projected price of $599.
MORE FROM Atari Corp, 1196 Borregas Avenue, Sunnyvale, CA 94088. Tel: (408) 745-2000

MIDI TIMES FOUR

Mellotron Digital Corporation is now shipping MUART 4 Port MIDI, for IBM XTs, ATs and compatibles, complete with a PC controller card, MUART junction box with four MIDI Ins, Outs and Out/Thrus, Spirit (an "enhanced" version of 48 Track PC II) sequencing software and cable.

MUART allows the simultaneous recording of up to four separate MIDI networks, recording and playing back with a timing resolution of 600 pulses per quarter note. The software (see July '87 review of 48 Track PC) features 48 tracks, complete individual MIDI event editing, and real-time interactive controls. Sixteen MIDI channels per port, MIDI song position pointer, SMPTE timing references and a system exclusive manager are all featured in the package.

The suggested retail price is $695.
MORE FROM Mellotron Digital Corporation, 36 Main Street, Port Washington, NY 11050. Tel: (516) 944-6789

SOFTSYNTH CARRIES ON

Digidesign has announced two enhancements to the Softsynth digital synthesis program for the Macintosh and Atari ST computers: version 2.1 update and the first Softsynth Sound Library (see March '87 MT for a review of version 2.0).

The version 2.1 update adds compatibility with the Akai X7000 and S700, Casio FZ1, Roland S50, S10, S220 and MKS100, Oberheim DPX1, and all samplers conforming to the MIDI Sample Dump Standard. New features include Phase Locking, Mac II compatibility (Mac version only), and the ability to transfer to other programs directly from Softsynth. Samplers already supported include the Akai S900 and S612, E-mu Systems EII (Mac version only) and Emax, the Sequential Prophet 2000/2002, the Ensoniq Mirage/Multisampler, and the Korg DSS1.

The Softsynth Sound Library consists of three disks, each containing 12 Softsynth parameter files for use as-is or with a bit of tweaking.

The version 2.1 update is available to registered owners for $25; the Sound Library is also $25.
MORE FROM Digidesign Inc., 1360 Willow Road, Suite 101, Menlo Park, CA 94025. Tel: (415) 327-8811

AND NOW, A BRIEF ANNOUNCEMENT . . .

Bacchus Software Systems' Voice Manager Series, TX81Z Graphic Editing System and the TX802 Graphic Editing System now work with the IBM Music Feature Card and the MPU-401.
MORE FROM Bacchus Software Systems, 2210 Wilshire Boulevard, #330, Santa Monica, CA 90403. Tel: (213) 820-9145

♫ **Computer Notes**   Desk   File   Edit   MIDI   Song   Setup

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Ad Lib
Personal Computer Music System

A musical starter system for the IBM PC and compatibles, this package includes an audio plug-in card and sequencing software. *Review by Chris Many.*
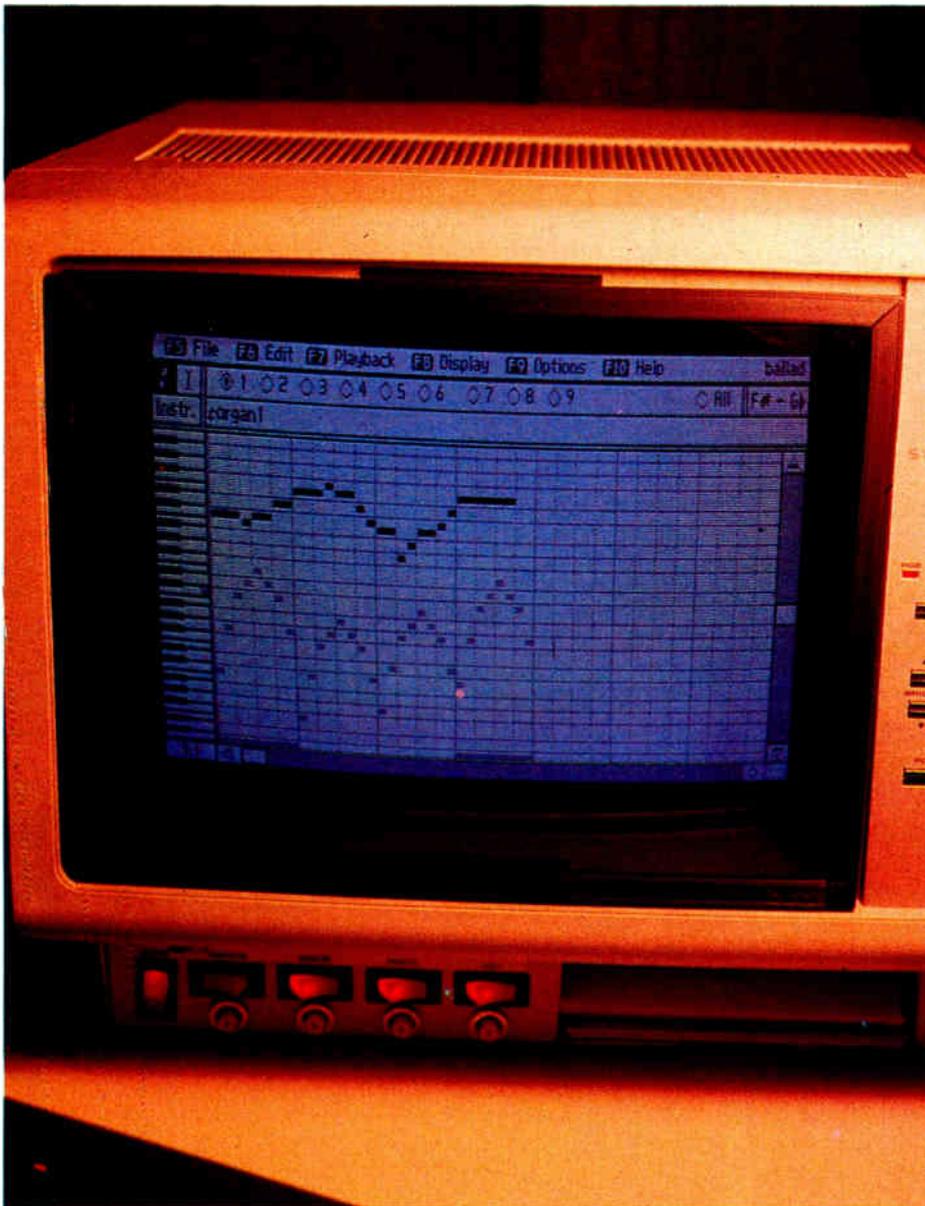
AD LIB HAS joined the fray of companies bringing low-cost FM technology to the ever-growing base of IBM PC/compatible users with their Personal Computer Music System (PCMS). Really a basic music system, it shouldn't be confused with a professional setup. The PCMS is a synthesizer-on-a-card package, which plugs into an empty slot in your PC. There's a single ¼" stereo audio jack and a volume control on the back of the card, so it's pretty simple to plug it in, connect it to your stereo, speakers or headphones and get started.

PCMS comes bundled with two software packages, Jukebox, a playback and demo program, and Visual Composer, a rudimentary sequencing package specifically designed for use with this system. To hear what PCMS sounds like, boot up the Jukebox program, select one of 22 selections and sit back and listen. You know right away this is an FM-based system once you hear that distinctive sound. Unfortunately, this is only a two-operator synth. The card itself is clean, it doesn't suffer from any crosstalk or electronic noise bleed from the rest of the PC's cards or electronics. The sound quality of it is not bad, just a little empty of overtones. Even so, it's multitimbral to the tune of nine separate sounds, or six sounds plus a drum kit. As with any multitimbral system, you're limited by how many voices are available, so this means you don't have much room for chords if you're using the drum kit and have only six voices with which to play.

Turning to the Visual Composer, you have a basic sequencing program that will interface with MIDI if you've got an MPU-401, or similar card, installed. Reminiscent of the Macintosh environment, the sequencer itself is a display of a grid, representing the keyboard (vertically) and beats (horizontally). You input your music by drawing it on the screen, one note at a time, one voice at a time or by playing it in one note at a time. This means if you want the piano to play a triad, you'll need to draw in the tonic with one voice, go to the piano voice assigned to sound two and draw in the 3rd, and repeat this procedure to input the 5th. A little cumbersome to use as a sequencer, for sure, but it gets the job done.

Any sound can be selected for any sound position, again six of them if you're using percussion, nine if you're not. Transposition, cut and paste and tempo adjustment are the other main features included here. In addition to a mouse or a MIDI keyboard (if you've got the hardware), you can input notes from the computer keyboard.

The Instrument Maker is a separate software package that allows you to edit or create your own voices. It's one of the better introductions to FM technology and an easy-to-use editing screen allows you to alter or create voices rather simply, hearing your changes in real time. The fact that PCMS only uses two operators certainly limits the amount of harmonic quality

you're going to be able to coax from the board, but if you've never heard a six- or four-operator system (as a lot of the IBM-owning public haven't, believe it or not), it sounds quite good. And the truth of the matter is, when you have a lot less parameters to deal with, it makes it much easier to grasp the concept of FM if you're only manipulating one modulator and one carrier. Forget complex stacking and the like, this is simple stuff.

Envelope adjustment is done Mac style, arrow bars that are increased or decreased to alter the Attack, Decay, Sustain or Release rates. All manipulations are done similarly on a single screen, including modulation feedback, level scaling, vibrato, etc. It's much easier to do your editing using a mouse, but keyboard access is an alternate means of adjusting parameters. Once

you've got a sound you like, store it on disk and call it up just as you would any other sound when you're using Visual Composer.

Another software package, Music Championship #1, is also published by Ad Lib. It's an educational product, designed to help you (or your kids) listen for and identify seven basic musical characteristics like tempo or key. A fun diversion, it's really for new, young musicians who can play a musical game while learning. Difficulty levels can be set, and there are extra points for bonus and mystery questions. It also includes practice modes to improve the skills you'll need before you sit down to play, and basically a good, though light, educational program.

Overall, Ad Lib has a good sounding board, that as far as I can see, is aimed at the broad base

of PC owners who want to expand into the musical arena without spending a lot of bucks. Professionals can use it, but why spend the money on such a rudimentary hardware upgrade, when for a little more cash you can get a four-operator synth. It's a very basic package, and from that viewpoint, it's a good entry level product for non or new musicians. It's well packaged and produced, the manuals are easy to read, and it provides a good gradient to interest someone in music who owns an IBM or compatible. ∎

PRICES PCMS (includes Synth Card, Jukebox and Visual Composer), $245; Instrument Maker, $49.95; Music Championship #1, $39.95
MORE FROM Ad Lib Inc., 50 Staniford St. Boston, MA 02114. Tel: (800) 463-2686

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C O M P U T E R

real music through ARTIFICIAL INTELLIGENCE

Illustration *Colin Cantwell*

Taking the technology of music another step into the future are researchers and scientists developing artificial intelligence applications for music. Our reporter spearheaded the latest conference on the subject in New Jersey. *Text by Harvey P. Newquist.*

ARTIFICIAL INTELLIGENCE IS the hottest thing in computer science since the invention of the microprocessor. The ability to make machines more human, to think and behave like humans, is the driving force behind AI. And although it is big stuff at places like General Motors, American Express, and IBM. AI has also found its way into the realm of music. How? By creating programs that "know" how to accompany a soloist, by recognizing patterns and tempo and playing accordingly, even by helping to compose music in a specific style.

Let's talk a little about artificial intelligence and what it is before discussing some of the experimentation going on in our musical backyard. The term "artificial intelligence" was coined in 1957 by a group of researchers from Massachusetts Institute of Technology, Dartmouth, and Carnegie-Mellon University. At that time, most computers were used for government purposes, doing mathematical calculations in hours that would normally take humans days. Interestingly enough, these were

the kind of math problems that your basic pocket calculator can do now with its eyes closed. Anyway, these gentlemen felt that the machines they were using to compute numbers could, and should, be used in the same way that we use our brains – to think. They came up with the name artificial intelligence, and proceeded to work on machines that could see, talk, hear, think, reason and understand.

All of this conjures up visions of HAL, the computer in *2001: A Space Odyssey*, or C3PO from *Star Wars*, doesn't it? And those are exactly the kind of machines that AI researchers have been attempting to develop – computers that behave like humans. Along the way they've managed to get certain parts of the man/machine down pat: vision systems that can "see" and are used to guide industrial robot arms; computers that can "hear" the human voice over a microphone and respond accordingly; and even software programs called expert systems that can clone the knowledge of a specific expert onto a floppy disk.

Kind of scary picturing what all of this could do to the humans that are creating such mechanical monsters, isn't it? After all, HAL destroyed the entire crew because he felt that their reasoning was inferior to his. But never fear. For all of its worldly sounding wonder, AI is still very much an infant in the world of technology. All of the things that I have mentioned above are far from being anywhere close to the capability of their human counterparts. Yes, the systems do work, but they tend to know only one thing, such as how to tell the difference between a Coca-Cola bottle and a Pepsi bottle, or how to repair the hydraulic system of an F-16 jet. Try talking to one of these things about Ravel, or show it something that has not specifically been programmed into memory, and you won't even get the courtesy of a blank stare. If the machine hasn't been told, shown, or programmed to understand the nuances of your particular area of interest, you'd be better off asking the kid down the street for his opinion. That's how limited these things can be.

But suppose for an instant that an AI system was created that did specifically address the nuances of music, from theory to composition to recording and performing. What if this computer were programmed with intelligent information about things like tempo, scales,

modes, and rhythm? And let's say it was good enough not only to "know" about these things, but also good enough to teach them?

THE SEMINAR

A group of respected music technologists gathered together to discuss these very issues at an artificial intelligence conference in Atlantic City, New Jersey on October 30, 1987. In a setting otherwise concerned with such esoterica as AI on Wall Street or using personal computers to develop a computerized tax expert, the session on **AI in Music** looked at how artificial intelligence was being used in creating the newest generation of music technology.

The session was chaired by Linda Sorisio, project manager of IBM's expert system application for teaching college students music theory. The program, called THEory of MUSic Expert Systems (THE MUSES), was developed by Sorisio, a former opera singer, and Jay Tobias at IBM's Scientific Research Center in Los Angeles. The presenters included Tobias, Roger Dannenberg of Carnegie-Mellon University (and also a consultant to Apple Computer founder Steve Jobs' NEXT, Inc.); Mira Balaban of State University of New York; Roger Kendall of UCLA; Peter Capell, also of Carnegie-Mellon; and Bernard Mont-Reynaud, a professor at Stanford University, and an associate of John Chowning, developer of FM synthesis and "father of the DX7."

Roger Dannenberg, whose research is being supported by Yamaha, stole the show with his computer-accompanied trumpet playing. The system he developed could detect tempo changes in Roger's playing and adjust its own speed accordingly. More than simply a form of the MIDI "human clock" concept, the computer followed along with Dannenberg's score, playing accompaniment based on where he was playing, and how fast – in real time.

As the demonstration went on, he performed a 12-bar piece which the computer analyzed during his performance. After 12 bars, the computer kicked in with its own accompaniment on synthesizer, using a pattern it had derived from analyzing the notes that Dannenberg had played. The important thing to note here is that this was not simply a sequence which was repeated by the machine, it developed its own accompaniment sequence for the performer's trumpet soloing.

Mira Balaban's research is based on the concept of CSM (Computer Study of Music), using the computer for a general, formal study of Western Tonal Music. Balaban chose WTM for research because people raised in western culture have an intuitive, unconscious (unstudied) knowledge of Western Tonal Music. This knowledge enables people within the culture to accept, reject, or infinitely analyze any new music features presented to them. Thus, she believes that WTM is a good area for scientific research in AI, because there is a judgment criterion from experienced or "expert" listeners that can be used to create an intelligent, analytical computer system. She

MUSIC TECHNOLOGY FEBRUARY 1988

developed CSM using a language developed specifically for AI called PROLOG, and it currently is used to support further research into WTM. An example of this research is the analysis and understanding of varying musical theories in western culture, such as Schoenberg's harmonic progression hypothesis.

Peter Capell's work in advanced music technology (done in conjunction with Dannenberg) has resulted in a project called Piano Tutor, which is an intelligent keyboard instruction system. The program, which runs on a personal computer, is designed to provide lesson material to a student, while monitoring his or her progress, and then planning changes to future lessons based on the student's capabilities at certain levels. For instance, if a problem arises in a student's ability to perform at a certain instruction level, the machine can scan through its memory of the student's performances up until that time, and may perhaps conclude that the student was mistakenly given credit for having fulfilled prerequisites for the lesson, and that missing skills must be taught before going further. The lesson is then terminated by the system which plans its own lesson outline to help the student in the problem areas.

Jay Tobias presented The Muses system mentioned earlier, developed at IBM under Linda Sorisio. Similar in concept to Kapell's work, The Muses is interesting in that it is being done as a project by a huge corporation such as IBM – normally a company concerned with business America's bottom line. Instead of being strictly a piano tutor, The Muses is geared more towards teaching college students principles of music theory. The program is really a series of expert systems combined together that monitor such things as performance, knowledge of theory, and continuing level and rate of learning. The Muses currently runs on an IBM mainframe linked to personal computers and synthesizers for multiple student access.

Roger Kendall explored the use of computers to model intelligent human musical behaviors. He argued that much of what is implicit in music, especially concepts such as style and performance techniques, cannot be gleaned from texts, and certainly not from existing symbolic musical representation (the type of music notation that we have been using for centuries). Style, for instance, requires active participation on the part of the learner, and can possibly be taught and expressed in the form of computer models. To this end, Kendall has designed a system called Maestro which is interactive with the user, and helps to explore the concepts which can't be written out in sheet music or instruction books.

Bernard Mont-Reynaud has been doing work at Stanford at its famed Center for Computer Research into Music and Acoustics (CCRMA). His presentation discussed the importance of the similarity between research in artificial intelligence and research in music. Both involve understanding how the human mind recognizes and perceives certain symbols and patterns (signals), and then how the mind analyzes those ▶

signals and acts upon them. Of particular interest was his discussion of how the physical and scientific representation of music and sound (via waveform/pattern computerized analysis) corresponds amazingly well with the way we've developed our notation system. For instance, a computer digitization of one bar of music (done strictly with the sound of the music) looks remarkably like a musical staff with the individual notes written out in standard notation. Reynaud's research lends itself particularly well to a synthesis of using computer science to understand both intelligence and music.

All that I have covered here is barely the tip of the iceberg in these individuals' presentations. It is important, though, that some light is finally being shed on the use of machines that think in the creation and understanding of music. Surprisingly, very little of this work was possible without the use of a variety of MIDI interfaces and MIDI instruments. Technology begets technology.

THE PRESENT, THE FUTURE

But lest you think that all of this artificial intelligence technology is limited to universities and research & development organizations, you might want to think again. While it may be a while before companies like Yamaha get around to commercializing Roger Dannenberg's computer accompanist, there are already a few products on the market which take advantage of artificial intelligence. The infamous Kurzweil line of keyboards, from the 250 on up, are based on a form of synthesis which uses AI to create (and in some cases, anticipate the creation of) its popular sounds. Personal Composer, the PC software package for composition and sequencing from Jim Miller, uses an AI programming language called LISP to allow for development of applications such as those mentioned above. And the Apple Macintosh II, widely being considered as the best thing for computer music since the Moog, has an internal architecture designed to optimize AI applications of all types, including music.

To prove that talk of such things doesn't take place only in the dark, nether reaches of some cold, clammy techno-electronics lab, the whole idea for this AI in Music conference actually took place as Linda Sorisio and I were wandering that bastion of the music industry, the Winter NAMM show. There amidst the Spandex, big hair, and drum solo booths, we plotted how to get music technology researchers together to discuss AI in one place. With guitarist Eddie Van Halen signing autographs in a booth down the aisle, and Beatles producer George Martin checking out digital recording components in the next booth, it seemed to be a more suitable starting place for discussing any kind of music than a laboratory did.

Who knows? If the technology keeps pushing ahead, maybe you'll be able to buy advice, training, and expertise from these guys for the price of a good AI software package. Having this kind of "artificial" intelligence would be the next best thing to the real stuff. ∎

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Compu-Mates
R100 DrumDroid

Tired of trying to do complex step-time drum programming on your machine's small LCD? If you've got an ST and Kawai's R100, this program may just fit the bill. *Review by Scott Gershin.*

PROGRAMS FOR DRUM machines have been few and far between. Yamaha had one for their short-lived CX5M computer and Intelligent Music has UpBeat for the Macintosh, but there haven't been too many machine-specific programs which simplify the beat-entering process. Now that it's NAMM show time, it's time to make our year's wish lists and I'd like to add a few of those kinds of programs onto mine.

Compu-Mates, a software company that specializes in software and hardware for the Atari ST computers, however, does have a librarian/utility package for the Kawai R100 drum machine called DrumDroid. I recently received a copy and put it to the test.

With whips and chains in hand I booted up the program and was dazzled with the Atari's rainbow of colors. The main screen's title name flickered at me until I managed to make my way back to the computer (after momentary blindness) and moved the mouse, stopping the sequence of colors. It was a feat.

