MUSIC TECHNOLOGY

SEPTEMBER 1988
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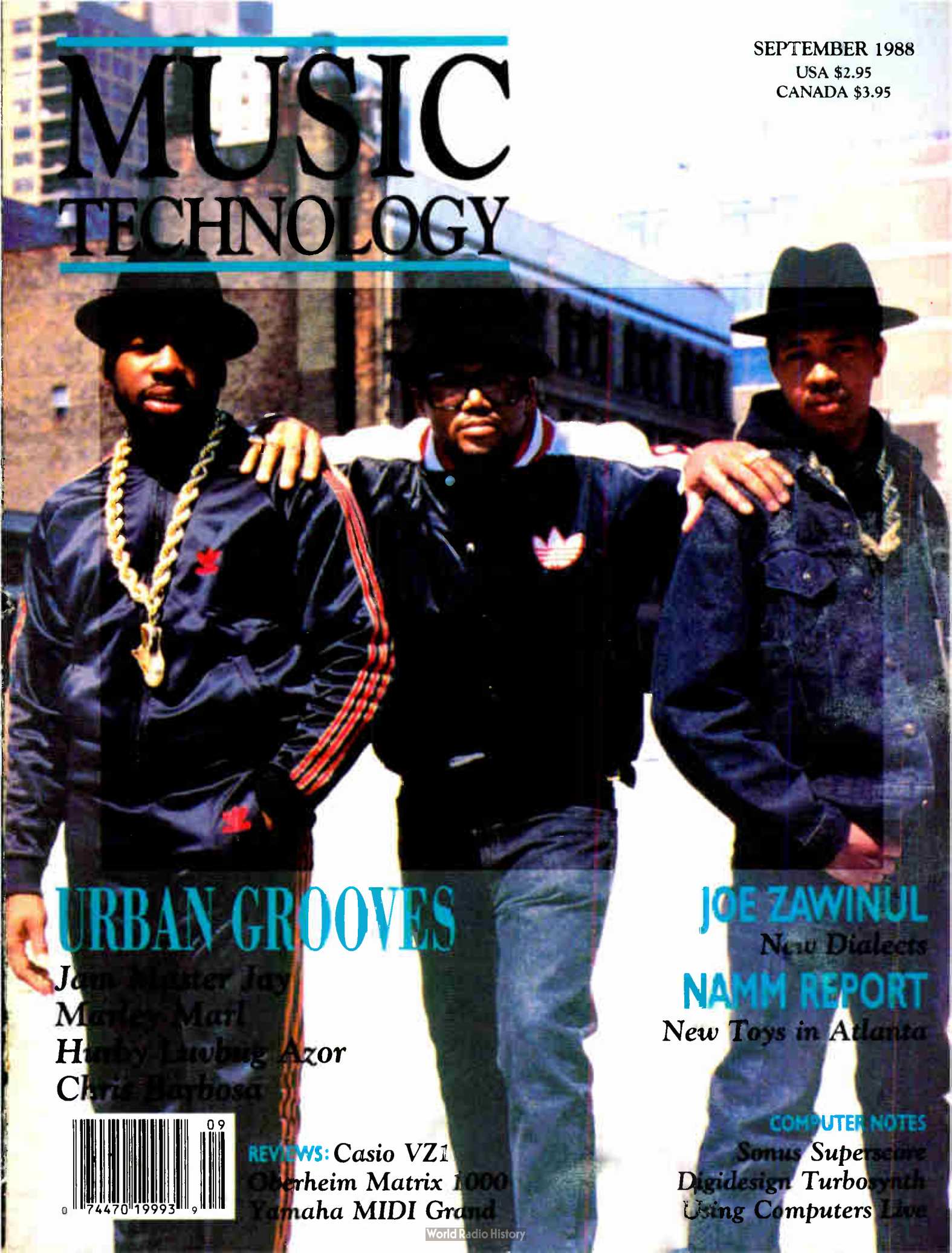