As for specs, the R100 DrumDroid gives the user the ability to view drum patterns in a gridwork of squares that represent MIDI clocks (96), which obviously helps a great deal in the creation of complex drum patterns. The DrumDroid can also play and change tunings of each of the drum samples in the R100 remotely, via the computer keyboard. Computer keys "I-L" represent the three banks of eight drum sounds and keys "Z-?" control the tuning of the last key (drum sound) pressed. Computer key "B" represents normal, unaltered pitch.

The program consists of seven screens which are: Pattern Creator, Song Library, Chain Library, Multi-Mode, Play Scan, AutoDroid, and MIDI screens. For the MIDI hackers out there, there's a neat little utility package in DrumDroid that enables the user to monitor any MIDI data coming into the ST in Hex or Decimal numbers.

Let's first take a look at the Pattern Creator screen. From this screen you can create patterns in real time by using the computer keyboard, or by inserting rhythms in step time using the gridwork of squares. Needless to say, this visual aid greatly increases your accuracy for complex patterns. When the mouse is clicked over the piano keyboard illustrated on the screen, that specific drum sample will be triggered, enabling you to hear the sound. To place a sound on the gridwork, you first select the sound via the keyboard and then click the mouse on the desired beat or fraction there within. The sound

(or sounds) on each square can be identified by moving the mouse to it; the program will automatically display the name of the sound (or sounds) on the top of the screen.

By clicking the pull-down window labeled "controls" the user can send the information created on the gridwork to the R100 or vice versa; the program can receive information from the machine and display patterns programmed on it on the gridwork.

The next screen is called the Multi Mode Selector which gives the user control over the pitch, pan, level, and sensitivity of each drum sound (globally or individually). You can create a whole array of presets on the ST and then write the entire set over to the R100.

On the flip side of that, the MIDI Key-In screen enables you to pick one of the R100's drum sounds, choose its pitch and panning, and

then assign it to a specific key on your keyboard or drum pad. If you're using Roland's Octapads this can be handy in saving extra drum pad configurations or taking the same drum sound, such as congas, and assigning different pitches over several adjacent keys to create a multiple pitched conga patch.

Both the Song and Chain Library screens help in aiding to construct, swap, copy, name, and do the usual piecing together of ideas on the R100. The Play Scan screen is set up to instruct the user on how to play all 24 drums on the computer keyboard. One neat little feature is that F1 starts the currently selected pattern on the R100, F2 stops it, and F3 continues it.

For those of you who consider themselves bordering on the absurd, DrumDroid incorporates an artificial intelligence section that will create random drum combinations or random drum fills.

In trying to form an opinion about the program, the first thing that you must remember is that the R100 DrumDroid is more of a utility package than an actual editor. The reason for this is due to a limitation of the R100, because you apparently cannot access all of its features through MIDI System Exclusive messages. And because this is a utility package, it's necessary for the user to interact with both the Kawai R100 and the ST for the program to be of any use. The program definitely does what it set out to do, though, and adds a few bells and whistles to boot.

I did have one problem with the program. Even though the gridwork editor was created only as a fine-tuning tool, I personally would have liked to see the gridwork assign a specific row number to specific drum sounds instead of just randomly placing any of the drums sounds on any of the squares in the gridwork.

For those of you with Kawai R100's and Atari ST's this is a very useful aid in fine tuning those complex drum parts and expanding the capabilities of the drum machine. If this program were incorporated with a sequencer and made to work with any of the other drum machines, it could be a killer package (hint, hint). For the record, the program uses a hardware key and the manual is fairly straight ahead, as long as you're familiar with the R100. So remember don't beat it, don't eat it, just read it . . . ∎

PRICE $99

MORE FROM Compu-Mates Inc., 8621 Wilshire Blvd, #177 Beverly Hills, CA 90211. Tel: (213) 271-7410

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MASSIVE
— *MEMORY* —

Photography courtesy of Micropolis

If you've ever run out of space on a floppy or cursed the amount of time you've wasted waiting for slow disk drives, you're a prime candidate for a hard disk. The following overview explains how they can make your life a whole lot easier. *Text by Stefan Lipson.*

YOU'VE PROBABLY SEEN references to hard disks in enough places to start wondering what they are and whether or not you need one for your computer. If you are working with your computer on a regular basis and you expect to save a lot of sampled sounds, sequences, and/or documents, then the answer is a resounding yes. A hard disk is the one piece of hardware which seems like a convenience but becomes a necessity in no time.

In short, a hard disk is a storage device for computer data. Also referred to as a Winchester disk, fixed disk, or hard drive, a hard disk has two primary advantages over standard floppy disks; greater storage capacity and easier file access. For computer users, the flexibility that a hard drive affords really is a necessity. Aside from superior disk access times (the amount of time required to retrieve data from the disk), a hard disk offers you the ability to keep organized in a way that a floppy disk simply cannot.

To get an idea of how much easier it is to work with a hard drive, consider the following analogy to storage space in a New York City apartment.

Imagine that you live in a small apartment (the apartment represents your computer) which has no closet or storage space (ie. no hard drive). The apartment is so tiny that you must leave all of your personal belongings in packing crates (a file is a belonging, a crate is a floppy disk). You are so incredibly cramped that you can only open one crate at a time and have one of your belongings out at any given time (come to New York if you don't believe this scenario can exist). Now, let's try to get something accomplished. To take a shower, for example, you must first find the box that contains your soap (locate the right floppy with the right files). Next, you must remove it from the crate for the big hose down.

Fine. Now, you can take a shower. You enter the shower and wash off the caked-on mud and chicken feathers from the nightmare gig you played last night. After the shower, you step out and you want to dry off, but first you need to find a crate so that you can stow the soap (remember, you can only have one item out of an open crate at any given time). Don't forget that the exterminator comes to the apartment every day and sprays the floors with a highly toxic substance, so don't even think about throwing stuff on the floor (sounds like New York to me). Once you put away the soap, you need to find the crate that has your towel in it – save your file on the floppy diskette. After that, maybe you want to put on some deodorant. But hold on, first you have to put away the towel.

Starting to get the drift? While my girlfriend considers that the above would be an admirable way for me to live, it's bad news in a computer environment. It is incredibly time consuming,

tiresome, and it inevitably leads to lost or destroyed – overwritten – files.

Enter the hard drive. Offering anywhere from 10 to 300 Megabytes of memory, a hard drive gives you plenty of computer closet space. That means no more swapping floppy disks in and out. A hard drive with DOS, your programs, and

disk, removing all of your data from the disk.

As bad as these problems sound, you can minimize them by backing up your data onto floppies and having recovery utilities (such as Mace for the IBM) on the drive. Tape backup is another, albeit more expensive, possibility.

But hey, now that you're convinced that you

"Hard drives are also a safer medium. They don't get tossed around and they won't meet the fate which many a floppy diskette has met, death by the dreaded Coca Cola spill."

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your samples/sequences installed on it lets you move directly into your applications without ever touching a floppy.

Having a hard drive also means that you have less to worry about when managing large files, such as samples. With a floppy diskette you must constantly monitor your available disk space for fear of running out. Granted, you can run out of space on a hard drive as well, but it takes considerably longer. And speaking of samples, one other important use for hard drives is as a mass storage device for samplers. By connecting the SCSI ports available on many popular samplers to a hard disk, you can quickly load and store samples.

Hard drives are also a safer medium. They are either inside the computer itself or housed within a protective shell sitting alongside the machine. They don't get tossed around and they
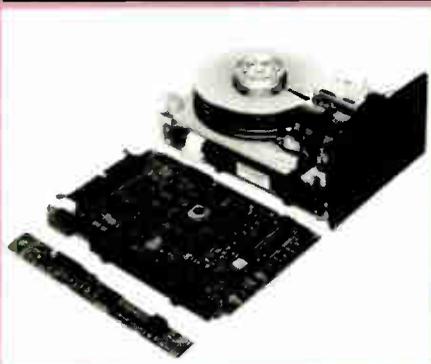
Seagate Technology's ST4192N, a full-height 5¼" high performance hard disk drive with embedded controller, SCSI interface and 160Mbytes of formatted capacity. The drive has an average access time of 17msec.

won't meet the fate which many a floppy diskette has met, death by the dreaded Coca Cola spill.

Of course, no technology is perfect and the hard drive is no exception. The fear of every hard disk owner, for example, is the dreaded disk crash where the hard drive refuses to give you read or write access. In less technical terms? It dies. Another problem is the accidental "format" in which you mistakenly reformat the

MUSIC TECHNOLOGY FEBRUARY 1988

should go out and buy a hard drive right this very minute, let's take a look at what hard disk options are available to you, the advantages and disadvantages, and the associated costs.

- **Internal Drives:** An internal drive is a hard disk that is built into the machine, and is not detectable except for the presence of a small light on the front panel of the computer which blinks when the drive is engaged. Internal hard drives make life easier if you are planning on moving the machine or if you have limited desk space. Not all machine types offer an internal hard drive option.

- **The Hard Card:** IBM (and compatible) users also have an option known as a hard card. Just like other peripheral cards, a hard card fits into an expansion slot inside the PC and works just like any other hard drive. A 20Mb hard card can cost anywhere from $300 to $800, depending on the manufacturer.

- **External Hard Drives:** A hard disk can also be an add-on component, sitting outside of the machine. These vary in both capacity and price, depending on the machine. For example, a Mac 20Mb external can be had for about $600 while the same capacity drive can cost twice as much for an Amiga.

WHICH TO BUY?

If you're trying to decide what size drive to buy, do a little calculating with your floppy files to see how much disk space your programs and sounds consume and then think about how much more you think you'll need. If you are looking at a particular drive, be sure to check the cost of the next larger drive: the difference in price may not be much more and it never hurts to have additional disk space.

While no technology stays around forever, you won't have to worry about hard drives becoming obsolete for a while. They are still the accepted storage medium, as evidenced by their presence in the new generation 386 machines (the new IBM PS series) and the monster Mac II, so don't base your decision on possible obsolescence. If you work with a lot of files and you want to increase your productivity, check out a hard drive for your system. ∎

♫ **Computer Notes**   **Desk**   **File**   **Edit**   **MIDI**   **Song**   **Setup**

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MICRO REVIEWS

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Dr. T's MT32 Editor/Librarian

Voice editing software for the Roland MT32 and Atari ST.

Review by Bob O'Donnell.

BOY ARE THEY fast. Or rather, boy is *he* fast. Bob Melvin, The Caged Artist, who writes voice editing programs for Dr. T's seems to be able to crank out programs for new synths as fast as the Japanese manufacturers can make them. In the case of his latest program, his speed will be greatly appreciated because the instrument in question can only be programmed via MIDI SysEx messages. It's made even more important by the fact that the MT32's lack of battery backed-up memory means that you have to load your own sounds into it every time you turn on the machine.

Like all the other Caged Artist programs, the MT32 Editor/Librarian does not make use of the ST's GEM environment, but instead uses a hybrid interface that is intuitive and easy to work with. When you boot the program, you're presented with a list of all the available timbres in the MT32 and from there you can load in banks of timbres and move individual ones around. To move to the editing portion you simply highlight a voice and then select the Main Edit function from the menu of commands. Once there you can make any changes to the sound you like and all changes will be sent to the MT32 in real time. If you want to hear the results of your work the program incorporates an ingenious method of note triggering. You simply click on the mouse and its location on the

screen will determine what note is sent and its velocity level. The program can display envelopes (and bias curves) in graphic or numeric form and to adjust the graphs you simply click on a point and drag it.

The program also includes pages for adjusting the MT32's System Parameters, its Patches and something Melvin calls Set-Up mode, which basically gives the MT32 the equivalent of overall instrument configurations. Set-Ups include Timbre-to-Part assignments and patch parameters for each of the Parts. The package also includes a D50 Convert program which will convert Dr. T's format D50 files into MT32 compatible files. Unfortunately, there are a lot of differences between D50 and MT32 voices so the process doesn't always work very well.

At the moment, the program doesn't have a special page for the Rhythm Part, but it's apparently being worked on. The problem is that despite the MT32's extensive MIDI implementation its documentation has been rather lacking. To their credit, Melvin and the folks at Dr. T's have figured out a few things even Roland didn't know and they've managed to quickly put together a very good, thorough, easy-to-use program. Good stuff. ∎

PRICE $129

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

Synthia

Digital synthesis software for the Amiga.

Review by Stefan B. Lipson

SYNTHIA, FROM THE Other Guys Software Company of Utah, is intended to turn your Commodore Amiga into an inexpensive digital synthesizer. To be sure, the program promises a lot. Additive synthesis, subtractive synthesis, plucked string synthesis, interpolative synthesis, a percussion editor and more, all for under $100 list. Does it deliver? Well, Synthia definitely lets you create some nice sounds with the Amiga's built-in sound chip. Each different synthesis module has its strong points as well as a unique set of parameters to shape the sound.

In the additive synthesis module, for example, the main editing screen lets you use a mouse and sliders to set a host of options including phase depth modulation, phase shift and the sound's length. The special effects section lets you specify low, high, and band pass filters, not to mention a choice of frequency and bandwidth envelopes across 16 harmonics. The additive synth module lets you further edit the sound by selecting from a library of 16 user definable envelopes. You can also indicate the mix of the first 16 partials here. Regardless of the defined waveform, you may then mirror, reverse, scale, or invert the wave. The percussion module includes another set of editing tools including several different ring and noise filters as well as a detune function. Synthia also comes with an extras disk which includes a number of different patches, including a good organ sound.

Some problems do exist with Synthia, however. I had a problem triggering it via a MIDI controller and there are also a few problems with the interface. For example, when editing a sound the program uses a number of sliders to adjust the various parameters of that sound. The sliders, however, have no numeric display, so you can only estimate where sliders should be set when trying to recreate a sound. I also had a problem loading Synthia instrument files. The system allowed me to load a non-system file that was incompatible with the software. The result? Instant crash and burn and time to reboot the system.

I also found that certain keystroke combinations resulted in a note staying on, ie. infinite sustain. Because the Amiga uses four

channels, I found that pressing four keys simultaneously cleared the lingering note. While that is a solution, it's not one you want to rely on while playing. Finally, while Synthia offers a number of advance editing features, it fails to adequately explain those features in the documentation.

Again, Synthia can generate some nice sounds; it offers a variety of sound editing capabilities, including a harmonic mixing table, several filtering options, and extensive wave shaping capabilities. Unfortunately, Synthia also has some problems. If The Other Guys clean it up a bit and give it MIDI capabilities (on all channels), this could turn your Amiga into a good low cost sound module. Until then, don't be too surprised by the quirks. ∎

PRICE $99
MORE FROM The Other Guys, PO Box H, Logan, Utah 84321. Tel: (800) 942-9402
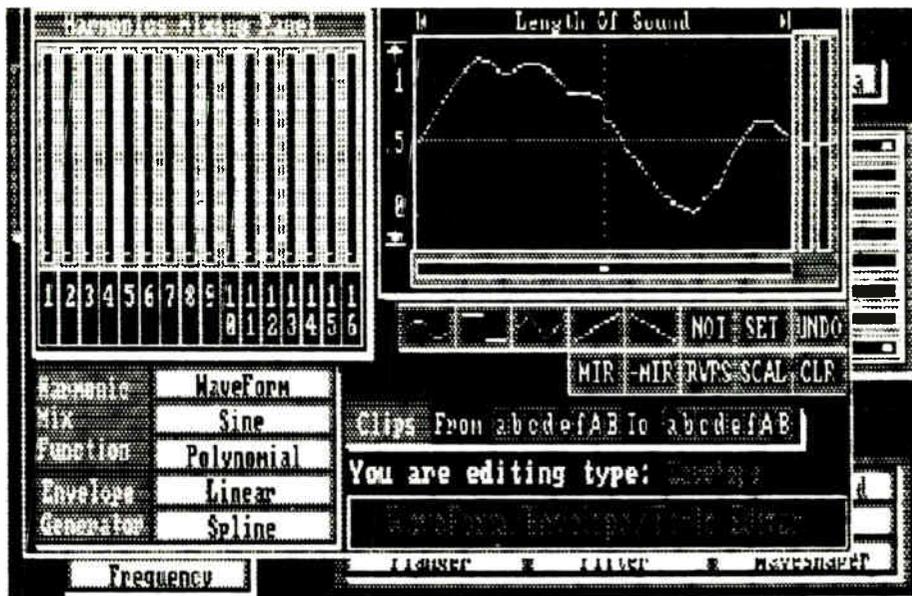
FOR ANYONE WHO has considered doing any programming on the Yamaha TX81Z, the need for a computer-based editor/librarian is obvious. The TX81Z, for all its power and low cost, suffers from the same single rack-space control limitation as other Yamaha products. The temptation to program the TX leads one into a confusing web of menus. The process is certainly not impossible, but intimidating? Yes.

The Beaverton Digital Editor/Librarian is a Macintosh-based program which offers the means to archive TX voices, banks of voices, performances, and so forth, and to alter the parameters which comprise them.

The package consists of two separate programs: the Editor/Librarian, which lets you get TX sounds back and forth between the TX and the Mac, and tweak envelopes and the like; and the Bank Editor, which is geared more towards rearranging voices in and between TX banks, but also contains the more unusual editing tools. In fact, the most noteworthy features of the Beaverton package are the Universal Algorithmic Slider (UAS), and the Compare Window.

The UAS provides a mouse-controlled slider which simultaneously adjusts multiple parameters according to the current "Mask," a window which enables or disables each parameter, and sets either positive or negative polarity of the adjustments. Thus, all four operator's attack rates could be increased while release times are decreased and all other parameters remain intact. Masks may be stored on disk, so you can load different customized masks for different editing needs. The masks become very important when using the program's "randomize" features, which help take the predictability out of a programming session.

The Compare Window displays the parameters of two voices, with all common parameter settings in bold – useful for quick visual A/B comparisons.

All told, the potential for an exciting program is certainly there. Unfortunately, the split into

Beaverton Digital
TX81Z Editor/Librarian

A voice editing program for the Yamaha TX81Z and the Macintosh.
Review by Rick Davies
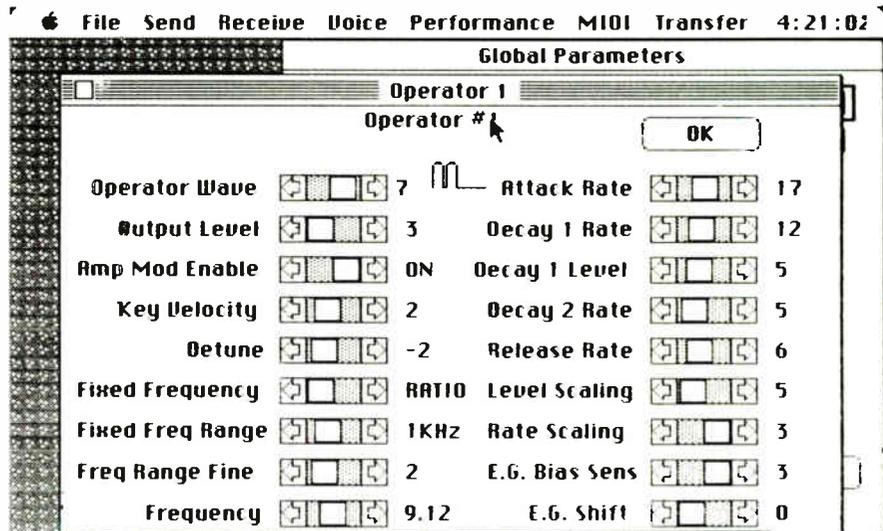
two programs is, I feel, a major drawback. Many times after randomizing a voice, I felt like tampering with the new data, but was forced to deal with transferring from the Bank Editor back to the Editor/Librarian. As Beaverton pointed out, using the new Multifinder for the Mac makes it possible to have both programs booted simultaneously. Still, I would prefer to use a program where all functions are integrated into a single package. I also found that the character of the two programs differed significantly, and I'm afraid this did little to bridge the gap.

Operation in each program was far from smooth; in the Bank Editor, I constantly wrestled with the voice selector window, which provided very shakey control. The UAS, though a wonderful concept, was hindered by slow

screen updates, which made what I had hoped would be a fast editing process into a test of patience.

The documentation was minimal to a ridiculous degree; not even including instructions on how to boot the program, or how the programs are organized.

Beaverton has come up with some clever programming tools such as the UAS and Crosstalk copying features, and these *do* present interesting possibilities for die-hard programmers, but if you're picky about the tools you use, I recommend that you first test drive the Beaverton Editor/Librarian for yourself. ∎

PRICE $139
MORE FROM Beaverton Digital Systems. PO Box 1626, Beaverton, OR 97075. Tel: (503) 641-6260
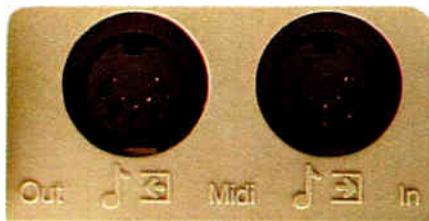
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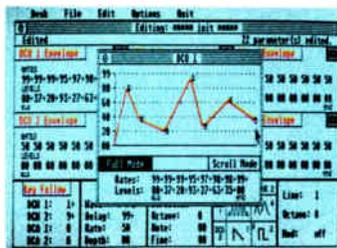
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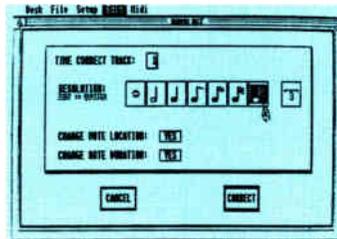
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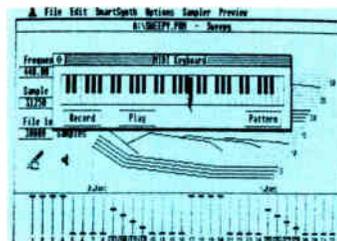
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Dominant Functions
Tiff Sequencer

Dominant Functions, a new player in the PC-software market, breaks the price barrier with Tiff, an IBM sequencing package. At $1.55/track, how can you go wrong? *Review by Michael Stone.*

IBM-PC USERS IN the "general" marketplace (spreadsheets, etc) have for a year or more enjoyed low (under $100) software prices, and most publishers no longer use copy protection. The MIDI software market for the PC has yet to mature, with many publishers still demanding hundreds of dollars for their products. A good number of people have been sitting on the sidelines, hesitant to purchase any PC-based package until prices come down.

But if you need to make champagne music on a beer budget, you could choose Tiff, a 64-track recording package that works and keeps it simple. For $99, most musicians will be delighted by this easy-to-use program.

Tiff, not copy protected, requires the usual hardware: a hard disk, DOS 2.1 or higher, 320K of memory (512K is preferred), and an MPU-401 or compatible MIDI interface unit. The program records and plays back using the hard disk rather than RAM, so the length of your recordings is limited only by your free disk space.

THE PROGRAM

When you invoke Tiff, you'll be greeted by a cutesy copyright message that scrolls back and forth. The other graphics in the program are impressive, all the way to the program's end which issues the technomorphic exit greeting, "■ PROGRAM TERMINATED ■."

After the opening screen, Tiff displays a menu of all the songs on the hard disk. You may name your songs whatever you wish, up to 20 characters, as opposed to DOS' awkward file naming rules. Another nice surprise – one I'd like to see in other programs – is the half dozen-or-so sample songs they give you. (You can instantly demo the program for the curious, or for anyone who wants to know how you spent the $99.)

Any function in Tiff is easy to use, but by far the easiest is song selection and playback: you pick the song name with the cursor arrow keys and go. Tiff is great for those playback-only applications such as weddings, Bar Mitzvahs, and record company A&R men.

The song selection menu has single-letter commands to rename, delete, copy, or list the actual DOS filename of a song. We want to get to the fun part, so we'll hit 'S' to enter the sequencer itself. Unlike virtually all other PC-based sequencers, Tiff has only one screen which is used for every function.

```
                    Sequencing: String Quartet # 4

 Cue      Track Name      Ch Loop    Begin:    2   Punch In:    2   Measure
 01 •    First Movement   01 ---     End:     12   Punch Out:   3     3:
 02 o  String Bass        01 ---
 03 o  Cello              02 ---     Lead-in Bars:  1   Total Measures:  356
 04 o  Viola              03 ---     Tempo:       150   Starting Tempo:  150
 05 o  Violin             04 ---     Metronome:    ON   Beats per Measure:  3
 06 • ------------------- •• ---     Clock Source: Int  Filter:          ON
 07 •    Second Movement  01 ---
 08 o  String Bass        01 ---     ((RECORDING))  Press  SPACE BAR  to stop.
 09 o  Cello              02 ---
 10 o  Viola              03 ---     Recording Memory Remaining:    00011626
 11 o  Violin             04 ---     Disk Memory Remaining:         2791424
 12 • ------------------- •• ---
 13 •    Third Movement   01 ---     Pitch Wheel:     ON   High Note:    B
 14 o  String Bass        01 ---     Modulation Wheel: ON   Octave:     11
 15 o  Cello                         After Touch:     ON   Low Note:    C
 16 o  Viola          ┌──────────────┐oice Changes: ON   Octave:      1
 Play  RECORD         │Selected measures
                      │From measure pointer│it  Sound  Name  Channel  Loop   Quit
                      │ENTIRE TRACK  │
                      └──────────────┘
```

The first 16 tracks and their names are listed on the left of the screen; on the right, the measure pointers, tempo, lead-in measures, and filter settings. The one-letter commands – Play, Record, Bar, Edit, Tempo, Filter, Sound, Channel, Loop, Name and Quit – are displayed at the bottom. The subcommands for these, also single-letter commands, are shown in pop-up windows.

The commands are real simple and they work. Whether or not the set of available Tiff commands will satisfy your needs depends on your personal level of MIDI mastery and what you want to do. The most painful shortcoming is the lack of a note/event editor, promised as a future enhancement. Do you want to see a display that shows whether music has been recorded on a track? Tough Tiff. The misnamed Edit key only inserts and deletes whole tracks. What you lay down is what you get, right down to the program changes.

Other features are lacking as well: the beginning point of a loop, for example, is always the beginning of measure 1. You can filter various types of MIDI information, but you can't reposition an event or transpose by even a semitone. The time signature can be set only as a number of beats per measure.

Program changes and other events can be recorded, but not entered manually. Therefore, at the beginning of every track, you'll want to lay down the appropriate program change command to ensure that the track plays back in the proper voice.

You can skirt around various limitations such as these using your innate creativity and skill in punching in and out. You can, for example, record a difficult passage at a slow tempo and play back the entire piece at normal speed. However, you might have to play a passage dozens of times to achieve the same exactitude as you could have had with a note editor.

Can you cut, paste, insert and do fancy time signatures? No. Is Tiff fast enough to play 64 tracks at once? I doubt it. I have no idea. Could be. You're missing the point. If you have a roomful of gear and the latest 80386-based PC, you probably already have a software package with ten times as many features.

This program is ideal for someone who has a modest investment in MIDI hardware and wants to jam, for example, someone using a single polyphonic synth. (The sample programs supplied with Tiff were designed for the Casio CZ230.)

If you have a PC and are looking to get started in home MIDI recording, this program is highly recommended. If you play a MIDI synth and have bought/are contemplating buying a $699 PC/XT clone with a hard disk, ditto.

I know that my last comment will bring letters from all you Mac/Atari/Amiga users who will point out that the musician I just spoke of, new to computers, could just as easily have bought a machine like yours with its fancy user interface, etc. I'll summarize the reasons a musician might opt for a PC clone: 1) You can buy 'em cheap. 2) You can expand 'em cheap. 3) You can probably get a decent-paying day job should you become at all familiar with the PC.

One of the best parts of Tiff is its thorough and well-written manual, with special help for the various DOS messages that may appear. (Most IBM-PC software manuals omit any discussion of DOS, leaving you on your own.) Even the cretinous "Abort, Retry, Ignore"

messages that were designed into the operating system by Microsoft, and which appear in the middle of your screen at inopportune times, are explained in detail.

The program avoids many of the common mistakes that software developers make, such as the creeping elegance that sets in when too many features crowd a program. The minimalist approach pays off handsomely in ease of use. There are just some features that other programs have that Tiff doesn't. (Conversely, other programs may offer features you'll never use.)

Is Tiff really as bulletproof as is claimed? The back cover of the manual says that the program's primary reliance on hard disk storage "eliminates the possibility of losing hours of hard work due to power loss or lock-up." This kind of claim makes us product reviewers salivate. Let's test it.

I jammed hard for a couple of minutes (a rough, tough Tiff riff??) and, in the middle, hit the Big Red Switch. I'm glad I didn't play anything I wanted to save, 'cause the track you're currently recording stays in RAM until ya save it. Bye-bye track. In fairness, if you get the track down, the program does save every track you record. So Tiff could well be more bozo-proof than some competing programs that store your entire performance in RAM. You might not lose hours, but you could lose minutes if somebody pulls the plug. Moral: not even the most highly-touted software program is Murphy-proof. Always save your work as often as you can. You do make a backup copy of your hard disk from time to time, don't you?

Perhaps you're curious, as I was, about how the program came to be named "Tiff." Turns out the authors were writing a program to be called "Tiffany" and decided to release an abbreviated version. OK, but the name seems to lack something . . . if you enjoy "Breakfast at Tiff," is that part of a complete breakfast?

Tiff's authors say that future enhancements will include a translation program to convert Tiff files to one or more "industry standard" data file formats yet to be determined. Other expected features include IBM PC Music Feature card compatibility, and improved editing functions.

CONCLUSIONS

The profusion of new software developers such as Dominant Functions in the PC music marketplace is going to make things crowded and more competitive, leading to fancy, cheap sequencer packages and an eventual industry shakeout. My immediate expectation is that the makers of other packages will follow Tiff's lead, reducing prices and abandoning copy protection. If this happens, Tiff will face much more real competition than it currently does.

If you're just getting started in IBM-PC music making, if you prefer an easy-to-use, no-frills sequencing program, or if you just don't have a lot of money, plain Tiff rests its case. ∎

PRICE $99

THE RHYTHM MACHINE WITH TODAY'S SOUNDS, TOMORROW'S FEATURES AND YESTERDAY'S PRICE

Assuming you haven't already heard its incredibly low price, the first thing that will impress you about the new Roland TR-626 Rhythm Composer is the sound. We went back to the studio to create all-new high-resolution PCM samples of the finest percussion instruments to give you the latest in today's sounds. And that's just what you'll find on the TR-626: round woody-sounding basses, tight full snares (even including a gated-reverb snare) toms deep enough to please a Phil Collins, clear, vibrant cymbals, and the most complete selection of latin percussion instruments that'll really add some spice to those dance tracks. Thirty digital samples altogether, and each one is tunable as well as level programmable.

Then, since we'd come up with all-pro sounds, we just had to balance it out with the state-of-the-art in pro-performance features: like the most musically-natural and accurate programming software anywhere—combining the best of real-time and step programming with visual accuracy through its sophisticated LCD Display Window. To make the rhythms sound as real as the samples, we've included shuffle, flam and accent features.

On the TR-626 you'll also find songs up to 999 measures,

eight assignable outputs for separate processing of the instrument samples, stereo mix, tape sync, MIDI sync and trigger out. Finally, in a fit of nostalgia, we threw in a price so low it sounds like the good old days: just $495.00.* But probably the most important performance feature is one you won't find anywhere else—and it's an idea that makes the TR-626 the first drum machine that's really usable in live performance. We've added a Memory Card Interface that allows you to load-in stored songs and patterns as fast as you can push a button. Up to 18 songs worth of drum data can be saved and loaded in a flash from the credit-card sized M-128D Memory Card.

If you think all this sounds like the most exciting drum machine to come down the pike in a long while, you're right. Because while the idea of a drum machine isn't new, the idea of a drum machine with some really new ideas of sounds, features and price is positively revolutionary. See and hear the TR-626 today at your Roland Dealer. *RolandCorp US 7200 Dominion Circle, Los Angeles, CA 90040 (213) 685-5141.*

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Akai S900 Sample Editors
Steinberg Sound Works
Drumware Soundfiler

Two new visual editing programs for the S900 and the Atari ST are tested side by side for versatility and ease of use. And the winner is . . . *Review by Scott Gershin.*
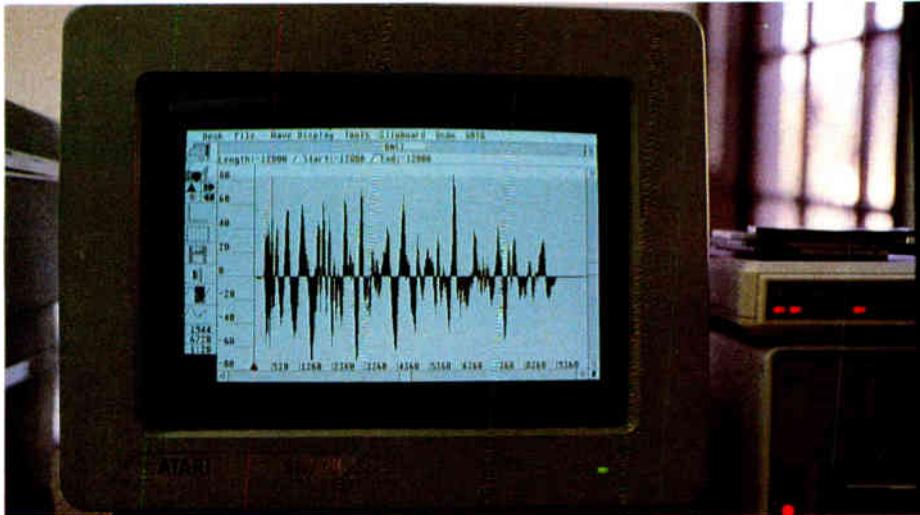
Steinberg S900 Sound Works
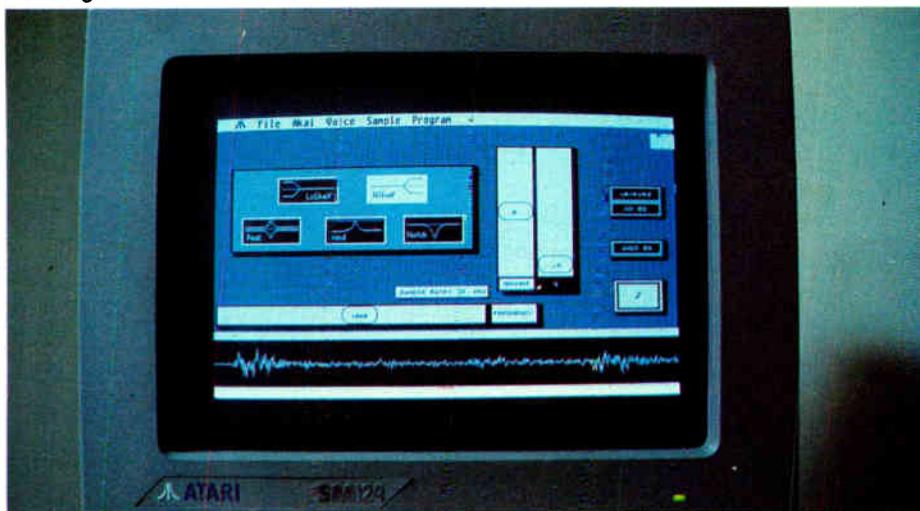
Drumware S900 Soundfiler

Photography Rosemarie Rounseville

FOR THOSE OF us with Atari STs and Akai S900s who've been watching with slight envy at the releases of waveform editing software for the 900 on the Mac, the time has come. Two waveform editors have recently been introduced, filling the long-neglected gap. This review will take a "battle of the programs" approach, but determining the winner is up to you. What features do *you* need the most? Before the gladiators enter the arena, let's take a look at what a waveform editor does. If you already know, go make yourself a corned beef sandwich – and try

to keep the mustard off the page.

A waveform editor enables the user to view a sample through graphic representations using lines, filled spaces and combined dots. The program will allow you to zoom in or out of a specific section of the sample so you can get a closer look at that section of sample. Through cut, paste, merge and reverse commands, the user has the power to manipulate an existing sound to create a new and unique one.

Like many patch editors for synths, wave editing programs also allow you to directly input

values for parameters, rather than having to slowly increment them. One of the constant frustrations in using the Akai S900 is having to enter data through a data wheel, scrolling endlessly through the available options. It's enough to make you give up after spelling out three characters of a sound's intended name.

Now without further ado, allow me to introduce the contestants . . . In disk holder number one, we have in blue and white, Steinberg's **Sound Works** from Germany, distributed by the folks at Russ Jones. And in disk holder number two, weighing in at three and a half in black and white is **Soundfiler** by those guys at Drumware – the same company that provided drum chips for the Linn and Oberheim drum machines.

SOUND WORKS

A few months ago, Steinberg released their waveform editor Sound Works for the Akai S900 after great success with their sequencer program, Pro24. The program arrives in a large four-ring binder that's stored in a cardboard packet. I suggest that before jumping over your equipment and thrusting in your disk that you go through the manual at least once. It is *not* the easiest program to operate. Before booting the disk, Sound Works will require insertion of their hardware key into the cartridge port. Make sure not to lose this because Sound Works will not allow you to run the program without it (and replacing a software key creates its own challenges). Sound Works will run on high and medium resolution (black & white and color) monitors.

The main screen consists of nine icons on the left and three waveform screens. The top icon is a diagram of a filing cabinet which enables the directory of samples and programs. After clicking over the decided name, a new window will open displaying memory parameters, sampling rate, etc. To increment parameters, click the mouse over the parameter you want to effect and while holding down the button, drag the mouse to the right. To decrement, follow the same procedure except drag the mouse to the left. This could pose a problem if you have limited space for mouse movement.

Sound Works has three screens, each containing a sample stored in RAM (Atari memory permitting) that you can have immediate access ▶

Steinberg: Sample zoom display
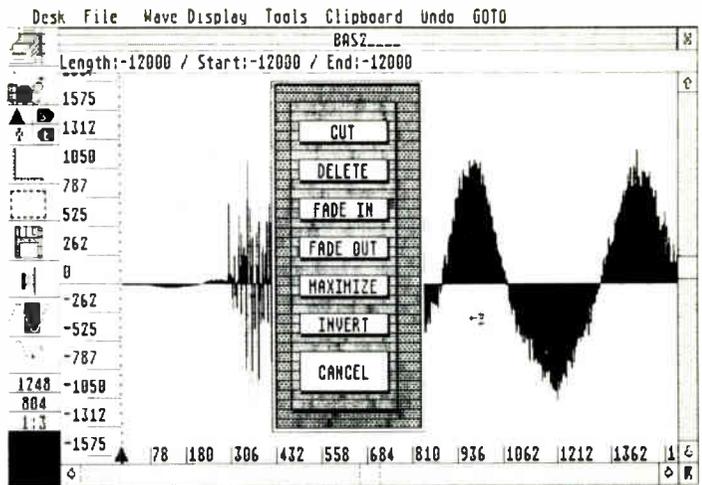
Steinberg: Sample editing options.

to by calling up the desired screen. A drum kit icon allows you to access the trigger input controls – if you happen to have the drum trigger inputs on your S900. Next in the hit list of icons are four symbols that are used for editing samples. These symbols can be dragged into the waveform to mark start, end, crossfade, and clipboard functions. As you keep descending, the next icon is a software toggle that turns the timing markers on and off and which can be used like a ruler to give visual aid in locating a specific portion of the sample. This is helpful when zooming in on a section of the sample and for keeping track of where you are.

Icon number five is a square of broken lines that is used to zoom in on a specific area of a sample – one of two ways to zoom. The disk icon that follows enables MIDI junkies with four Meg Ataris and a shelf full of hard disks to specify one of eight drives that can work with your set up. (Remember that RAM disks can be used.) Continuing down the shaft, we come to the speaker icon, used to trigger the sample for play through the Atari speaker. The eighth icon is used to activate the edit mode for sample manipulation. And last, but certainly not least, is the icon that enables the user to draw his or her own waveforms or patch up and redraw bad sections of sampled material.

The way the Sound Works program works is that a sample is transferred over MIDI to the ST and then is loaded onto a formatted work disk that is used as a buffer and back-up area. The disk is useful because it is faster to access the Atari's disk drive than to constantly transfer from the Akai. To monitor the sample, Steinberg has created a lower quality version of that sample in the ST that can be played through the computer's speaker. Note that the quality leaves a lot to be desired and is only useful for referencing. If you are working on a large number of samples, have plenty of formatted disks available. Buffer memory is used up quickly.

To modify any of the parameters in edit mode, you have to work your way through several windows to arrive at a screen of listed numbers and values. The same is true for the key group screen, and changes have to be made first in the computer, then transferred to the Akai and then transferred back to the ST to verify that change – all of which has to be done by the

user. Love them key strokes! Anyone have a copy of *War and Peace?*

Unfortunately, Sound Works doesn't make full use of the ST's graphics capabilities – the last thing I want to see on these pages are rows of numbers and values. I suppose I'm spoiled. Sound Works does, however, contain three waveform screens that can be toggled through, with quick access to each. Some of the impressive bells and whistles in this program include a 3D display of the sample's wave shape over time. But beware: it takes a while to draw out each section.

The program includes a function that will allow you to maximize the volume of your sample if sampled too low, which helps the signal-to-noise ratio. There is also a Go-To function that will automatically take you to different sections of the sample – a great time saver.

One small bell is that the system allows you to format disks and do disk management functions while still in the program, a key feature considering the way this system works. It also contains an AM/FM synthesizer with four oscillators that can be controlled both with AM or FM synthesis. One drawback is that none of it is in real time. The Atari has to recalculate each time you change a parameter and turn them into a sample that can be transferred to the Akai. But for those of you with patience and time, you can end up with interesting results. Remember that you can cut and paste samples with the synth sounds.

SOUNDFILER

The next contender is a program that we weren't sure would make the deadline for Battle of the Programs. But at the last moment, a messenger on a white steed came riding by to deliver a package containing the valued disk. Impulsive as usual, I decided to boot up the program before reading the manual.

When I did and scanned through the pull-down menus, I was happy to find that most everything was straightforward and easy to use. The prototype manual provided quick definitions of those few symbols that were completely foreign.

Only the high resolution, black-and-white version was available because the program is so new, but the color version is scheduled to join its brother in the immediate future – with a couple more bells and whistles thrown in.