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NAMM REPORT

New Toys in Atlanta

REVIEWS: Casio VZ1
Oberheim Matrix 1000
Yamaha MIDI Grand

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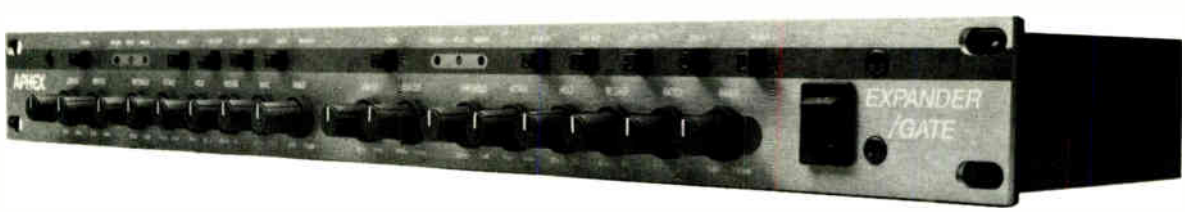

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VOLUME 3, NUMBER 2
SEPTEMBER 1988

EDITORIAL

Standardization

WHETHER I LIKE it or not, it seems standards rule my life, on both a technical and personal level. Now that may sound a bit harsh at first, but I think that governing standards can make life (and work) much easier to deal with and much more comprehensible, despite their possible restrictions.

Technically, of course, standards are extremely important and extremely necessary. The more that can be done to encourage the standardization of various types of connections and means for communication, the easier it is and will be to work with several different tools, be they hardware or software. Unfortunately, manufacturers don't always see things this way – as the incompatibilities between various types of computers and other equipment will readily attest to. Of course, there's legitimate reasons for not wanting to standardize; the fear of "sameness," fundamental differences of opinion on how things should work, and the lack of foresight in many existing and proposed standards being the primary ones. I can't deny the need for variety, nor can I blame anyone for fear that a standard will restrict them from achieving whatever task or goal it is they want to achieve. The trick to get around these objections, however, is to develop a worthwhile standard that does allow for growth. Plenty of examples which have an important influence on musicians do exist.

First and foremost, there's the MIDI standard. Almost single-handedly the Musical Instrument Digital Interface has created a revolution in the composing, recording and playing of music. Now, there's a new related development – the proposed standard MIDI file format was finally agreed upon by several software manufacturers at the recent Atlanta NAMM show. (Because it's a disk standard and doesn't actually have anything to do with how information is sent over MIDI cables, it's not part of the official MIDI specification.) MIDI Files, as they are called, are sequencer files which can be imported from one sequencer into any other (as long as the sequencer in question supports the standard) – regardless of the computer, or software program they were created with, and without the need for special conversion programs. (For those of you who work with word processing programs, they are the musical equivalent of ASCII text files.)

The real beauty of this development is that it allows you to quickly and easily work with several types of compositional, notational and sequencing programs with different types of editing features, for example, so that you can have the best of all worlds.

Another important standard which should be of growing interest to musicians is the AES/EBU standard for the transmission of digital audio. First developed to send stereo digital signals between various types of digital two-track recorders back in 1983, it too is being added to (a multi-channel format is being developed) and will soon have an impact on how we connect together signal processors, mixers, recording devices and even instruments. (For more information on this standard, see the *Digital Audio 102* article in this issue.)

But standards like this can also be applied to how you work and your creative output. The difference is that you don't have a panel of experts who can create the standard for you – you have to create your own. You need to somehow define what your capabilities are, determine the pre-existing standards in your area of interest (which admittedly isn't easy), and then try to set a reasonable goal you think you can attain. This last point is particularly important because any unreasonably high standards will end up causing more frustrations than results, though low to non-existent comparative standards will promote an atmophere that condones the production of worthless garbage.

Any standard really bases itself on shared expectations; machines need to know what type of data to expect and how to process the appropriate information, and similarly, you need to know a quality level that your music and programming can be expected to approach. Taken in this light, and remembering to give yourself room to grow, standards can actually be quite liberating. ■

Bob O'Donnell

In case you're wondering, no, we didn't forget about the results of our Readers' Survey and the prize winners – we were just overwhelmed by the response. Both the results and the winners will be announced in next month's issue.