The Soundfiler uses a dark background with the waveform drawn in white (this being the opposite of Sound Works screens), and has two waveform displays that display each sample. On the bottom of the Loop/Draw screen the display represents the full sample and utilizes two pointers (top and bottom) to set the start (top pointer) and end (bottom pointer) points within the sample. If looping is activated, the top pointer controls the beginning of the loop. The waveform display in the middle of the screen is used to view a selected portion of the sample that can be seen as a whole on the bottom screen. It includes functions that enable you to
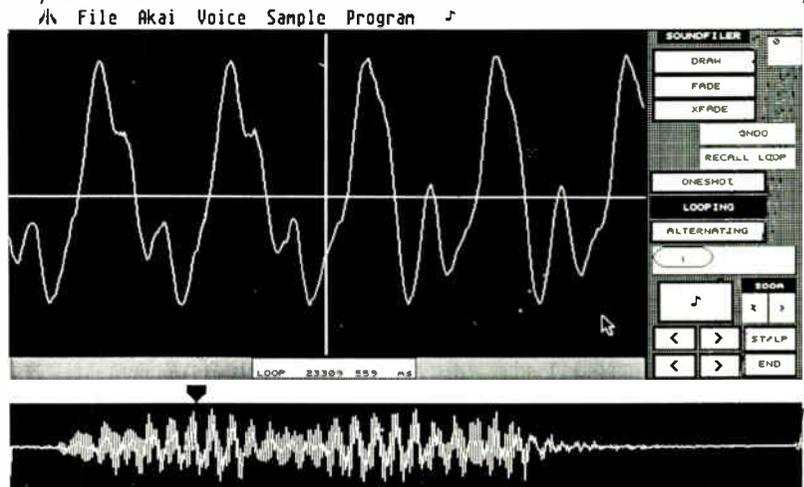
Drumware: Loop and draw display.

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Drumware: Digital envelope display.
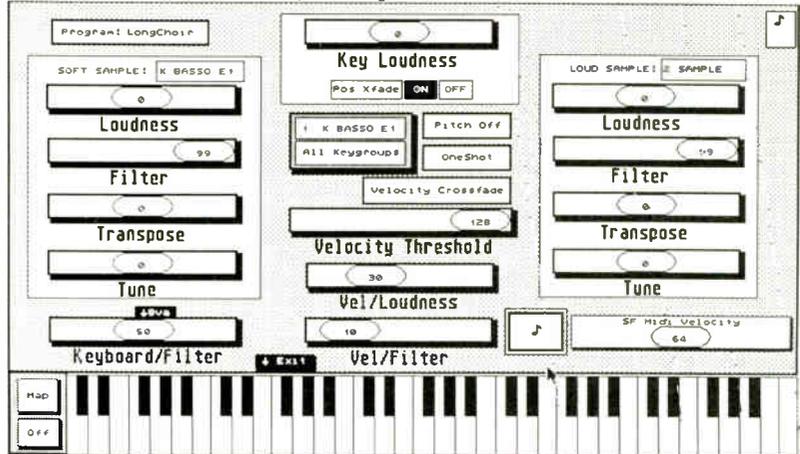
Drumware: Knob I page.

zoom in and out of a designated area. These are done with arrow icons and a box above displaying the ratio of the zoom.

Similar features are found on Sound Works except that when you click on the eighth note icon in Soundfiler, the Akai S900 will play the sample that you are viewing. You don't have to worry about trying to listen through the Atari's cheap speaker. All edits, except for actual sample manipulation (cut, paste, reverse, crossfade looping, and draw), can be done in real time. This also includes normal looping functions. When a parameter is changed, it is automatically sent to the Akai and the program responds that the update has taken place (only one key stroke).

All of the parameters in Soundfiler are represented by knobs and sliders, giving the user a quick glance representation of how the sample is set up and modified. The envelope on the Knobs 2 page is made up of five points joined by a line that can be changed by clicking and dragging one of the points to its new destination, thus updating that parameter in real time.

Any sound can be played by clicking any of the piano keys in the program's display. What happens is that this will trigger the corresponding sample assigned to that key in the Akai. The Knobs 2 screen also contains all of the LFO and velocity attack/velocity release parameters that are represented as sliders. The Knobs I screen looks similar to that of Knobs 2 except that it contains all the parameters for filtering soft/loud

samples in the key group.

Both Knobs I and 2 screens are accessed through the Keymapping screen by dragging the mouse into the designated squares. The Keymapping screen is very well illustrated, using bar graphs to show the range of the first 16 key groups. (The remaining key groups can be viewed by simply toggling a software switch.) To find which sample is assigned to which key group, all you have to do is call up the catalog.

Soundfiler also contains a 16-bit equalizer
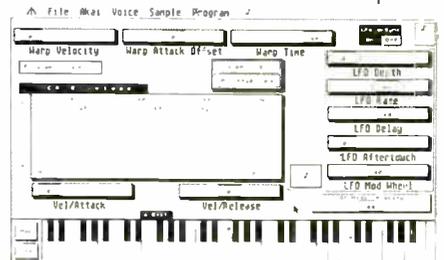
Drumware: Knob 2 page.

function, and there are five filtering choices – Hi/Lo Shelving, Notch, Bandpass and Peak with full cut/boost, frequency, and Q controls. This can be very useful in shaping your sound and can be used to compensate for the lack of low end on the Akai.

Soundfiler also includes a screen that allows you to cut, paste, merge, and hand draw the waveshape and envelope of the sample. The Soundfiler, like Sound Works, requires a formatted blank disk in the disk drive to be used

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as a buffer and backup section. Unlike Sound Works, however, this is the only time that Soundfiler needs to use the disk drive as a buffer area.

A new feature of Soundfiler allows you to control crossfade time while looping a sample, so you can decide on whether the sample will have a long crossfade or something just short of a butt cut.

TO SUM UP

Both programs offer a number of similar functions. They both allow you to cut, paste, and merge waveform sections; hand draw wave shapes; provide control over looping and other S900 functions; have zoom capabilities and allow for multiple floppy and hard drives.

Sound Works enables the user to manipulate the sample without having to transfer back to the Akai by creating an eight-bit version that can be used for referencing through the computer monitor's speaker. It can also display a sample in 3D – a harmonic display – and it contains an AM/FM synthesizer that can be combined with samples (memory permitting). On the down side of things, though, Sound Works requires a large number of key strokes and mouse clicks to accomplish many functions and requires that you have a small arsenal of formatted disks to use as buffer area.

The unique capabilities of Soundfiler, on the other hand, include a 16-bit equalizer with five filter responses. Soundfiler also works in real time and is quick and easy to use. It gives the user control over the crossfade time and uses knobs and sliders instead of rows of data. One

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Drumware *"Soundfiler contains a 16-bit equalizer function, and there are five filtering choices. This can be very useful in shaping your sound as well as compensating for the lack of low end on the Akai S900."*

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other difference is that Soundfiler displays the sample in two screens, thus creating the ability to see the sample as a whole as well as the selected section.

These are just a few highlights about each of the two programs. Both are very powerful and come with their own set of bells and whistles. Soundfiler is an easier program to use initially, but with time and patience Sound Works can be mastered.

The winner can only be determined by your personal needs for an editor. The unfortunate truth is, however, that my attempts at trying to teach either program to make a double-decker corned beef sandwich on rye have been miserable failures. Maybe in the next versions . . . ■

PRICES Sound Works, $349; Soundfiler, $299
MORE FROM Russ Jones Marketing, 17700 Raymer, Northridge, CA 91325. Tel: (818) 993-4091; and Drumware, 12077 Wilshire Blvd, Suite 515, Los Angeles, CA 90025. Tel: (213) 478-3956

LEVEL II

TRACK-MODE RECORD/PLAY

OPEN-MODE EDIT

COPYIST V1.4

PVG

ROLAND MT-32

PROGRAMMABLE VARIATIONS GENERATOR and MASTER EDITOR

The PVG was written to allow the user to computer to generate variations from any previously recorded musical part. It can produce changes or permutations in pitch, dynamics, controller values, rhythm and expression, according to instructions that can be entered in detail by the user. For example, the beginning of a part can be given small pitch changes, while the end of the part is given large velocity changes. The program can produce a single variation on a part or a series of variations that evolve over time. The variations produced can contain some randomness or be fully determined.

Integrated with the PVG is the Master Editor, an editing and orchestrating tool of unprecedented power and flexibility. It allows the selection of notes for editing operations by a wide range of criteria, including pitch, velocity, interval from next or last note and position in pattern.

ONLY FROM THE DR!

OFF THE RECORD

Reviews by Adam Ward Seligman

"They have great technique but they're not songwriters." – Abel Garcia, Guitarist, The Affect

MUSIC 102 IS now in session. The popular theory that you can be successful if you can hold an instrument, look good on stage and scream into a microphone has been a part of the rock world for years. Nothing else truly explains such flatulent groups as the last fifteen years of The Rolling Stones or Starship. But what about the groups that do have great technique, that have *studied* music (heresy!) and also are capable tunesmiths? Who are they, and does anyone really listen to them?

RUSH
Hold Your Fire
Polygram

RUSH HAS BEEN putting out an album every year or so since I was in junior high, and while they have only recently taken off on radio and with videos, they have always sold a lot of records. The last few offerings from the group, starting with *Moving Pictures* and continuing with their current release *Hold Your Fire*, have been increasingly polished, with smoother production values, less screeching from Geddy Lee, and a dedication to certain musical concepts (odd time, extended solos from guitar or bass, heavy drum overplaying) that everyone else in rock has shied away from.

The new album features what will probably be their biggest hit yet, 'Time Stand Still' which features guest vocals from Til Tuesday's Aimee Mann, an excellent bassist from the Berklee College of Music. This song manages to be a love song with ecological values and a bridge in 7 (and yes Abel, you can dance to it!). It's also so seductive and pretty that I didn't realize it was Rush the first time I heard it on the radio. The fact that it has a strong video on MTV should further

increase the sales, and maybe inspire a few budding rock musicians to check out a minor harmony or two.

Now that the vocal problem has been solved, and the keyboard overdubs organically worked in instead of showing up like bombast or ballast, the remaining area that Rush needs to work on is lyrics. I don't know Neil Peart's politics, and for a long time thought he was an Ayn Rand inspired techno-fascist or even worse, a Libertarian like my brother-in-law Don. But after going through the lyrics of the last four Rush albums I must admit to a confusion which is genre inspired. Bits of futuristic science fiction, heavy apocalyptic themes, ecology, lack of communication as the cause of the world's problems . . . I have reached a point where I don't even try to analyze Rush anymore. I sit back and listen, and relax, and hope that the millions of teenagers buying the record are doing better in political science than I did, and are not buying into the futility/elitism of Neil Peart. ∎

THE FENTS
The Other Side
Passport Jazz

UNCLE FESTIVE
Say Uncle
Optimism

JAZZ HAS ITS own share of techno players and two new fusion records represent the diversity of a form that most jazz critics wrote of years ago. Torn between the hard-edged intensity of Mahavishnu or Return To Forever and the lyricism of the Pat Metheny Group, two styles of fusion have developed. The Fents represent the hard screaming guitar and fascination with rhythm that Rush was

formed around. Uncle Festive is more concerned with melody and other soft values, like piano solos or acoustic guitar. The nice thing is both records are equally good, striving for a sound and achieving it.

The Fents are led by four amazing kids in their twenties, early thirties. Adam Holzman has been on the road playing keyboards with Miles Davis, Ted Hall has opened for Allan Holdsworth, Laurance Cottle has played bass with Eric Clapton and drum master Moyes Lucas has played with a range of artists from Kenny G to Jeffrey Osborne. They are a potent combination when left to their own devices, which on this record include heavy synthesizers, fretless bass, a guitar sound so burning that it left ashes on my CD player, and a drum intensity that reflects what Billy Cobham did with Mahavishnu.

The opening track 'Where's My Producer' is a good intro to The Fents style of fusion. Heavy metal guitar lead, a backbeat that never gives up, a driving bassline and some of the best synthesizer leads I have ever heard. Then, these bastards have the balls to lay back into a piano solo! Good God, is that fair to an electric fusion fan? You bet your . . . earplugs it is, because when it goes back to the fusion the contrast makes it even more exciting. 'My Body's a Temple' is the token odd-time song, but what Lucas does with the time underneath Hall's spiritually felt solo is beyond me. At one point Hall is in five, Lucas is in four, Cottle is alternating, and Holzman is comping to it all. 'I Don't Want My MTV' is a series of rock and jazz-rock clichés played with a rollicking sense of humor, and yet another series of great instrumental prowess from a group whose publishing company is called "How Come You Guys Don't Have A Singer Music."

Uncle Festive has a background so clouded in mystery and darkness that it involved months of bribes, midnight phone calls, and a visit to a secret clinic in Mexico to uncover. Then I found out that the four musicians in the group, John Pondel, Ron Pedley, Marc Levine and Bud Harner actually play with, wait, there's a knock at the door, I'll be right back . . . Never mind.

The tunes on the second record they've released – they have one coming out on Denon in early spring – range from the driving new wave beat of 'Anything L' to the African percussion of 'N'Gaundrie' to the bonus CD track 'Where's Marcus,' which features an odd time signature. The players are excellent, especially Ron Pedley, who seems at home on any keyboard, but whose piano skills are beyond belief. The Art Tatum homage on 'Say Uncle' or the ballad 'Renewal' which is recorded live to DAT, demonstrate one of the most exciting keyboardists to come along in years. The rest of the band is equally good, with John Pondel's guitar synthesizer soaring over Marc Levine's five-string bass and Bud Harner's subtle but "in the pocket" drumming.

Any questions, class? Oh yeah, where Uncle Festive met? They play with Barry Manilow.

Class Dismissed. ∎

HORN PLAYERS REJOICE. You are no longer confined to the limitations of only being able to play sax or clarinet sounds. Don't get me wrong – to me, the sax is the most wonderful sounding acoustic instrument around; but still, we have been largely ignored by the electronic music manufacturers (except for them trying to sample and/or synthesize our sounds). Until recently, nothing has been done to encourage horn players to get into synthesizers. Those of us who have caught the hi-tech bug have had to switch to playing keyboards, which in my case has led to more requests for John Cage's monumental "4:33" (which consists of four minutes and 33 seconds of silence) than "Dexterity."

Well fret not, sax controllers are beginning to come out of the woodwork. Two major companies have released "pseudo saxes" and a few smaller companies are sprouting up showing their own hardware. Artisyn, one of these new companies, has come up with a very good looking, functional, and potentially expressive instrument.

The Basics

UNPACKING THE INSTRUMENT you'll find the SX01, which is the controller itself, a power supply, a power cable, an SX01-to-power supply cable, a mouthpiece and a neck strap. The SX01 is the "sax" part of the system. It features a cut-down version of the standard sax keys which gives the front a very slick look. The keys are high-quality momentary contact switches which, once used to, allow for extremely quick playing.

On the back of the unit there are two buttons that are used for octave keys. A total shift of four octaves is possible, that's five octaves of "C," so no more shifting to side keys or alternate fingerings is necessary to get that extra octave. Two more buttons are also on the back that enable you to shift through the different parameters and patches stored on board. These two buttons are used in conjunction with a rotary pot that works as a data entry wheel. A pitch-bend and modulation wheel are also on the back, located so as to allow access with the right-hand thumb. The pitch-bend wheel has a center detent but does not have a return to center spring. This allows it to be used as something other than just a pitch wheel such as a pan control, portamento amount, or as a second modulation wheel. I particularly like the feel of the modulation wheel. It is very smooth, which makes it easy to bring in subtle amounts of modulation.

In addition to those already mentioned, here are the two strongest features that separate the MIDISax from all of the rest. The first is an eight-character LED display and the second is the fact that you use a standard alto sax mouthpiece. The display

Photography Rosemarie Rousseville

Artisyn SX01 MIDISax
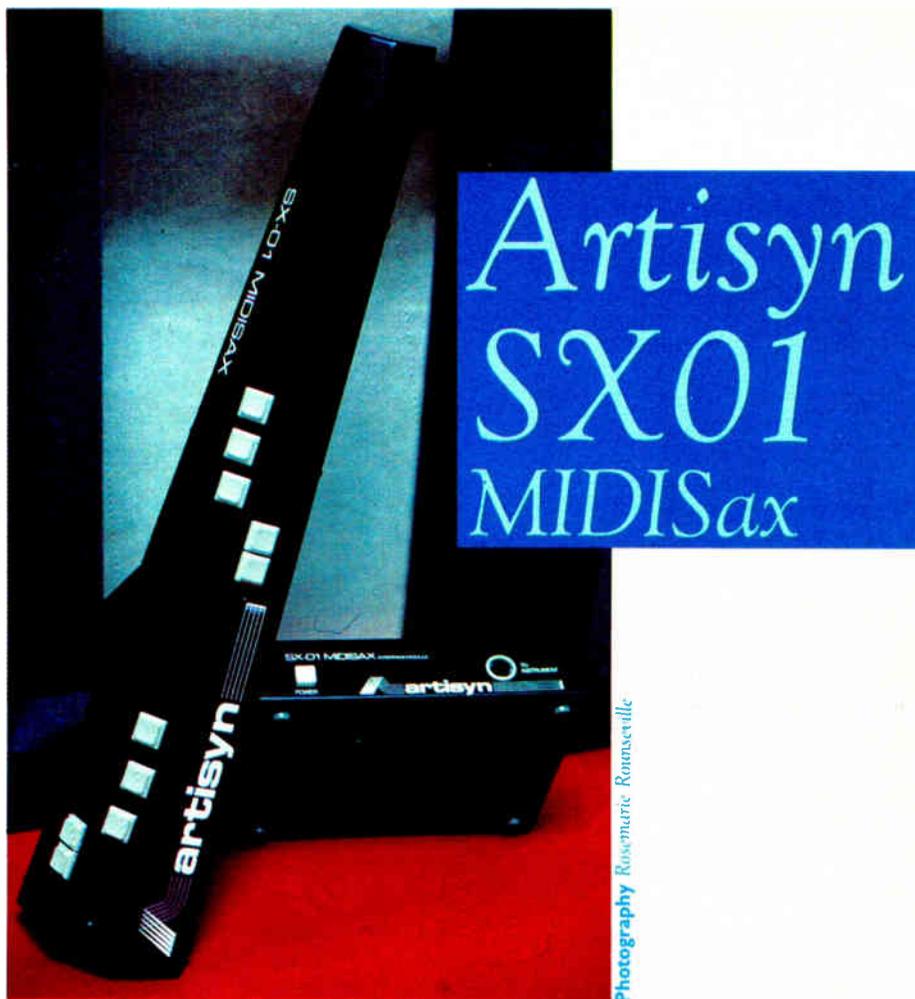
The saxophone-like wind controller from this new, small company combines the best of the old with the best of the new by featuring a pressure-sensitive reed and an extensive MIDI implementation. *Review by Glen Darcey.*

is positioned so as to look you in the face as you play. It can display what patch you are currently on as well as all of the editing parameters within a patch. In a live situation, this is of great importance because you can't always hear yourself and stage lights make most LED and LCD displays disappear. (One time I called up a screaming cat sound because I couldn't see what program I punched in.) Because the SX01's display is bright orange and shaded, stage lights probably won't affect its visibility.

Before I start up on mouthpieces I should mention the reed. The MIDISax comes in two versions, one with a reed that senses lip pressure and one that doesn't (you can upgrade to the pressure version for $150). The unit I received came with the pressure-sensing reed, so when I mention the reed, that is the one to which I am referring. The reed is some sort of a plastic that doesn't vibrate. (No sound is generated by the reed, so it doesn't need to vibrate). It's a lot thicker than a standard reed to prevent breakage and because it has to hold the electronics. If it had a

standard reed number it would be a nine and a half.

One thing to note is that because this is an electronic instrument, the mouthpiece has nothing to do with the actual sound. I did find, however, that playing a mouthpiece I've used for eight years helped make the transition from my sax to the SX01 easier. It uses a standard alto sax mouthpiece and actually comes with a Rico Royal C7. I tried a variety of mouthpieces to see if they all worked. I had no problem with any of the plastic ones but the big test for me was to make sure it worked with my Brilhart and Berg Larsen metal mouthpieces. One thing I did notice was that the different chamber sizes did change the way the breath pressure responded and different facings (the amount of gap between the reed and the mouthpiece) changed the reed flex amount. I found that a small chamber and an open facing made the best combination, but of course that's a personal preference.

The SX01 is connected to what is called the PSIM (power supply interface module)

via a 20ft connector cable which is supplied by Artisyn. The PSIM is a small black box that sends power to the SX01 and receives data which is to be transmitted as MIDI. It also contains the MIDI In, Out, and Thru jacks. The only problem I found was that I didn't have anywhere to put the box. An optional set of rack ears would be greatly appreciated because it seems to be exactly one rack space high, though only about a half a space wide.

MIDI

SOMEBODY AT ARTISYN put in many hours of overtime developing the MIDI implementation for this beast. The SX01 can transmit on any of the 16 MIDI channels, and attack and release velocity are supported as are your standard pitch and mod wheels. The unit also transmits the amount of reed flex, or lip pressure applied, as well as the wind pressure, the key speed (how fast you are playing), and something Artisyn calls Continuous Envelope Map Code, which is an on or off type of command (ie. Sustain pedal information). Now you might say that is incredible enough as it is, but they didn't stop there. Not only are reed flex and wind pressure transmitted, but the instrument transmits two reed flexes and two wind pressures, both of which can either be transmitted on the same or different MIDI modulation channels. Are you satisfied now? Obviously they weren't because they didn't stop there. Every one of these modulations, excluding the pitch and mod wheels, can also be scaled to transmit within certain value ranges.