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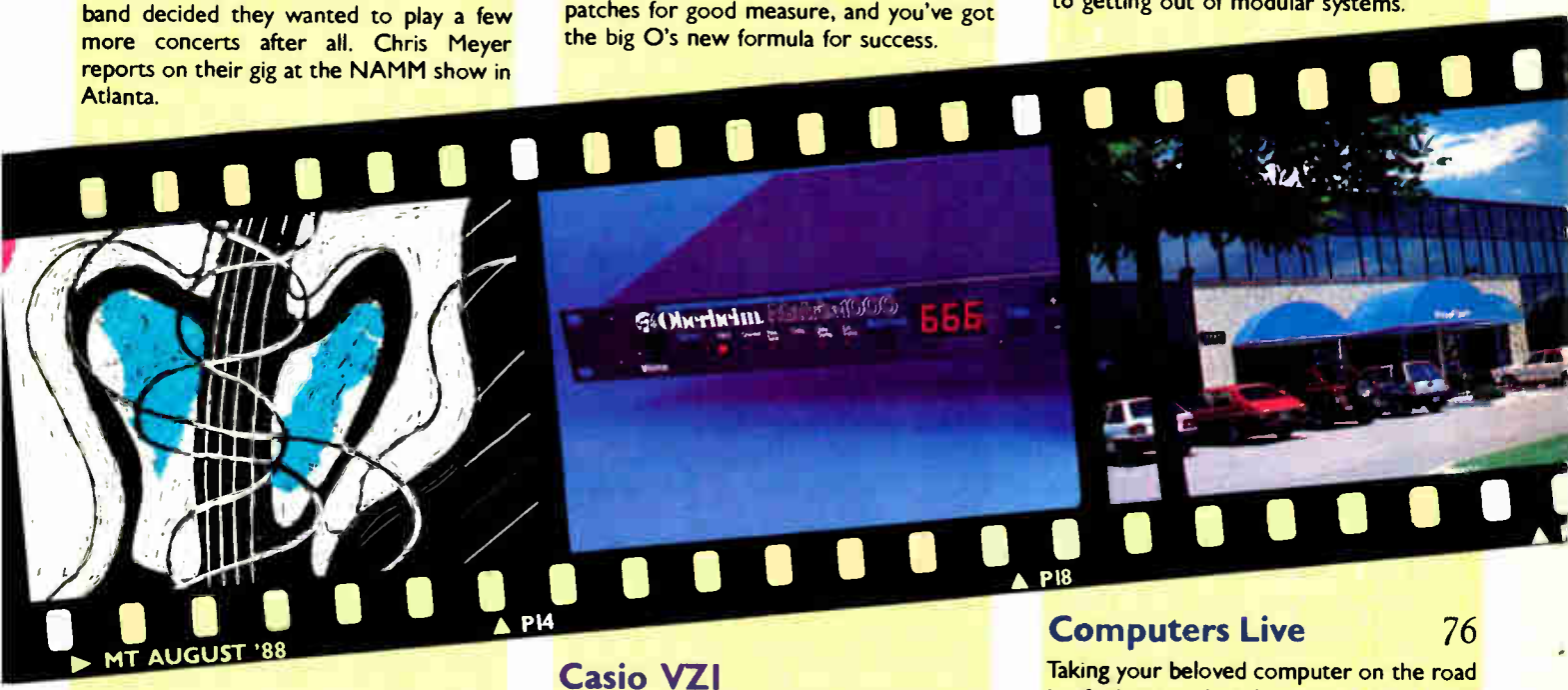

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READERS' LETTERS

Send any questions or comments that you may have to: Readers' Letters, Music Technology, 22024 Lassen Street, Suite 118, Chatsworth, CA 91311.

LETTERS

Dear Music Technology,

With regard to readers wishes from the manufacturers, I have been waiting for years for a manufacturer to come out with a true all-in-one music workstation. Several manufacturers (Korg, Roland, Ensoniq) currently claim to have done so but they always leave out something vital.

The main omission that prevents me from buying any of these "workstations" is a very simple one to include but no-one seems to have done it yet. When mixing different sounds together in a studio the thing that makes the biggest difference to the depth perception and space is the different amount of reverb, delay, etc, on each instrument. You don't want the same amount of reverb on the snare as on the kick, the strings, etc. When you have only two or four outputs on a keyboard you end up having to put the same overall amount of effects on all the sounds – which then limits you to demo quality. You could have master quality, however, with the simple expedient of having one extra page in the software that enabled you to control how much volume of each instrument gets sent to the on-board reverb – just like you can now control the volume of each individual instrument in a sequence. The cost of doing this would be minimal.

The next evolutionary step would be to include EQ for each instrument and then you'd truly have an all-in-one workstation that you could do masters on. All you'd need to add would be a tape deck for vocals and guitars, and a sync unit and you'd be in business.

The ultimate complete home studio machine would also include basic sampling capability and digital automation of its onboard mixer. This should be possible for considerably less than $4000. A built-in cassette deck would round it off and you'd be able to just plug in your headphones and make complete tapes all in the one unit. The first manufacturer who releases one of these will clean up.

Dr. MIDI
Hollywood, CA

Dear Music Technology,

Re: Letters, July '88 (which talked about how many new instruments seem specifically intended to produce pop music):

You should have told Charles Williamson to stop making up excuses for any mediocre compositions he produces in his basement. People make music, machines only help (or hinder).

As for microtonality, especially as regards to "interactivity" – it has been available via pitch-bend facilities since the day the voltage-controlled oscillator was invented. One need only develop the discipline of using it, as guitarists and sax players have over the past 30 years. Good music is not the result of "accidents." Give it up, or turn it a-loose.

Ted Greenwald
NY, NY

Dear Music Technology,

Chris Meyer has missed a vitally important issue in his June '88 editorial. Copy protection is merely a symptom of a larger, much more frightening, problem in the computer and music industries. Building protection into synthesizers is not going to solve it. The issue is that of copyrighting computer data; in our case, patch and sample data.

The three major questions are: Should this data be copyrightable? If so, how much data is enough to justify a copyright? And how similar must two pieces of data be before copyright infringement occurs? Let me present some evidence.

A typical synthesizer patch contains less than 400 bytes (characters) of data. Is this enough information to justify a copyright? Too little? How about a 512,000-byte digital sample? If this is enough, how about 511,999 bytes? Where do you draw the line?

There's no reason that data size should be the only deciding factor. Maybe a 50,000-byte sample of a one-of-a-kind musical instrument should be more "copyrightable" than a 100,000-byte sample of a common sneeze. And how about a 20,000-byte computer program? Computer programs are data too . . . ! (A distinction might have to made there, too, with Apple dividing applications into "data" and "resource" forks on the Mac. – Tech Ed.) *Clearly, the issues here are complex.*

Just look at the mess currently in the recording industry. With the invention of the CD, recorded music has also become computer data. Artists sue each other, claiming copyright infringement, because two songs' basslines are nearly identical. How similar do two songs have to be before it is considered copyright infringement? You can't simply count similar notes, because the instrumentation and feel of those notes may be entirely different. You can't just say "after N seconds of similar music, it's

infringement," because a single note can last indefinitely. There is no standard.

To top it off, the theoretical computer science problem of comparing two sets of data to see if they are "nearly" the same is very, very difficult. And it doesn't even take into account artistic matters like "look and feel."

Recently, a group of governmental copyright people approached a well-known computer music facility, looking for information. "People are coming to us, asking if they can copyright their samples. What is a sample?" The lawmakers are almost completely uninformed. And they will be making laws. Soon. Does this bother anybody?

Daniel J. Barrett
Dept. of Computer Science
The John Hopkins University
Baltimore, MD

You bring up several very interesting points Dan, that will, in fact, affect all of us. Does anybody else out there have any ideas?