Unfortunately, there isn't enough space in this article to fully explain these scaling functions and how they can be used, but I'll try to give the *Readers Digest* version. Each continuous controller has a range of 0 to 127. A gain, offset, threshold, and limit value can be given to each modulation source. The gain is the value that is set to determine the slope of the response. The limit value limits the upper end of this response slope. So, if a gain of 127 was set with a limit of 100, you would have a steep slope, or a radical amount of modulation, until you got to a value of 100 at which point the slope would stop and no value above 100 would be transmitted. Just as, if you had a gain of 80 with a limit of 60, you would have a mellowed slope that would end at 60. Now if that isn't confusing enough, a threshold value can also be set. What this does is sets a point on the lower end of the scale at which modulation begins. So now you can have nothing happen at both the upper and lower ranges of your scale. The final touch to completing that dazed and confused stare on your face is offset. Offset adds to all of the values given so far, and its effect is shifting the range of the overall modulation.

A good example would be if you were

bending pitch from D3 to A3 and you added an offset to the modulation amount. You might now be starting at G3 and bending to D4. Notice that the amount is a fifth in both cases, it has just been transposed up to a different area of the keyboard. Are you still with me? If not, don't worry. The owner's manual uses nine pages of text, graphics, and mathematical formulas to explain these functions and I had to read it twice to get a firm grasp of what was going on. The main thing I'm trying to get across is the great amount of flexibility that can be derived from this system. The only weakness I found was that there was no way to create zones that would enable you to change MIDI channels in different areas across the keyboard, although MIDI channels can be changed through velocity, which also has its own gain, limit, threshold, and offset values (not that again!).

Programming

ALL OF THESE features would be nice, but useless, if they didn't have the ability to be stored and recalled easily. Luckily enough, the good people at Artisyn thought about this and gave you ten user programmable patches and sixteen factory preset patches. At first, you may view this

as too few memories, but I couldn't see using more than five or six patches in a live performance situation anyway. These patches can then be played in one of two modes: Linked and Unlinked. Linked mode allows you to assign a user or factory patch to a MIDI program number. If a Linked patch is called up, it will set up your MIDISax's preset and transmit a patch change command to your receiving synthesizer. I found this to be a nice feature so as not to have to remember "did patch 4 go with patch 29 in the bridge section," and risk getting nasty things thrown at you.

Performance

CONSIDERING HOW FOREIGN this thing looks, it is very easy to adapt to. I found the key placement to be spaced very well and the keys themselves were extremely fast. The only hard things to adapt to were the octave keys, which I finally became used to after a weekend of solid playing. Also, because of the fact that it senses notes so much faster than most people can play, I encountered note glitches on occasion. The one saving grace to my sloppy playing is that you can adjust the rate at which the keys are scanned. It can be set from so fast that almost every other note glitches to so slow that it doesn't pick up notes played (great cheers for this feature from someone who has played an old Lyricon). The pitch and mod wheels at first glance may seem to be strangely placed but once you strap the thing on, you find that they are easy to access, yet won't get in the way when you don't want to use them.

Conclusions

WELL, AS YOU may have guessed, I like this thing. I found it not only fun to play, but also an extremely useful tool that lets your synthesizers come alive. I have got to the point of not liking to use envelope generators to control any part of the sound, opting for some other continuous controller. It was especially refreshing to be able to use the Release Velocity modulation on my Oberheim Xpander. It was equally refreshing to be able to use my own mouthpiece because I (and most sax players) bite down enough to wear holes in the top of mouthpieces. I hate to think that because of that I might have to replace the entire instrument or get costly surgery done that may leave my controller in the shop for an extended period (not to mention the resale value a few years down the line).

The bottom line is that if you are in the market for a wind controller, don't buy until you try this one! That's an order. ∎

EVERY LITTLE BIT

How many bits make a sampler sound good? Unfortunately it's not quite that simple, as sampling quality is dependent on a host of other factors – not the least of which is your ears.

Text by Chris Meyer and Scott Peer.

I T WOULD BE an understatement to say that the past couple of years have seen musical instruments undergo many changes. Only the far-sighted, the optimistic and the naïve would readily recognize today's musical marketplace.

Most advances have been the result of cheap digital technology – sampling instruments being perhaps the best case in point. With the appearance of eight-bit samplers, a sound generation technique became available to the average musician which was previously only available to users of advanced music computers.

Then there was the 12-bit sampler. More recently, the '87 Winter NAMM show brought us affordable 16-bit sampling in the Casio FZI. This has since been followed by the more expensive but equally exciting Emulator III and Prophet 3000 at the Summer NAMM. No doubt, a similar tidal wave of new instruments will arrive in the near future.

Another understatement would be that this proliferation of sampling instruments has left the potential purchaser with a lot of machines to look at and listen to. And listen, indeed – for not only does the normal synthesizer criterion of "new sounds" come into play, but also the new issue of "sound quality."

Unfortunately, the cold yes/no logic of digital electronics doesn't readily translate into sound quality. As you may have noticed, similar methods of encoding (storing sounds digitally in memory), sampling rates (how fast the instrument takes a "photograph" of the sound), and A/D resolution (analog-to-digital accuracy – akin to the smallest object that can be made out in our "photograph") don't necessarily add up to similar sound characteristics. This alone has left many scratching their heads on a few occasions.

So why doesn't an S900 sound like a 2002? They both have identical encoding schemes (linear), A/D resolutions (12-bit), and similar sampling rates. What does E-mu Systems actually mean when they say their Emax, with eight-bit data, is equivalent to 12-bit linear machines? And why do some people (mostly other manufacturers) moan about the Casio FZI not being a "true" 16-bit machine? It certainly uses 16-bit electronics.

A little research and a few test sessions uncovered clues which can be pieced together to solve the mystery. In the process, it'll be necessary to explain the magic that manufacturers resort to in their quest to give us high quality at a low price.

Resolution and Quantization

LET'S START WITH an easy one – why samplers with different resolutions (loosely translated to numbers of bits) sound different. Linear encoding is not only the easist to explain, it's the most common – where the number of bits directly

Illustration Stuart Catterson

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translates into the resolution of the sample. The whole purpose of digital recording is to make a smooth, "round-peg" analog signal fit into a notched, "square" digital hole.

For the technically minded, the number of bits we have to play with translates into how many discrete "notches" we have to fit the sample into. The number of notches (signal levels) may be calculated by raising 2 to the power of "x," where "x" is the number of bits. Another way to translate bits into a more meaningful value is multiplying them by 6dB – this is the theoretical dynamic range (softest to loudest) of the sound we're trying to record.

We'll start by looking at the range of an eight-bit analog-to-digital converter (ADC), which is from 0-255 levels, and has a dynamic range of roughly 48dB. The normal analog signal output by the A/D converter ranges typically from -10 to $+10$ volts, or 20 volts overall. In this case, each bit output from the ADC will represent (20V/255 levels) 78.4 millivolts.

Resolution of this sort presents a couple of problems. For example, what if the input signal to the ADC falls at 5.70 volts? The digital representation output from an eight-bit ADC would fall somewhere between 200 and 201. So, if we represent the 5.70 volt signal with the digital number 200, or 5.68 volts, an error – distortion of the signal – occurs. This error is called *quantization* error. The maximum error is obviously half the resolution between levels (in the worst case of a signal falling precisely between them), and the average error is about one quarter of this resolution. The audible effect is referred to as *quantization noise*. What it translates to sonically is that the sound is audibly less smooth, with the quantization noise sounding like a cross between ordinary noise and a balloon being squeaked.

As the signal level goes down, this sound gets worse. In our eight-bit case, if the input signal was down 42dB from its loudest output (a common occurrence at the tail end of sounds, such as percussion), we have only one bit left to represent the signal. Here the error is practically as large as the signal itself – with a subsequent drastic rise in distortion and quantization noise. This can be heard as the squeaky balloon effect at the end of tom tom samples, for example.

As you can well guess, quantization error decreases significantly as the number of bits in A/D conversion increases. For a 16-bit machine in our theoretical case, the quantization error is down to 0.102 millivolts, which is below the capabilities of most humans' perception. As far as dynamic range goes, the laboratory measured response of undamaged human ears (which doesn't apply to most of us) to music is around 130dB. Practically, it's

around 90dB for most people. Using the formula for finding dynamic range we get $16 \times 6 = 96$dB. We can see that 16 bits more than adequately takes what most music can dish out.

White Magic and Voodoo

IT IS A common misconception of our western minds that dragons, magic and other sorts of compromise are ultimately bad. This is not strictly true and it's not necessary to write off lower resolution machines at this point. One reason is that the cost of producing a "true" 16-bit machine may prevent it becoming a practical consideration. In designing any given instrument, keyboard manufacturers, in addition to dealing with the overall hassle of just running a business, must deal with the issues of user friendliness, manufacturing, marketing and of course, the bottom line – cost, probably the most important to many potential buyers. So, while it's understood that a 16-bit sampler should satisfy us musically, the cost of building such a machine may be prohibitive, depending on the area of the intended buyer – amateur, semi-pro, or professional.

In order to meet the needs and demands of the user, a company might decide that a machine doesn't necessarily have to handle the 96dB dynamic peaks that a 16-bit sampler can. When the signal-to-noise ratio, or dynamic range, is measured in the presence of the audio signal, some claim that about 60dB (10 bits) of A/D resolution are all that are really needed to keep most listeners happy. Several other methods of sample data storage and encoding exist that deliver at least this much range at significantly lower cost. Therefore, some manufacturers resort to these various forms of magic to get adequate sound quality at lower cost. Usually this means less memory (RAM) in the machine. Some of these magics are companding, floating point, and delta code modulation:

• Companding

Next to linear, this is the most common form of encoding. For example, virtually every digital drum machine announced in the past year uses this technique, as does the sound chip in the new Macintosh II computer. This method uses fewer bits (usually 8) stretched over a wider dynamic range by placing more space between the highest levels. To do so, the signal must be compressed upon sampling at the input into an eight-bit (or 48dB) dynamic range. Upon playback, the output electronics have to re-expand this eight-bit signal into something more – typically 72dB (or 12 bits worth). The term *companding* originates from this compression/expansion process. Expansion of the signal is either done in the analog domain (by a circuit similar to the one you'd use to

compress a guitar) or by a special DAC (digital-to-analog converter) known as a COMDAC, which makes this round-to-square remapping of bits to voltage.

Nothing comes free – there is still quantization noise generated when the signal falls at a voltage level in between those the COMDAC represents. By the nature of the system, this error is larger at higher levels (since there is more space between them), but the sheer loudness of the signal tends to cover it. But there is less error at lower levels, which need the higher resolution. The system works best for sounds that are loud for a short section of their overall period, such as drums and percussion. Often, major equalization is still necessary to cover the faults – this takes the form of high frequency boost at the input and corresponding cut at the output designed to cut the quantization noise with the excess signal.

• Floating Point

Another piece of white magic is known as *floating point*. In this case, most of the bits are used to describe the signal as if it were linear, and the remaining bits are used to scale the signal's loudness. Taking our eight-bit example above, imagine adding on three more bits (range of values 0-7), with 0 representing "off," 7 representing "+/−20 volts," and values in between having different ranges. In this way, one can fake higher resolution at lower amplitudes (you always have eight bits to describe the signal). Because the same number of bits have less range to cover, the quantization error (and therefore quantization noise) comes down at low levels. Yes, it takes more hardware and software (and therefore more chance of error) to pull this off, but you can get by with reasonable resolution with fewer bits.

Again for the technically minded, it works like this: a simple hardware translator prescales the signal level entering the ADC circuit and stores this input scaling in, say, two- or three-bit format. This gives a significant cut in RAM costs by representing a 90dB S/N ratio that normally needs 16 bits of linear encoding in 14 or 15 bits of memory. Oddly enough, one of the few commercially available machines to use this scheme – the Kurzweil 250 – actually uses 18 bits – 10 for the signal (there's our 60dB again), and 8 to scale it. The new Kurzweil 1000 series also uses a form of this, and with the current emphasis on higher sound quality, more manufacturers are likely to try this scheme.

The new Ensoniq EPS uses a cross between various techniques and resolutions. It has a 13-bit linear input. Inside, the sounds are stored in 16-bit linear format, and any digital processing is carried out to 16-bit precision (which cuts down on distortion due to math errors).

On output, again the top 13 linear bits are taken, but these are processed with a DCA with 12 bits of loudness resolution. Some call this a form of floating point, but soundwise nothing is gained beyond 13-bit linear. On a theoretical level, you could try to call this 25-bit linear, adding the 13 and the 12, but one does not get 25 bits worth of discrete steps – there are some overlapping regions (ie. sample at 50% and DCA at 100% versus sample at 50% and DCA at 50% – they're the same thing). What this *does* buy the EPS is a claimed 96db of *dynamic* range (from lowest sample and DCA level to highest sample and DCA level) with a signal-to-noise ratio of a 13-bit sampler (theoretical maximum +78db). Any sampler with a VCA or DCA beyond its DAC will behave similarly; therefore, dynamic range may vary among samplers with identical formats by applying this floating-point way of thinking.

• Delta Modulation

Another commonly used method to cut ADC and memory costs is a sampling method called Delta Modulation. This is the storage in memory of the difference in level from sample to sample, as opposed to the absolute value of each one. Several older DDL's along with the E-mu Systems Emax and Emulator II use this scheme. The Emax, for example, uses eight bits to represent the differences of roughly 12-bit resolution samples taken by the ADC. Delta Modulaton not only decreases the amount of memory needed for a corresponding 12-bit linear system by two-thirds, but significantly decreases the cost of the A/D conversion circuit (it is more expensive to accurately translate from audio into digital than the other way around).

Again, there are drawbacks to this approach to sampling. One is the fact that when the input signal increases rapidly, the eight-bit representation often cannot follow it in a single sampling period, and must take several samples to "catch up." This causes an overload in the natural slope of the input signal, which is referred to as "slew rate limiting." Another drawback is the error that occurs when a sample differs
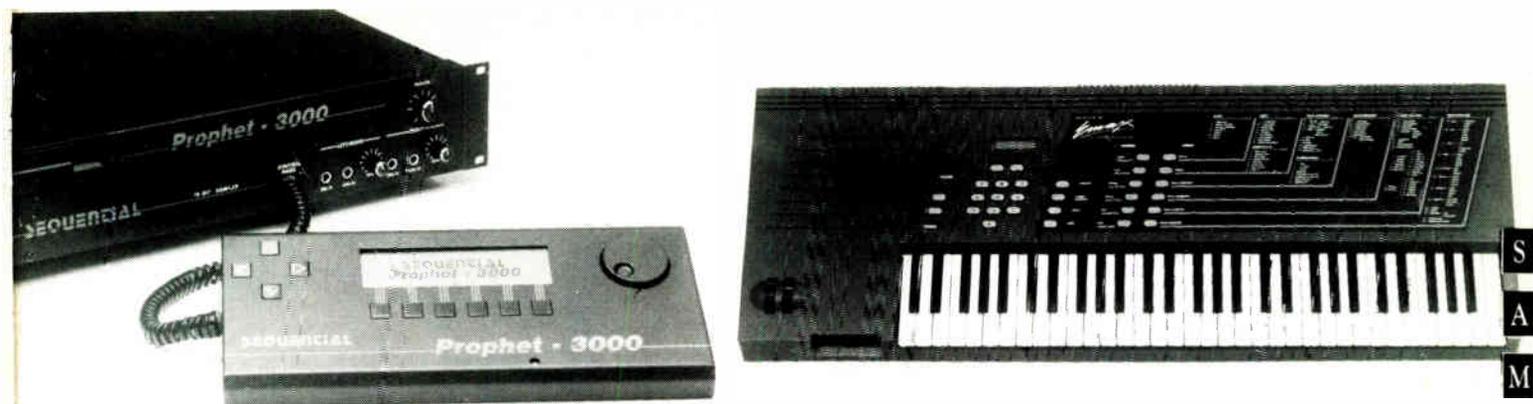
less than a full positive or negative step from the previous sample – the same as the linear quantization problem.

However, these sources of error can be easily overcome. First of all, you can increase the sampling rate – doubling it doubles the slew rate of the machine so that it takes less time to catch up (after all, a signal can only change so quickly). Reducing the minimum step size in the system increases the number of bits and hence reduces quantization noise.

Another reason for differences in the sound quality between various sampling machines is the rate at which the sound is actually sampled. Some years ago Nyquist declared that sampling at twice the bandwidth of a signal would permit the capture of all information necessary for recording and reproducing it. However, in the years since, it's transpired that more than twice the sampling rate, about two-and-a-half times, is actually necessary. So, for the 20kHz audio spectrum the sampling rate needed would be around 50kHz. Interestingly enough, almost all hi-fi applications use a sampling rate less than this theoretically necessary rate. CDs and PCM recorders both use 44.1kHz as their sampling rate (some CDs use only 14-bit A/D too). So, differences in sampling rates of machines with similar encoding systems and resolution can limit the amount of information they are able to extract from the input signal, thereby affecting the sound quality. Try sampling the same sound at two different rates – say, 31kHz and 42kHz – on the same machine, and see if people can really hear the difference.

Keep It Clean

THE INSIDE OF a digital sampler is no place to find pure analog signals. A lot of strange radio and clock frequencies are floating around in there trying to get at and spoil our virgin sound. A lot of attention goes into laying out the circuit boards and electrical shielding. This has a lot to do with how many unwanted noises and disturbances mingle with the sound between the input and output.

And then comes black magic. Digital doobries don't come cheap, friends, so some manufacturers play tricks in

hardware to cut them down. The most common of these is *multiplexing*, which means time-sharing the resources of a machine. The most common case of multiplexing is RAM – instead of having eight copies of a sound to play back eight voices, the voices use the same copy of the sound for a brief instant of time, and make do with this slice until they get another one. Some retain this information in digital form, and then feed it to a DAC. This analog signal must be remembered by the DAC with a sample and hold (S/H) circuit. The quality of these S/H circuits, and how fast they are updated, determines the quality of the output signal. Some instruments even multiplex the DAC's but this requires very fast S/H circuitry.

So far, the discussion has mainly been about playing back a sound at its original pitch. Changing the pitch of that sound introduces a whole new area of voodoo – *pitch transposition techniques*. Some samplers, such as the Prophets, merely speed up or slow down the "sample rate" at which the sound is played back out. Most have some variation of a *constant playback clock* scheme. What this means is that samples are being fed to the output DACs at a constant rate, no matter what is happening to the pitch of the original. When the desired transposition is different than the playback rate, reading off the samples in the original sound don't line up time-wise with trying to play them out. Some – such as the SPI2 and the SPI200 – merely drop samples that hit between the desired outputs. Others use various schemes to guess what the sample might look like at that point in time by using *interpolation* between the nearest samples that would get played. How good this interpolation is has a lot to do with sound quality – the Waveframe AudioFrame is perhaps the most advanced at this. Because there are no published specs for distortion of a transposed sound, use your ears to decide.

It is because of these tricks that a machine with 16-bit ultralinear electronics may end up sounding little better than a 12-bit machine – let alone a CD. But it helps keep the cost down.

So, why do CD players and R-DAT

machines cost less and sound better than samplers? Well, the parts used in a CD player cost less because their development can be shared over tens of thousands of units. Subtly different parts are used in sampling instruments, with only thousands or even hundreds of units being sold to recoup the R&D cost. Eventually, when there are tens of thousands of samplers built, these parts will have been paid for, and samplers should cost less. Admittedly, this is grossly oversimplifying matters, but it gives you some idea of the situation.

Back To Analog

SO FAR, THE differences in A/D resolution and sampling rate have been eliminated (and the different encoding schemes have at least been described) in an attempt to explain why samplers don't all sound the same. So, let's use the same A/D resolution, the same sampling rate and try not to compare apples to oranges by comparing a LinnDrum to an Emulator II. Result? Our samplers *still* sound different. Why, why, why?

The final difference comes down to our old friends noise, frequency response and distortion. Some of these demons have familiar manifestations; others are new.

• Aliasing & Input Noise

As you may or may not know, low pass filters are used in most samplers between the input and the ADC in order to prevent *aliasing*. This occurs when frequencies are present in the input signal which are greater than half the sampling rate, and are mistakenly characterized at a lower frequency. Imagine that pictures are being flashed in front of you, but this time you are also opening and closing your eyes at one second intervals. If your eyes are open for a half second, and closed for a half second, and the picture is changed every half second (once while your eyes are opened, and once while they are shut), you will see only one picture every second. It gets more complicated if the rate of change of the pictures is not sync'd with your own "sample" rate. A similar effect occurs when those high frequencies are input to an ADC and are digitized at a lower frequency. The samples taken are colored.

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Enter low-pass filters. These are placed before the ADC to filter out frequencies higher than those we can sample. However, because the filters are built of "analog" components, there's bound to be noise, frequency response changes, and distortion involved. The seriousness of this depends on the quality of components used, which is reflected in the cost. Also, there may be a certain amount of equalization added at this input stage, to cover the deficiencies of the system or to brighten the signal.

The noise present in this filter circuit is what governs the lowest noise ("noise floor") of the instrument. Even if there is no signal present this is as quiet as your sampler is ever going to be. Therefore it has a significant effect on the resulting signal-to-noise ratio, and from that dynamic range.

The noise floor can be measured by how many bits on the ADC "toggle" with no input signal present. As more bits of noise appear, a greater noise characteristic is recorded along with the desired signal, lowering the sound quality. In other words, the more bits of noise present, the fewer bits are available to record the input signal. And that's one figure we haven't seen listed on any spec sheet.

• Output Filters

The input is not the only place you'll find a filter. "Reconstruction filters" smooth our digital signal back into analog, and other filters are sometimes used for timbral changes – you know, that voltage-controlled job that you used to find on real synthesizers. These too have different frequency response, distortion, and noise characteristics. Their shortcomings are not as noticeable on synthesizers because the sound source – the oscillator – was always running at full volume, masking noise. However, with a sample, changes and colorations to the original are much more noticeable.

So, we're back to the same differences that we had back in the days of analog Moogs, ARPs, Korgs and Rolands – the differences in components and circuits used in sound generating.

For a while, it looked as if the manufacturers should print frequency, distortion, and signal-to-noise specs along with their samplers, and those would tell us which sampler sounded better. This still isn't a bad idea – it would at least give us an indication which sampler was more "accurate." But *better?* Well, which guitar amp, microphone, or synthesizer sounds better? It's time to go trust your ears again: get down to your local music store with a handful of sound sources that you think you'll be using (or CD recordings of them – it's much easier to carry around than a grand piano), sample them into different instruments and *listen.* ∎

ENO*sense*

Musician, producer or philosopher? Brian Eno's involvement in music spans all three, from his successful production of Bowie and U2 to the philosophical meanderings of a very serious composer. *Interview by John Diliberto.*

"T HAT QUESTION, 'WHAT does it mean?' really says, 'What does it symbolize? Well my notion is that art *does* something, not that it means something. Its meaning is what it does. It's a catalyst for some kind of conceptual behavior in the viewer or audience."

Those don't sound like the concerns of a Top 10 record producer, and in many ways they aren't, although Brian Eno has certainly produced his share of popular albums, including two by U2, three by Talking Heads and a few by David Bowie. But producing hit records is not what makes Eno one of the most compelling figures in music today.

Brian Eno is a new kind of musician, a hybrid of technology as philosophy, and sound as music. He's produced prototype new age albums with his Ambient Music Series, avant-garde recordings by John Adams, Michael Nyman and Gavin Bryars,

and punk records by Ultravox and Devo. Eno synthesizes elements of indeterminacy, minimalism, rock and electronics into a Unified Field Theory of music.

In the last sixteen years he's cut a swathe across most major trends in music. He was an original member of Roxy Music, the pioneering British rock group with Bryan Ferry, Phil Manzanera and Andy MacKay. They extolled sexual ambiguity and future shock, making music for the Orwellian year 1984, in 1972. One famous photograph of the era captures surrealist painter Salvador Dali having tea with an androgynous Eno, in makeup and shoulder-length hair – the epitome of *A Clockwork Orange* decadence. "I still get people writing to me asking me why I don't wear makeup or things like that," laughs Eno.

Drinking tea in the London apartment of his manager, Anthea Norman-Taylor, it's difficult to believe this charming, exceedingly articulate man with a receding

hairline and preppy clothes is the same person who snarled out the lyrics to 'Needle in the Camel's Eye,' 'Dead Finks Don't Talk' and 'Baby's on Fire,' songs from his first album, *Here Come the Warm Jets*. "Yeah, nearly everything I did before last week seems like it was done by another person, I have to say," says Eno with bemused detachment. "And in a way they are, you know. There are so many voices in you all the time, and circumstances are calling one of those people to the fore or another one. The mix is always being adjusted within you. I always think of myself like a sort of 48-track tape and it's always possible to re-mix that and circumstances will be re-mixing that all the time. So I can still recognize in me some of that person, but the emphasis has shifted a lot. He was a very extreme mix of the person, the whole bunch of possibilities, that makes me. He was one extreme mix."

That person left Roxy Music after their

second album, *For Your Pleasure* and embarked on a solo career that has been nothing if not diverse. His first two solo records, *Here Come the Warm Jets* and *Taking Tiger Mountain (By Strategy)* employed all-star musicians playing songs that ranged from the whimsical 'Cindy Tells Me' to the demonic 'Third Uncle.'

By his third album, 1975's *Another Green World*, Eno had changed, and it was reflected in the music. No longer the decadent art trollop, he was more like a Zen master. Robert Fripp was one of the guitarists on *Another Green World*. "Brian

> *"I have a grand piano at home which I never play other than to sit in the room bashing one note. I hit a note and I listen to it decay . . . it never is uninteresting to me."*

doesn't really have a very strong musical background in terms of the craft of music," says Fripp. "But what he does have is good taste. He has good taste and a perception of what's right that very, very few musicians have. So working with Eno, it's refreshing to hear the few notes, but right, rather than the many, many, many that are wrong from most musicians of my acquaintance."

Jon Hassell agrees. He and Eno have collaborated since his *Fourth World Volume 1: Possible Musics* album in 1980 and continue to produce together. "I think he has an extremely good sense of when things are boring and when things are exciting and how long things might go on. Perhaps because he's on the outside of it slightly. You know he'll be the first one to tell you he's not a 'professional musician,' thus giving himself a domain in which he can operate from a non-specialist point of view."

On *Another Green World*, Eno became what he called a "non-musician." By that he meant he didn't have a great deal of proficiency on traditional instruments, but that his medium was the manipulation of sounds and instruments through synthesizers and the recording studio.

"By saying I was a non-musician I certainly never meant to imply that musicians were therefore uninteresting," explains Eno. "What I meant to imply was that there were new possibilities about which required new talents and they were not traditional musical talents. And the use of the studio was one of those things. Because at the time, this was in the early '70s, that was just really the beginning of the 24-track studio and extensive processing and all of the things that we now regard as standard studio practice. Well, those were technologies and techniques that really weren't anything to do with traditional musical skills but were of course of great interest to musicians. Well, what happened with a lot of those virtuosi that you're talking about was that we formed an extended musician. They could do what

they already knew how to do. They didn't really have the familiarity with technology, the concept that the technology was more than simply a decorative transmitter of what you were doing, but that it was a part of what you were doing."

For most of the seventies Eno worked out this concept on his own albums and in collaborations with other musicians, notably Robert Fripp, who was then the guitarist with King Crimson. They recorded two albums of tape loop duets *No Pussyfooting* and *Evening Star*. "When I worked with Fripp we became one musician," recalls Eno. "It wasn't a case of two musicians playing together. It was a case of me hearing what he was doing and somehow extending that by the things that I knew were possible with studios and with synthesizers."

Eno began working more and more in this fashion, finding musicians who were open to his manipulations and concepts of sound. West coast pianist and composer Harold Budd is another musician with whom Eno worked, on the albums *Plateaux of Mirror* and *The Pearl*. Eno's manipulations processed Budd's piano into haunting audio illusions. "It's a role that I've done with Harold Budd quite a lot as well. It's a nice position for the musicians because they are responding to what I'm doing at the same time. It's not something that is 'Well, you play your bit and then I'll tart it up in the studio.' It doesn't usually happen like that. What usually happens is that they're playing and I am doing something with the sound at the same time, and they're hearing that and, of course, their playing is then a response to that. So that the circuit is a live circuit, if you see what I mean. That's why I say it's like one musician rather than two."

"I was listening to the treatment at the same time that I was playing," concurs Harold Budd, "because that colored and influenced what I actually did play. So it was all at the same time."

Budd described one of the processes Eno used to get those ephemeral piano clusters. "Basically it's just a feedback loop going through several chorus boxes. So a certain sound is given additional chorus effect and then before it ever gets out to the board, it has several more at the same time. And so it's just a matter really of using the faders on the mixing board. If the faders are way up you get a very, very huge complex harmonic scheme, just because everything that is otherwise inaudible is amplified tremendously because of the chorus effect."

Budd was part of Eno's *Ambient Music*, a series of records concerned with environmental sound. It grew out of his

enthusiasm for minimalism and his own 1975 album, *Discreet Music*. Eno placed two Revox tape recorders side-by-side, recorded music from a synthesizer onto the first, threaded the tape to the second, which played back the music and routed the signal back into the first deck in a continuous cycle. It was a variation on the processes of Terry Riley's *A Rainbow in Curved Air* and Steve Reich's *It's Gonna Rain*. Eno has always been influenced by minimalism. The first work he performed in public was "X for Henry Flynt," an hour-long piece by LaMonte Young, where the forearms are pounded on the piano.

"*It's Gonna Rain* was very, very important as well and both of them had a lesson for me that I've never forgotten, which is that the relationship between input and output is a very complex one within a piece," says Eno with deference. "*It's Gonna Rain* uses a very, very small amount of original material but it produces a very complex shifting output. It interested me that the artwork could be a system of amplifying detail, amplifying by analysis in a way. And for me what's interesting about minimalism is not that people use very few elements but that very few elements can mean a lot.

"Minimalism makes it very clear that listening is not only a very active process, but it's a very creative process as well. When you're listening to *It's Gonna Rain*, if you're enjoying it, what you're enjoying is your own perceptual processes doing something. They're reconfiguring that material. They're making constructions of it. They're comparing this moment with that moment. They're filtering things. They're amplifying other things. So really a lot of what's happening with minimal music is not so much to do with you looking at a work operating outside of yourself, it's to do with you looking at your brain operating on something and that's a very fascinating process.

"Now, to a composer like myself of limited technical resources, this is good news because it means that not only do I have the technologies that I'm used to using, like recording studios, synthesizers, at my fingertips. I also have this big device, the human brain, which I can also somehow make use of as part of the work."

The outgrowth of all this was a series of records called *Ambient Music*. With pianist Harold Budd and zither player Laraaji, he recorded environmental sound pieces that could be used as the proverbial sonic wallpaper, or listened to closely for their subtle changes. *Music for Airports*, *On Land*, *Day of Radiance* and *Plateaux of Mirror* were all harmonically simple, melodically refined albums that created their own sound world.

"I meant that as a prescription for composers to think of their own sound as environmental sound," says Eno. "To think of what they were making as part of the environment, just as the things we're

listening to now are. Composers were still making music as though people were buying the record, rushing home, putting it on and sitting in front of their stereo with their ears glued in the way that one watches a film or something like that. Now, I'm sure you'd agree that that isn't the common experience of people listening to music any more. Music has become part of the tapestry of your life like lighting is or like the environmental sound that you hear anyway. So although people do still listen to music like that, a lot of the time they're not doing so. I was excited by the idea of making music that acknowledged that and said 'Here's a music especially for that. Here's a music that is intended to merge into the environment.'"

You might ask why Eno didn't just record environmental ambiences like 'Dawn at Okeefenokee Swamp,' 'Thunderstorm' or 'Subways at 42nd Street.' Well, he often did, they just didn't sound that way after he got finished processing them. "In On Land, I think on nearly every track there's environmental sounds but it's quite often so processed that it isn't very obviously environmental. It's, for instance, slowed down a great deal or treated or mixed in with electronic sounds.

"Another way of using environmental sounds I worked with for a while was recording things through specially constructed strange microphones. So this would be an ordinary microphone with a big tube on the end, for example, and sometimes the tube would go out through a car park and back into the studio. So there would be, theoretically, some resonance from the outside world affecting the sound as it went down. These experiments were limited in scope and not all that successful." He laughs in the self-deprecating way that makes Eno so eminently charming.

OUR DISCUSSION OF tape loops and repetition brought one of Eno's Oblique Strategies to both our minds: "Repetition is a form of change." In 1975, Eno and artist Peter Schmidt developed Oblique Strategies, a series of over 100 cards, each with an epigram printed on it. Whenever Eno would reach an impasse in the recording studio, he would pull out an Oblique Strategy card which might say "Discover the recipes you are using and abandon them," "Make a sudden destructive, unpredictable action," "Honor thy error as a hidden intention," or "Tape your mouth," and do whatever it said.

"So the Oblique Strategies were really a way of getting past that panic by reminding myself that there were broader considerations than the ones I could remember that moment in the studio," explains Eno. "So when I got into a panic

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of some kind, thinking, 'Oh dear, where is this going? It's not going anywhere. Oh, this just sounds like what I was doing two years ago' – all those kinds of things that frighten you. I'd pull out one of those strategies and it would tell me something, really. It would say, 'Okay try this course of action,' and I was quite religious about them. I used to absolutely drop everything and follow that course of action, so I didn't pull them out lightly because I knew it could mean jettisoning whatever I was doing at the time to do something completely bizarre, sometimes like take a long walk or something – the last thing you want to do if you're panicking about not doing anything that day."

Eno doesn't carry the cards with him any more; they're embedded in his consciousness. But Another Green World was heavily influenced by their use. Oblique Strategies are similar to the I Ching as used by Cage, in that they took the decisions out of the composer's hand. Eno described how one piece on Another Green World, 'Spirits Drifting,' was shaped inadvertently by an Oblique Strategy.

"When I started making that piece I was really at the end of my tether," says Eno. "I'd been working on it for the whole day and it seemed that we had almost nothing on tape and it sounded like a piece of crap to me at the time. It really sounded awful. And I was . . . this is the truth. I was

feeling for rock music I think. It was a sort of industrial extrapolation of what was going on in soul and funk records. It had a much more European feel than those things had.

"On the other side of the record, of course, was another direction. It was one that I think I had already taken. But it was very much the landscape direction. One side was urban and industrial. The other side was suburban." Eno laughs again.

He teamed up with David Byrne and Talking Heads on their second album in 1978, *More Songs About Building and Food*, and their collective interest in African and funk rhythms and unusual studio ambiences created the most exciting rock music of the early 1980s. And Eno's work with U2 has been monumental in sound and popular impact. Working with his co-producer of the last several years, Daniel Lanois, they created the audio stage where Bono and The Edge generated their psycho-social theater.

THROUGHOUT ALL THE music, all the changes, whether producing the largely acoustic avant-garde Obscure label or the African group Edikanfo, Eno has maintained a lingering reputation as a synthesizer wizard. It's a reputation Eno really doesn't want.

"For me the problem with synthesizers has always been that the sound that you hear is a direct result of the movement of a very small number of electrons," he confesses. "Therefore the regularity and the evenness of the sound are awe-inspiringly boring. A grand piano is the result of the movement of so many factors, so many environmental, climatic, physical, geological factors, that in fact, a piano never sounds the same twice.

"I have a grand piano at home which I never play other than to sit in the room bashing one note. I hit a note and I listen to it decay. And I hit the note again. I listen to it decay. And I think I spend a couple of hours a week hitting that note.

▶ standing at the synthesizer crying as I was playing it because I thought 'I don't know what I'm doing.' And I took out an Oblique Strategy and it said 'Just carry on.'"

He rocks back in his chair, laughing at the recollection. "And the first thing was that I pissed myself laughing because it was such a low level answer to what I was expecting. You know, I was expecting something that would have me sitting down and scratching my head. It said 'Just carry on.' So I just carried on and in the next half hour or so that piece suddenly gelled into something, and in fact it gelled into something that I still like very much."

Over the years, Eno has been surrounded by musicians and artists of like mind, who look at music as an adventure, not a calculated product. At first, Eno worked with lesser-known musicians, often on their debut recordings. In the late seventies he produced the *No New York* anthology, Devo's *Are We Not Men? We Are Devo!*, Ultravox's *Ultravox!*, and John Cale's semi-comeback, *Fear*. Towards the end of the decade he began working with more established artists, notably David Bowie

during his controversial *Low* period in West Berlin. They recorded a trilogy of albums together, *Low*, *Heroes* and *Lodger*, that sank Bowie deep into Eno's thickly textured sound-pieces. The second sides of *Low* and *Heroes* were dense atmospheres orchestrated by Eno in the studio. The opening sides, however, were a new kind of rock sound, with sharp, angular rhythms, choppy guitar

chords and fragmented lyrics. Listen to the drum sound and you'll hear the origins of the Phil Collins gated-reverb attack that became so ubiquitous in the 1980s.

"Both sides of that record are new directions in a way," says Eno reflectively. "Well, of course the first side – just on a purely technical level – the drum sound on the first side of *Low* became *the* drum sound for the next X years. I mean it still is now to some extent. But the rhythm section feeling altogether was rather a new

"Synthesizers interest me more than samplers . . . because they do introduce new sounds into the world, and in working with them I learn a lot about how sounds are made up."

I've got one, it's a G-sharp on my grand piano, that I just love the sound of, and it never is uninteresting to me. There's always little exotic overtones that I haven't heard before that are oscillating in different ways. Now, my reservation about synthesizers has been that they aren't like that. You wouldn't hit a G-sharp on a DX7 for two hours and expect the same kind of excitement from it."

His whole approach to synthesis has been to get it to sound more alive and textural,

which is what he does in the studio. Using everything from tape manipulation to signal processing is a way of creating a liveliness and presence of sound.

Eno uses a Yamaha DX7 extensively, but he still won't use samplers, even though they have acoustic sounds and waveforms. "I'm not very interested in them," says Eno. "Conceptually, synthesizers interest me much more than samplers do. A sampler is a tape recorder as far as I'm concerned, and it isn't conceptually very much more interesting than a tape recorder. Synthesizers, however, do interest me, for two reasons. One, because they do introduce new sounds into the world, and the other is because in working with them I learn a lot about how sounds are made up. The DX7 has been very useful for that. I use it almost as much as a research tool for seeing how a sound is made. What happens when this hits this? Why does this sound like that? You find that a very specific relationship between two operators produces something that sounds like a grand piano. And you think, 'I wonder what it is in the physical makeup of a grand piano that demands precisely this relationship for its imitation,' as it were. I'm not interested in imitating grand pianos per se, but I am interested in finding out how sounds work.

"So my solution has been to make the equipment unreliable in various ways. I used to like the old synthesizers because they were like that. My first synthesizers, the EMS and the AKS, the suitcase one that followed, the early Minimoog – they were all fairly unstable and they had a certain character. I mean character has really to do with deviations, not with regularity. They were very Latin in that sense. And then, of course, I used to feed them through all sorts of devices that also had a lot of character, that were in various ways unpredictable. And the interaction of a lot of these things started to create sounds that had an organic, uneven sound to me."

Although it's difficult in the computerized, digital synthesizers like the Yamaha DX7, Eno has found a way to introduce some character into that as well. "I've found ways to de-stabilize the DX7 a little bit to create interactions between it and other instruments that are more interesting," says Eno with a gleam in his eye. "I don't have very good voltage supply, for instance. Within the patches, I build in certain elements that don't repeat. For instance on the original DX7, the level of envelope generator four, there's something wrong with it in the basic programming of the synthesizer and you can use that to create non-repeating patches. If you have that set to under 50, you'll find that the synthesizer doesn't quite know how to handle it and it behaves unpredictably. They've sorted this out unfortunately on

the second generation, so I still use the first one, and that's an important element of quite a few of my patches."

All of these experiments and the music coming out of Eno's new 24-track home studio have been heard by very few people. The only places they can be heard are at his video exhibitions. Eno is bored with making records. "Records haven't got quite that *frisson* of novelty that they did have for me fifteen years ago, or ten years ago," he admits.

He's been concentrating his time on multiple-screen video installations set in unusual locations, like botanical gardens or churches. But he still makes music for them, using multiple auto-reverse cassettes.

He has from eight to 48 channels of sound and music, running non-stop in random synchronization. Coincidentally, one of Eno's early influences, John Cage, is currently working with a similar system.

"This is so much closer to the feeling that I wanted in ambient music," says Eno. "I want the notion of something that was steady-state in the sense that it was always pretty much reliably similar, but it was never exactly the same. A little bit like any natural process, like watching a river. It doesn't pull many really big surprises on you, but at the same time it never repeats itself perfectly. So I wanted to make some music that had that homogenous but ever changing character to it." ■

Kurzweil 1000

Digital Sample Player & Wavetable Synthesizer

Though it's not exactly a less expensive version of the K250, the company's latest offering does have the same high quality, preset sampled sounds, as well as a number of other interesting features. *Review by Chris Many.*

FOR MOST MUSICIANS, products from the Big Three of synthesizer manufacturers, New England Digital, Fairlight and Kurzweil Music Systems, have been slightly out of reach due to the hefty price tag. Even the Kurzweil 250, the least expensive of the lot, is well into five figures for a full configuration. Granted, each music system offers an incredible amount of bang for the buck, but many musicians just don't have that kind of money. Fortunately, Kurzweil has begun repackaging their successful K250 technology into several new formats, the RMX250 (a rack-mounted version of the big unit) and the 1000 series, a scaled-down version of the 250 minus the sampling, sequencing and sound block upgrade options. All of these products are considerably less expensive than their big brother, and the 1000 series is really in line pricing-wise with Yamaha, Roland, et al. So does this mean that the incredible Kurzweil Grand Piano is now within reach of everyone's budget?

Well, almost, and a lot more sounds and features as well. The 1000 series comes packaged either as a rack-mount unit, the 1000PX, or with a keyboard. It's a 76-key master controller, too, not your normal run-of-the-mill five-octave plastic variety. I worked with the keyboard version, which comes with the Professional Expander sounds. Rack-mounted versions will also soon be available with other sounds. The Guitar Expander (GX), the Horn Expander (HX) and a String Expander (SX) are each oriented to a particular sound group, and include many of the 250's resident sounds, relating to that group. The

Professional Expander covers a wider range of instruments, so you can choose to get a basic building block of sounds that includes pianos, strings and horns, although not everything that's included in the specialized expanders.

The Low Down

SO HOW DOES the 1000 differ from the 250? Well, first and most important, it is not a sampler and there is no sampling option, period. It *is*, however, a digital sample player, all the samples being ROM based. Consequently, there's no loading of sounds or long waits for disk drives, because each sound is immediately available. You should think of the 1000 as a synthesizer, not a sampler, although the sounds you'll be working with are all digital samples or digitally-encoded waveforms. Keep in mind though, these are not your usual "factory samples" that require further tweaking; these are the same finished sounds you'll find in the KMS 250.

Some other major differences are apparent as well. First, there's no on-board sequencing, which on the 250 has become very sophisticated, and there's no space to expand your unit with additional sounds. (Although the idea is that for about the same price as 250 owners pay for a sound block upgrade, you can purchase an additional Expander – the GX, HX or SX.) There are also some big differences on the plus side. For example, the 1000 is 24-voice polyphonic (the GX, HX and SX have 20-note polyphony) as compared with 12-voice polyphony on the 250. Kurzweil has been a pioneer when it comes to multi-

timbral operation, and the 1000 maintains that tradition. The instrument is completely multitimbral – it can simultaneously operate on up to 16 MIDI channels – and you can either have the voices dynamically allocated or you can pre-assign the number of voices you need per patch. Thanks to the large polyphony and voice assignment scheme, there are plenty of high-quality samples you can have going on at once before the voices start to steal from one another. And of course the other big plus is the lower price.

The interface on the 1000 is via only eight programming buttons and a 32-character LCD (two lines of 16 characters each). It's an approach you'll either love or hate because of its simplicity as an interface but complexity in menu design. You don't have to learn a whole lot of buttons or combinations in order to program or play the 1000, but you can find yourself feeling a little overwhelmed with the myriad of possibilities when it comes to programming. When you can have up to four layers, plus splits, effects, bipolar envelopes, and lots of other modulation of the samples accessed by only eight buttons, things can get a bit confusing. That is, at least until you've learned your way around the menu options (which the excellent owner's manual really helps you to do).

Operation

TWO MODES ARE available on the 1000: Play mode and Edit mode. Play mode is very straightforward; it allows you to select and play any of the 128 available factory preset patches (or Programs in Kurzweil

terminology) or any of the 64 user-defined Programs which can fit within RAM. Using the Program buttons (Up or Down) you can cycle through the factory and user presets very quickly, although I wish you could just punch in a number for instant access, rather than cycle from I through 50 to get to program 51 (although to be fair, you can jump 64 sounds at a time and the scrolling function is very fast).

For the most part the factory presets are quite usable, with well-thought-out splits, good effects programming and fine use of the layering of different sounds. The available sounds depend on the unit you're working with, but on the PX unit, sounds like Grand Piano, Strings, Choir and other rich acoustic instruments are available. The Trumpets leave a lot to be desired on the PX (it's the basic Horn sounds from the 250, not the upgraded Horn block which I assume is what is included on the HX unit), and it would have been great to have at least one drum kit available in ROM. The sound of this unit belies the cost, however; it sounds just like its bigger brother and for a fraction of the price.

The assignment of MIDI channels to different Programs is also done in the Play Mode. The 1000 defaults to playing whatever program is currently visible in the LCD regardless of its MIDI channel assignment, so it's very simple to choose both channels and sounds. Three MIDI Modes are available; Omni On (Poly), Omni Off (Poly) and Multi Mode. Multi Mode allows you to play different sounds on different MIDI channels. When in Multi mode, each MIDI channel is either assigned or inactive.

If you're only interested in the 1000 series' preset sounds, this is as far as your learning will need to take you. It's really quite simple to operate this instrument. If you're so inclined, however, the editing aspect of the 1000 is very complete and gives you complete control over the layout of sounds and effects, including editing an intonation table, pan locations, and compiled and modular effects. By combining

one's own effects or modulations for the soundfile(s) in use.

Once you've entered Edit mode, the same set of eight buttons allows you to navigate around the Edit menus, and once you get the hang of it, it's not half as complicated as it seems. All options are reached by moving through the menus. Once you've cycled through the options and found the main topic you're looking for, you can then access the submenus underneath it. In its easiest form (using compiled effects), there are six main menus; when you use modular effects this figure is tripled, so you can see just how much power is being placed in the programmer's hands.

The first set of parameters you'll need to know about are those which affect the instrument globally. Things such as master tune, basic MIDI channel, transposition, etc. are covered in the Master menu. Your overall sensitivity is set using one of the velocity maps included. There are 13 preset velocity maps for you to choose from (including different playing styles using the K1000 controller, or a DX7), or you can edit and save velocity settings of your own. These are linear velocity maps, meaning you set the loudest and softest points and the Kurzweil calculates out six additional levels in between, assigning the softest as ppp and the loudest as fff. If you want a non-linear map (eg. velocity values from 60-70 will play very loudly, whereas 70-80 will play softer than that), then you can directly edit each of the eight groups.

Another global map you can produce is a MIDI map, enabling you to assign programs to different external patch numbers. For example, if you wanted to have program 64 called up when you pressed patch button #1 on your keyboard controller/synth, you would program it to do so via this MIDI mapping technique.

Taking a cue from the recent interest in tunings, there is also an intonation table that contains 17 different tunings, from Classical Just Intonation to Tibetan, or Indian Raga tunings to Werckmeister. Each

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Multitimbral *"Thanks to the 24-note polyphony and dynamic allocation voice assignment scheme, you can have plenty of high quality samples going on at once."*

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waveforms such as sine or square waves with digital samples, you can create some very different Programs.

Editing

A PROGRAM CONSISTS of a soundfile, which is either a collection of samples or a digitally-encoded waveform, layer parameters, and compiled or modular effects. Compiled effects are preset effects that we're familiar with like chorus, vibrato, phasing, echo, etc. Using modular effects, a great deal of control is available to create

of these tables preserve octave relationships (all C's are spaced 12 semitones apart), but the intervals between the octaves are tuned separately. When you edit or create your own intonation table, tones are relative to the tonic, which you choose. In other words, you are restricted to a 12-note scale, with octaves at either end, and the intervals in between are defined as Minor second, Major second, Minor Third, etc. Tuning of each interval is ±1200 cents (a hundreth of a semitone). You can name and save this ▶

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just as you can any of the globally assigned maps/templates.

Another option included here is MIDI mapping of splits and polyphonic limits by MIDI channel. By setting high/low note ranges you can adjust just what programs will play where on the keyboard and what their MIDI channel assignment is.

Other parameters set in this portion of the Menu are things like MIDI status, Mono or Stereo mode, key stealing, and pitch-bend range. Speaking of which, for some reason (I'm sure somebody has a "reasonable" explanation for this) Kurzweil limits the range to a maximum of

relative and keyboard tracked volume are also parameters within each layer. Stereo positioning, velocity adjustment, balanced crossfades and many more options are completely under the programmer's control, and it's surprisingly easy to work with, given the variety of parameters you can adjust for each layer.

And then there are the effects and synth controls. As mentioned above, two types of effects are available: compiled – which is basically a preset list of effects like vibrato, tremolo, etc, and modular – which essentially gives you control over familiar synth modules such as LFOs, Envelopes,

local LFO's, two global LFO's, two local Envelopes (ASR), two global Envelopes, two Mixers of two inputs each (allowing assignment of selectable control sources), two Invertors and Negators (explained below), amplitude envelopes, including looping of attack segments both forward and back, and pitch and amplitude controlled input. Quite a mouthful of modular effects for a sampling playback instrument. The Invertor/Negators give a good representation of just how far the analog/modular synth analogy has been carried into this digital format. An analog signal on an oscilloscope has maximum and minimum coordinates which determine its waveshape. Invertors and negators invert and reverse these signals respectively, generating new waveforms or new modulations based on the original.

Effects *"The flexibility in the envelopes rivals the best of traditional synths, and . . . the effects you can generate digitally would make you think there was a sophisticated processor on board."*

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six quarter tones, a minor third, up or down. Why such a small bend range, I don't know. And before we leave the subject of minor complaints, there's one more about the keyboard implementation: *no* aftertouch. The keyboard on the K1000 is great otherwise. It has a good solid feel (more like that of Roland's MKB-1000 than say, Yamaha's KX88), but no aftertouch. The Kurzweil 250 is the same way, and you can get used to working without it, but why omit something that commonplace in this day and age? Possibly, it was to give the keyboard the feel that it has, as pressure-sensitive keyboards all have that "springy" feeling, which there is none of with the K1000. Admittedly, keyboard feel is really a subjective kind of thing, but I like the keyboard a lot.

More Editing

LAYERING OF DIFFERENT sounds is not anything new to samplers or synthesizers, but it takes on a new meaning when you're dealing with 20 or 24 voices. Playing a piano with two layers (hard and soft pianos) adjusted to respond to velocity sounds a whole lot better when you don't have to worry about half of your notes being stolen – which can happen if you try something like this on a sampler with eight or even 12 voice capacity. It also means that creating a program with four layers of different, similar or even the same sounds being triggered from one note won't give you just a few notes polyphony. Layer functions include duplicate, which creates a copy of the previous layer you worked on, and import, which copies *any* layer from any program. It's a great shortcut when you have several favorite layers across different programs. You can also clean a layer of compiled or modular effects, leaving you with a new slate to begin with (or re-edit). Layers can be defined to ranges, much like channel splits were outlined.

Transposition, detuning, delays and

etc. All or none of these effects can be assigned to all or none of the layers with which you're working. When dealing with effects, it's one or the other, either compiled effects or do-it-yourself by the modular method. You can, however, have a program with modular on one layer and compiled on another. As an example of the 1000's menu structure leading you where you need to go, when compiled effects are used, you cannot access the modular effects menus.

When dealing with the modular effects menus, you can deal with local or global modules. Local modules deal only with the layer they are currently assigned to, global covers all layers. Local modules can be programmed to give independent and individual control over each of the keys played in a layer. For example, if you had an LFO programmed to create a tremolo effect, each note would start the LFO independently of the other notes pressed, while the other notes continued to function on the original triggering of the LFO each of them had created. There's a lot of routing that can be done here, and the inter-relationships between layers or velocity curves can generate some interesting (and useful) sounds.

Here's another example. Let's say you had an LFO controlling Vibrato. By routing the LFO level to a Velocity Trigger, you can control the amount of vibrato you hear by how hard you strike the key. By routing the LFO rate to another Velocity trigger, you can adjust the speed of the vibrato through your touch as well. By adjusting the high/low points of the velocity, you'll also establish at what points the vibrato will turn on or not be available. And although the K1000 doesn't itself have aftertouch, the 1000 series will respond to aftertouch information from another controller or synth, not only mono aftertouch, but polyphonic aftertouch as well.

Included in the modular effects are two

You begin to see the degree of programming available on the 1000 series, even though it's not your normal digital or analog synth. The flexibility in the envelopes rivals the best of traditional synths, and although there's no built-in reverb units, the effects you can generate digitally would make you think there was a sophisticated processor on board. Granted, to create effects like this you'll have to sweat a bit over the various modules – it would have been much simpler to dial up Hall or Reverse reverb patches – but the effects are there for you to program, if you're so inclined.

Conclusions

IT'S WONDERFUL TO see Kurzweil repackage many of their great sounds into more affordable units. You can't find a better Grand Piano for this kind of money, and the quality of the samples (for the most part) is excellent. The K1000 (with keyboard) is only a few hundred dollars more than the rack-mount, so don't kid yourself; if you can live without aftertouch and like the feel of a solid controller, spend the extra money and buy the keyboard version. It's a steal. (Although I can't help wondering why it is that the rack-mounted version is so close in cost to the keyboard version. It would seem to me that there should be a larger pricing distance between the two.) It would help to have a software extension to save and edit programs with, and Kurzweil normally has something like this available on the Mac. If it's not out yet, it probably will be soon.

So, if you've always wanted one of those ultra high-end samplers, but couldn't afford it, here's the next best thing: Kurzweil sounds with comprehensive editing features. ■

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PATCH WORK

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P R O G R A M M I N G

This is the page where MT's editorial team invite you, the readers, to demonstrate your own synthesizer programs. Send us your favorite sounds on a photocopy of an owner's manual chart (coupled with a blank one for artwork purposes), accompanied by a short demotape. Please include a description of each sound and write your full name and address on each chart. If we publish your patch, you'll be rewarded with a complimentary one year's subscription to MUSIC TECHNOLOGY. Interested? Then get twiddling and get scribbling!

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

ENSONIQ ESQI
O'Hear
John Shinko, Nanticoke, PA

Named after bassist/synthesist Patrick O'Hearn, this is a mellow bell-like piano sound, good for general keyboard work. It has a nice sustain quality and would probably sound best in new age compositions. ∎

	OCT	SEMI	FINE	WAVE	MOD1	DEPTH	MOD2	DEPTH
OSC1	+0	00	05	SQR 2	LFO 1	+04	ENV 1	00
OSC2	+0	00	05	SQR 2	LFO 1	+05	ENV 1	00
OSC3	+0	00	07	SAW	LFO 1	+5	ENV 1	00

	LEVEL	OUTPUT	MOD1	DEPTH	MOD2	DEPTH
DCA1	63	ON	OFF	00	OFF	00
DCA2	63	ON	OFF	00	OFF	00
DCA3	63	ON	OFF	00	OFF	00

	FREQ	Q	KEYBD	MOD1	DEPTH	MOD2	DEPTH
FILTER	035	00	14	VEL	+43	VEL 2	-15

	FINAL VOL (ENV4)	PAN	PAN MODULATOR	DEPTH
DCA4	63	08	ENV 3	+63

	FREQ	RESET	HUMAN	WAVE	L1	DELAY	L2	MOD
LFO1	20	OFF	ON	TRI	00	01	00	LFO 1
LFO2	12	OFF	ON	TRI	63	00	20	LFO 2
LFO3	62	OFF	OFF	TRI	56	00	20	LFO 3

	L1	L2	L3	LV	TIV	T1	T2	T3	T4	TK
ENV1	+63	+00	+00	63	00	00	08	00	00	01
ENV2	+63	+00	+00	00	00	00	00	00	00	00
ENV3	+63	+30	+01	59	22	00	28	47	37	16
ENV4	+63	+47	+00	29	63	00	41	63	44	09

	SYNC	AM	MONO	GLIDE	YC	ENV	OSC	CYC
MODES	OFF	ON	OFF	00	OFF	OFF	OFF	OFF

	SPL/L	S/L PROG	LAYER	L.PROG	SPLIT	S.PROG	S.KEY
SPL/L							

CASIO CZ101
Multi-purpose
Todd Barton, Ashland, OR

Here's a multi-purpose patch with potential! The seven available sounds are accessed by toggling Ring/Noise and Line Select on the CZ, creating a very usable combination of sounds. Todd suggests that you take the time to explore the Detune Octave setting as well for more possibilities.

– Line 1+1', Ring Off, Noise Off: *soft xylo sustain, good for sequences.*
– Line 1+1', Ring On, Noise Off: *flute xylo, good for leadlines.*
– Line 1+1', Ring Off, Noise On: *noise shot, useful for percussion tracks.*
– Line 1+2', Ring On, Noise Off: *chiff bass sound.*
– Line 1, Ring Off, Noise Off: *perc flute.*
– Line 2, Ring Off, Noise Off: *rubber band chiff.*
– Line 1+1', Ring On, Noise Off, Detune –1: *plucked piano in lower register.* ∎

LINE SELECT

1+1

(1,2,1+2,1+1')

MODULATION	
RING	NOISE
OFF	OFF

(ON/OFF)

DETUNE			
+/–	OCTAVE	NOTE	FINE
+	2	00	00

(+/–) (0~3) (0~11) (0~60)

VIBRATO			
WAVE	DELAY	RATE	DEPTH
1	15	38	7

(1~4) (0~99) (0~99) (0~99)

OCTAVE	
+/–	RANGE
0	

(+/–) (0~1)

1

DCO 1

WAVE FORM	
FIRST	SECOND
2	0

(1~8) (0~8)

E N V (PITCH)	STEP	1	2	3	4	5	6	7	8
	RATE	99	76							(0~99)
	LEVEL	62	00							(0~99)
	SUS/END		END							

DCW 1

KEY FOLLOW
0

(0~9)

E N V (WAVE)	STEP	1	2	3	4	5	6	7	8
	RATE	99	41							(0~99)
	LEVEL	83	00							(0~99)
	SUS/END		END							

DCA 1

KEY FOLLOW
2

(0~9)

E N V (AMP)	STEP	1	2	3	4	5	6	7	8
	RATE	78	38							(0~99)
	LEVEL	99	00							(0~99)
	SUS/END		END							

2

DCO 2

WAVE FORM	
FIRST	SECOND
4	0

(1~8) (0~8)

E N V (PITCH)	STEP	1	2	3	4	5	6	7	8
	RATE	50								(0~99)
	LEVEL	00								(0~99)
	SUS/END	END								

DCW 2

KEY FOLLOW
0

(0~9)

E N V (WAVE)	STEP	1	2	3	4	5	6	7	8
	RATE	76	92	50						(0~99)
	LEVEL	93	30	00						(0~99)
	SUS/END		SUS	END						

DCA 2

KEY FOLLOW
2

(0~9)

E N V (AMP)	STEP	1	2	3	4	5	6	7	8
	RATE	75	47	35						(0~99)
	LEVEL	82	65	00						(0~99)
	SUS/END			END						

● OBERHEIM MATRIX 6
RDONTMRE
Bill Arden, Minnetonka, MN

This one's a bit tough to describe . . . but perhaps the title (an abbreviation for "RadioNightmare") says it best. Bill says that to use it, start with Lever 2 all the way up. Hold a note somewhere in the octave below middle C, move the lever to the bottom of its range, then back up to about the center. Then release the key. Great for sci-fi sound effects. ∎

Matrix-6R Patch ___RDONTMRE___

	0	1	2	3	4	5	6	7	8	9
00 DCO1	Freq 0	Fr/Lf1 31	Sync 1	Pw 0	PW/Lf2 0	Wave 63	Wsel Wave	Levers Both	Keybd Keybd	Click Off
10 DCO2	Freq 0	Fr/Lf1 0	Detune 9	Pw 24	PW/Lf2 0	Wave 31	Wsel Pulse	Levers Both	Keybd Keybd	Click off
20 VCF/VCA	Mix 31	Freq 127	Fr/E1 63	Fr/Prs 0	Res 50	Levers Both	Keybd Keybd	VA1 40	VA/VI 63	VA/E2 63
30 FM/TRCK	FM 20	FM/E3 0	FM/Prs 0	TrckIn Keybd	Track1 0	Track2 15	Track3 31	Track4 47	Track5 63	
40 RMP/PRT	R1Spd 0	Trig Strig	R2 Spd 0	Trig Strig	Port 0	Spd/VI 0	Mode Const	Legato Off	Keymd Reasgn	
50 ENV1	Delay 63	Attack 63	Decay 40	Sustn 30	Rel 0	Amp 40	Amp/VI 63	Trig Strig	Mode Norm	Lf1Trig Norm
60 ENV 2	Delay 0	Attack 0	Decay 0	Sustn 63	Rel 60	Amp 40	Amp/VI 63	Trig Strig	Mode Norm	Lf1Trig Norm
70 ENV 3	Delay 0	Attack 0	Decay 0	Sustn 63	Rel 0	Amp 40	Amp/VI 63	Trig Strig	Mode Norm	Lf1Trig Norm
80 LFO 1	Speed 63	Sp/Prs 0	Wave Noise	Retrig 20	Amp 63	Ap/R1 0	Trig Strig	Lag Off	Smpl Keybd	
90 LFO 2	Speed 0	Sp/Kbd 0	Wave Tri	Retrig 0	Amp 0	Ap/R2 0	Trig Off	Lag Off	Smpl Keybd	

Matrix Modulation

	Source	Amount	Destination
0	Lev 2	55	VCF-Res
1	Lev 2	−63	VCF-Freq
2	Lev 2	−63	DCO2-PW
3	Lev 2	−63	DCO1-WS
4	Lev 2	30	DCO1-FM
5			
6			
7			
8			
9			

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Simmons
Silicon
Mallet

Vibes sounds seem to have become an essential part of any programmer's sound library since FM synthesis arrived on the scene, but very few people have learned how to play them convincingly – now the Mallet strikes back. *Review by Tim Ponting.*

UNTIL RECENTLY, THE sophisticated vibes player has kept a lonely vigil outside his local music store. Every month, some new gadget appears in the window. It might be a percussion controller that looks like a guitar, a keyboard with built-in drum voices or a mysterious box labeled "Trigger-to-MIDI Interface." Nothing that stirs the heart.

But lo, to accompany the new generation of wind controllers, a generation of MIDI controllers designed with the mallet player in mind has arrived and threatens to change the face of tuned percussion playing. Welcome to the land of MIDI, ye xylophonists. Gird up thy loins and edit thy DX7 parameters, ye olde vibe players. From the dreamy town of St Albans in the United Kingdom comes the Silicon Mallet. No, it's not a device for helping your little sister to get to sleep, but

a MIDI tuned percussion instrument. Connect it to an amplifier, and you have up to 99 different sounds at your fingertips (or mallet heads, as the case may be).

The Silicon Mallet consists of a three-octave set of velocity-sensitive pads laid out in the same way as bars on a vibraphone, and is designed to be used with either sticks or mallets. This may be expanded to four or five octaves with the two available expander units. Attached to the pad surface is the Voice Module (brain), containing the sound generation system and the trigger-to-MIDI converter. More of this later . . .

Getting Physical

THE MALLET COMES in three sections. The brain is slotted onto the playing surface, which in turn is screwed onto the frame. The angle at which the playing surface sits is adjustable, as is its height.

The brain is also mounted at an angle, allowing easy viewing of the patch editing controls. Once assembled, the Silicon Mallet is a somewhat unwieldy creature, particularly since it weighs in at a hefty 80 pounds. It makes me glad I'm not a roadie.

The playing surface itself consists of 36 bars – the "black notes" being slightly raised, with a red rim around the edge. (Earlier models, it seems, had a blue rim: aesthetically pleasing, but not terribly practical on a dark stage.) The pads themselves are christened "FS bars" – a new system based on the Force-Sensing Resistor. The basic principle is simple. Beneath the pad surface is a resistive film. The harder the stroke, the greater the change in electrical resistance – a change that can be converted to provide a trigger signal. Neat, huh?

This new setup has a number of advantages. The bars are extremely

► sensitive, and equally so over the whole of their surface. Thus, there are no nodal points (live spots or dead spots), and accuracy of stroke is not as important as on a real vibe or marimba bar. Crosstalk between bars is also minimized (for minimized, read "zero"). The raised rims allow a realistic response from glissando strokes: as on a vibraphone, the mallet "rattles" along the playing surface triggering each pad as it passes.

The actual texture of the bars is hard to describe. Imagine a thin layer of dense rubber over an unyielding surface . . . (fetishists should turn to our special

Expression *"In skilled hands, the Silicon Mallet is as sophisticated and as subtle an instrument as a velocity-sensitive keyboard with aftertouch."*

rubberwear section). Anyway, they have an excellent response, whether you're using soft beaters, ordinary mallets or even drumsticks. One feature of the bars that doesn't seem to be mentioned in the manual is their damping effect. If you're playing a voice with plenty of bite and attack, gently pressing the pad with one of the beaters (or a finger) dampens the sound. How this works, I don't know, but it's certainly an excellent feature and one which is bound to endear the instrument to professional players.

Supplied with the Mallet are four foot controls: switches for stepping through Program Changes and Sustain, and pedals for Modulation and Pitch-Bend (are there any four-legged vibes players out there?). Incidentally, the cords supplied to link pedals to brain are long enough to go twice round the room and back again. Nothing if not generous these Simmons people.

Now to the connections. The playing surface has four sockets. There are two 5-pin DIN sockets to which octave Expanders may be linked, a nine-pin System Interconnect socket which links the brain to the playing surface – and the good ol' AC socket.

The rear panel of the brain is a little bit more busy. In the way of audio outputs you find a pair of quarter-inch jack sockets labeled Mono/Voice 1-3 and Voice 4-6 (more of which later) and a headphone socket. Corresponding to the socket on the playing surface is another nine-pin multi-connector. These have locking screws to ensure that your wayward guitarist doesn't accidentally wrench out the most vital cord of 'em all. Four sockets are available for the footswitches, a cassette interface socket for dumping and loading user-defined patches, and MIDI In, Out and Thru sockets.

Well, that's the hardware out of the way – now for the brain.

Getting Technical

THE HEADPHONE VOLUME control is straightforward enough, but all those

flashing LEDs? As soon as you start playing with the Mallet, though, you realize how logical it all is. And then the awesome potential the machine has as a musical tool begins to sink in and you have to think it all out again . . .

First things first. The Mallet has 99 patches. In this case, a Patch consists of a sound and the information that goes with it: the System, Performance and Voice data, and information relating to what you might call keyboard split. Patches 1-19 are factory presets: the basic voicings on which all others are based. These cannot themselves be altered, although they may be modified and stored elsewhere. The other 80 patches (20-99) are user-programmable. However, when the Mallet comes out of its packaging these locations aren't empty: the folks at Simmons have kindly set up 80 patches, of varying quality from excellent through to . . . er, not quite so excellent. Although you can construct all these patches yourself by editing the factory presets, it is worth saving them to tape immediately. Then you're free to frolic at will (provided it's legal in the state you live in) and create your own patches without losing Simmons' efforts.

Once you've filled the available memory space with sounds, the odds are they're in the wrong order for your gig – if not the first, then it's bound to be the case sooner or later. But those nice boys at Simmons have come to the rescue again with a Sequence facility, allowing you to define a patch order for the relevant footswitch. Pressing the footswitch allows you to step through the programs patch by patch,

repeating them as necessary. Clearly, using the Mallet live is going to be no problem.

Switch on and the machine gives itself a quick electronic rub down, telling you when it's ready for action. Stepping through the patches is easy: using either the footswitch or up/down adjust control on the panel. As you play, six red LEDs flicker on and off in sequence. It's confusing at first, because they have nothing to do with the editing controls.

The Mallet is six-note polyphonic, and there is one LED for each note channel. Three channels are assigned to each of the two outputs; hence those labels Mono/Voice 1-3 and Voice 4-6 – let's just call

them left and right respectively. If you only have the left output connected (Mono) then all notes will be sent over this, reasonably enough. But if you have both outputs in use, and you play, say, a run of eight notes, the first three will be assigned to the left output, the second three to the right, and the last two to the left. And so on *ad infinitum*. This gives a stereo effect, which is far more subtle than it sounds. It gives rise to a characteristic "swirling" which is one of the Mallet's most attractive features.

I Hear Voices

THE VOICES ARE all FM-generated and, on the whole, they're excellent – both the factory presets and the 80 factory-initialized patches. There's absolutely no point in going through them one by one. Suffice it to say that there are a wide variety of straight tuned percussion voices (various vibraphones, marimbas, xylophones, glockenspiels, bells, woodblocks) and some rather wackier ones (such as 'Random Harmonic Bell' and 'Ringing Boo Bam'). Most are usable and if you don't like them, you can always start editing . . .

Now to the programming side of things. Broadly speaking, the Mallet has two modes: Program and Play. An LED indicates which is currently selected. In Play mode, the Mallet may be tuned in steps of hundredths of a semitone. It's also the mode in which MIDI information is set up.

In Program mode, patch data may simply be edited. If you haven't gone off for a beer or dozed off by now, you may have been wondering what Simmons means by System, Performance and Voice data. Read on . . .

In Program mode, the Select button – wait for it – selects either System, Performance or Voice programming modes. Choose the parameters you want to edit, select one of these three modes, and twist the appropriate knob at the bottom. The function of these controls is explained briefly below.

– System Controls: Patch Volume allows you to preset the relative volumes of each patch. In Play, this also acts as a Master Volume.

Dynamic Sensitivity is probably the most innovative and impressive feature of the Mallet. Set it to zero and the harmonics of the sound stay the same no matter how heavy the stroke. Set it to maximum and the harmonics generated will alter dramatically depending on the heaviness of the stroke. The result? The Silicon Mallet comes close to capturing the characteristic harmonic and dynamic properties of real mallet instruments.

The Transpose controls do just that, one by plus or minus three octaves, the other by semitones. If there is no transposition or only of whole octaves, then an LED

indicates Normal.

Select Scale allows you to switch between semitone and quarter-tone scales. The latter is useful for non-chromatically tuned percussion sounds. LEDs indicate which is selected.

- Performance Controls: Tremolo allows you to alter the tremolo (amplitude modulation) of the sound. The first sets the depth: zero in the center position, increasing as the knob is turned to the left. When turned to the right the tremolo increases in stereo; that is, the effect alternates between the two outputs. This enhances the swirling effect I mentioned above. The second tremolo control simply sets the speed of the modulation.

The Vibrato controls, as you'd expect, set the frequency modulation characteristics of the voice. The Depth control has a similar purpose to its tremolo counterpart: zero vibrato in the center position, increasing as the knob is turned to the left. If the knob is turned clockwise, however, the vibrato is increased out of phase: that is, vibrato is applied to each note at a slightly different phase. This is used to provide an unusual chorus effect – another innovative feature of the Mallet.

The Modulation Pedal Route control at first seems frighteningly confusing but in fact is quite simple. Using this feature, the pedal controlling the overall modulation (both vibrato and tremolo) may be set up in different ways. There is a choice of four programs: off; tremolo on & vibrato off; vibrato on & tremolo off; and both tremolo & vibrato on. The word "off" in this sense means that the effect will appear as programmed, that is, the pedal won't alter its characteristics. So if you use the Off program, there may be both tremolo and vibrato present but pressing the pedal won't change them. Reading back, that seems about as clear as mud. All I can say is when the pedal's under your foot, it's dead obvious.

The Pitch-Bend control sets the amount of pitch-bend that the footpedal can apply, from minus to plus a fifth.

- Voice Controls: these are all fairly self-explanatory: Brightness, Harmonic Content, Bite, Attack Time, Decay Time and Gate Time. These parameters were chosen by Simmons as being the most relevant aspects of tuned percussive sound that could be varied. In practice it works well, and after fiddling with a single data slider and hundreds of parameters on other sound generators, it is a relief to use good old analog-style knobs.

Splitting Up

DOING THE SPLITS can be a painful experience, and trying to explain the playing surface split facility on the Mallet is no exception. If you can bear with me for just a little longer . . . Each bar on the playing surface has a MIDI note number. In

Program mode, you set up the Split Position (by defining the MIDI note number at which the split starts), the Split Patch (the patch number of the voice you want to have to the right of the split) and Split Voice Assign.

This allows you to set the number of notes you assign to the left and right side of the split. So, for example, Split Voice Assign set to "4:2" will allow four-note polyphony on the left-hand side of the split, two-note on the right. In this case, three of the four notes from the left side will be sent to the left output, the fourth note and the two notes from the right split will be

sent to the right output. It sounds hideously complicated but it's not, and once again an unusual stereo effect is obtained.

The options available are 6:0 and 0:6 (for MIDI applications), 5:1 and 1:5, 4:2 and 2:4, 3:3, and 1:3. When using either of the last two options, Performance Data from both patches involved is used. Thus, the modulation pedal and pitch-bend pedals have a different effect on notes coming from each side of the split. In other cases, Performance Data is drawn only from the patch in which the split has been set up.

MIDI Mallet

WELL, THAT LEAVES us with one last feature of the machine to examine: its MIDI implementation.

This is the point at which you begin to realize the full potential of the Mallet. First off, it can be used to control external MIDI equipment in the usual way. For example, you can play your SDS9 drum brain from the Mallet playing surface, or the Roland TR727, or a DX7, or indeed any MIDI sound generator. The fun really starts when you start setting up playing surface splits with MIDI slaved instruments.

The MIDI parameters under your control are: MIDI channel (Omni/1-16), split MIDI channel (Omni/1-16), and MIDI transmit and receive modes. Unfortunately, MIDI information is set for the machine as a whole, so if you want a variety of MIDI channels or modes in different patches, you have to change them manually each time. This is fairly easy because the MIDI functions are accessed in the Play mode, but it soon becomes rather irritating.

The eight MIDI transmit and receive modes offer you a variety of setups. On the transmit side, you can send or withhold pitch change and control change data (from the pitch and modulation pedals respectively) and program change data. This allows you to control most aspects of

the slaved MIDI instrument from the Mallet. On the receive side, you can set the Mallet to ignore MIDI channel information, and respond to program change and performance data.

Split MIDI Channel allows you to set a different MIDI channel for the bars to the right of a split. Thus, you can play one MIDI instrument from the left side of the playing surface and another from the right. These MIDI splits may be combined with onboard splits so up to four voices may be controlled and mixed: one internal and one external on each side of the split. This is where Split Voice Assign comes into its own. For example, using a 6:0 split you could play an internal bass sound on the left with six-note polyphony and a DX bell sound on the right. Similarly, with a 3:3 split and appropriate MIDI information, you could play an internal vibes sound mixed with an external celeste sound on the left, and an internal marimba sound mixed with an external orchestral stab on the right. Once you've grasped the basics, the creative potential of the Mallet is limited only by your imagination – corny but true.

So there you are: a guided tour of the Silicon Mallet. Incidentally, the manual is well-written enough to make learning a pleasure. And did you know that "zusammenbau von Staender und Spielflaeche" is German for "mating playing surface to stand"?

Conclusions

AT AROUND $4000, the Silicon Mallet is going to be in a league beyond the reach of most of the noses flattened against the music store window. But with the Mallet you have a whole armory of percussion sounds at your fingertips – and access to even more over MIDI. Besides this, the Mallet could be considered a multi-purpose instrument. MIDI'd to a TR727 or similar, it can be played like a Roland Octapad. In skilled hands, it is as sophisticated and as subtle an instrument as a velocity-sensitive keyboard with aftertouch. The Mallet is a sound generator, a trigger-to-MIDI interface, and a set of highly sensitive pads. I can't afford it, but I have no hesitation about recommending it to anyone who can.

The mallet player has had to wait a long time to enter the world of FM synthesis and MIDI control but the Silicon Mallet has definitely been worth the wait. ∎

READERS' TAPES

Reviewed by Yung Dragen.

REVIEW

IN RESPONSE TO the amazing number of Readers' Tapes being submitted these days, my omniscient and omnipotent editors have decided to expand this column. In turn, I'll be regularly reviewing four to five tapes per month instead of three. Considering the impressive quality of the tapes I've been receiving, it's a challenge I'm eager to take on.

First up this month is the facetiously titled demotape *This Is The Worst Thing I'll Ever Do* by **Bob Gatzen**. Bob uses a raft of equipment, including a Dynacord ADD-one, Simmons SDS7, SDS5, MTM, and SDE, a DX100, Prophet VS, 2000 and 2002, a JP6, both an SP12 and a LinnDrum, and – yes – a Casio CZ101. It was recorded on an Otari eight-track with a good selection of effects and a Macintosh with a similar raft of sequencers and sound editors. Bob goes out of his way to mention that "most everything has been played live into the computer or live to tape in the case of the drums." Which makes me begin to ponder – are *sequencer* users about to start a form of snobbery among themselves? "No step time used." "Autocorrect left off throughout."

But on to the music. Someone should go grant Dave Simmons eternal bliss for opening up a whole new vocabulary to drummers. A very distinct Simmons MTM style of music has developed that I have heard from Bruford, a couple of readers' tapes, and jams I've been involved in myself (even though the drummer used a Yamaha PMC1). For the uninitiated, the MTM converts drum hits into MIDI notes, with all sorts of extra features like chords, sequencing and repeats. This gives drummers the key to playing melodies, too. And the melodies first emerging are very similar in texture – same key, *much* dynamic articulation per note, bursts of sixteenth- or eighth-note phrases with sudden breaks or note changes, and almost universally bell or electric piano type tones. Bob augments this with "real" drumming (it's all sampled percussion) and basslines or drones thrown in on the VS.

The result in this hour-long instrumental demo can be summed up (and we all like things neatly packaged – even our philosophy, don't we?) as an intelligent cross between jazz and easy listening

composed and played by a drummer. And a drummer's sense is indeed different than that of a keyboardist or guitarist – more staccato, with more emphasis on phrasing and dynamics than harmony and melody. The only weaknesses I find are when Bob actually goes for the keyboard himself (his phrasing is sluggish and lines overly dramatic), and too little tonal variation (indeed, I recognized the 2000's ROM sounds, its power-up defaults, being used throughout). Which paints me into another corner: is it a proper criticism to say that just because a musician has a wider tonal palette in front of him or her that they are obligated to use it? We don't impose this upon a trumpeter or a classical guitarist. *Real* bottom line – I *do* enjoy this tape a lot, and I'm excited by the new musical vocabulary I'm hearing.

Next up is *Subterranean Homesick Beanbags* by **The Dan Schaaf Ensemble**. Blues with a sense of humor? Bluegrass? No, try academia. Dan's ensemble produces music that sounds like a cross between romantic classical music and early Morton Subotnick. There are a lot of hurried segments of strict-dynamics eighth notes that sound like they were produced by either several berserk arpeggiators or

one of those new autocomposition programs (either the same note or almost random, with an avant-garde sense of harmony and melody). These are laced together with sustained notes and more flowing, classically melodic passages. Instrumentation is a cross between mostly rounded, bland synths (both analog and four-operator FM) and sampled acoustic instruments (lots of solo horns). Production is also simple and thankfully without gimmicks, which is the way it seems serious music should be – a bigger snare drum isn't going to save or kill the tune in this style of music (although that

white noise splash needs to be mixed way down).

Dan claims to have been writing and producing music for theater productions and documentary films for several years. Obviously, this is his other spiritual life. As with early Subotnick (closest comparison, 'Until Spring'), I found occasional runs or clusters of notes that made me stop what I was doing and cock my head in appreciation, but overall it didn't move me – more brain than heart involved (and occasionally the random note generator seemed to replace the brain). I feel about this tape the same way I felt about Subotnick (whose style of music emerged from the instruments of Don Buchla in much the same way as the Simmons MTM style of music is developing now). It's not my cup of green tea, but I still find myself pulling it out every now and then.

Which brings me to the hugely enjoyable tape *Plasma Bat* by **Hugh T. Caley**. Hugh is "a 26-year-old musician born and raised in the semi-mythical town of Metamora, Michigan. (He) moved to Wayne about a year and a half ago to be closer to the Detroit music scene, and because it was cheap." Imagine crossing the monster beat of someone like the

Beastie Boys (big snare and bigger kick, trash can cymbals, heavy guitar, orch hits, good TR808 work, and sudden breaks on tunes like 'Mega-Wiggle' and 'Dance Fiendish'), the humor and some of the instrumentation of Oingo Boingo (creepshow strings, plus the lyrical guitar and bass on tunes like 'Having Fun with Pointed Things'), and the delivery and humor of Frank Zappa! Modern pop with rough-boy instrumentation and a demented mind. There are a couple of slow-tempo vocal numbers ('Melancholera' and 'You Can't Tell Me') that switch the instrumentation to acoustic drums, fretless

bass, piano, and a more wistful voice, and a few almost pretty electronic instrumentals ('Kent Street Bells,' 'Lullabye,' 'Have You Driven a Fjiord Lately?') amidst the hilarious weirdness. Except for one tune done at a 24-track studio (cleaner production, but less edge), he did it all at home on a Fostex eight-track and mastered to PCM, and it sounds pretty damn good.

My only production complaints come with the slowed-down numbers – they don't have the same in-your-face production as the megabeat tunes, and need to be softened a bit (a full volume raw analog synth sweep is *not* new age material). Despite this, it's one of my all-time favorite independent tapes. Hugh also seems to have achieved a good deal of inner peace with himself over this offering – "I'm kind of proud of this little sucker. Whatever its flaws it's what I wanted, with a wide variety of sounds and feelings. It is not compromised."

We end this month with *Ancestral Voice* by **David Gilden**. David has command over (as in uses *and* programs) an insane amount of equipment, including a Memorymoog, TX816, Xpander, OB8, Emulator II, PPG Wave, Fairlight CMI, Steinway Grand Piano, and, yes – a Casio CZ101. He is also a rather accomplished player of the Kora (a West African harp) and Mbira Dza Vadizmu (a Zimbabwean thumb piano), with the help of Jimmy Earl on fretless bass. The music is very much new age, sounding like a cross between Ron Dexter's seminal *Golden Voyage* (David uses environmental sounds and wind chimes woven into his tapestry) and Andreas Vollenweider with the saccharine removed (his Kora often sounds like an acoustic guitar). Occasionally the synth patches need to be softened up (same fault as Hugh Caley, but to a lesser degree – filter swept sawtooths with no phase shifting or the like), but overall it is yet another splendid piece of well-composed (ie. non-repetitive) new age. David deserves to be on Fortuna or Windham Hill – this is gorgeous stuff. It doesn't pull any surprises (as Steven Van Handel or Jaxon Crow might), but it is better than at least 90% of the music professionally passing itself off as new age out there. ∎

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Bob Gatzen c/o Creative Music Inc., 506 Silas Deane Highway, Wethersfield, Conn 06109
The Dan Schaaf Ensemble c/o Crichet Forum Recordings, 319 Derby, Michigan City, IN 46360
Hugh T Caley 33535 Gertrude, Wayne, MI 48184. (Tape costs $7.)
David Gilden c/o Kora Productions, 165 Prospect Park West, Brooklyn, NY 11215
Send your demotape, along with some biography/equipment details and a recent photo if you have one, to: Reader's Tapes, Music Technology, 7361 Topanga Canyon Blvd., Canoga Park, CA 91303.
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