For the record, here at MT we're concerned with these kind of problems and have started to address them with our occasional series entitled, "Legal Issues and the Hi-Tech Musician." The first article, which was on copy protection, appeared in the Computer Notes section of our August issue. Look for more in the near future. Also, the original brickthrowing hasn't stopped, so stay tuned . . . - BO'D

Dear Music Technology,

I came across your magazine the other day in a bookstore (buying "baby" books for my recent firstborn!), and literally read it from cover to cover. Music Technology, combined with borrowing a friend's Yamaha DX21 a few months ago, has given me the desire to start a small (ie. "inexpensive") home music system.

Here comes the $64,000 question: How do I start? As an experienced computer programmer, some of the computer-related information in Music Technology I can understand, but quite a bit sounds Greek to me. Also, I have absolutely no idea what equipment/software is most popular, best for home use, where I go to buy this stuff, etc, etc. Any suggestions?

Daniel Schoch
Millington, NJ

Yes. First off, there are several excellent books currently available, including Michael Boom's* Music Through MIDI *($19.95, Microsoft Press) which we reviewed in MT November '87, which can explain basic concepts about MIDI and sythesizers, as well as introduce you to some popular products. The problem with any book, of course, is that it can quickly become dated, so if you want to keep up on the latest developments we humbly suggest you continue reading MT. You may want to pick up a copy of our August '88 issue, which has an index of everything we've covered over the last two years - there are a lot of good educational articles in other back issues.

As for where to buy the products, find a good local music store with knowledgeable sales people - they can prove to be of invaluable help when making purchasing decisions. Good luck. - BO'D

Dear Music Technology,

Your June '88 issue "Perspectives" article titled "But Isn't Standard Notation Dead?" really bent me. The bait, which got me to bite initially, was that piano-roll graphing and/or sequencing has virtue over computer music notation (CMN). But the hook I couldn't swallow was that a "fully sequenced modern composition/performance (which) cuts everyone out of the loop except the composer and the audience" has great virtue as well. I can't wait 'til magazines are fully electronic, so the authors of such fodder will speak aloud their words of wisdom, and the true patronizing flavor will leave the reader with the proper distaste.

First off, in my opinion, time/pitch relationships in music, which can be much more complex than any MIDI sequencer can begin to approach, are expressed with much greater eloquence, and if desired, much greater accuracy and/or clarity than piano roll notation.

On the other hand, given the immediate feedback and interactivity of a computer/synth rig, a novice sitting down to **edit** *(not read, not write) a piano roll display or a CMN display of the same musical material should find only the most minute gain from piano roll. About as much effort say, as it takes to read the manual.*

In the next paragraph, the author seems to imply that by using a patch dump or a sequence, a composer can specify details of sound/timbre/performance inflection to his/her satisfaction. And that storing one's compositions in a MIDI-based file is more durable than pencil and paper. I believe that MIDI will be completely obsolescent technically in 10 years or less (but may still linger for marketing reasons). So go ahead, trust your magnum opus to that medium if you like.

But the ultimate point of all this, which the author seems to be making, is that a composer would want to/can conceive of every parameter of his music, describe it to machines and achieve a perfect realization of what's in his/her head. Furthermore, he seems to think that audiences would enjoy coming to a concert to hear it. I think that no audience will be interested in coming to a concert to hear a recording (digital or otherwise) of a new work, when it is possible to simply download the samples to a digital player via a home communications link and listen in your own home. And if the audience response is gone, then this scenario means the composer's gratification comes from sales/distribution of the work.

Without getting into a discourse on why one composes in the first place, let me just say that this level of de-humanization is not what I long for. Music making can be a great sense of joy at human interaction. I understand the frustration a composer can feel when performers fail to interpret his/her music capably. But a composer is after all, as limited as any one else as a performer, and so is equally incapable of describing the desired performance. Only more practice can make a composer a better

performer or communicator. And I dare say that there is no composer alive or dead that hasn't been moved by an interpretation of a work never dreamed possible. It is the supremely staggering ability of each musician to interpret music, and to feel it uniquely, that makes it less desirable to use a mechanistic realization of a work. That is why CMN, with its infinite level of specification, from very loose (not possible with sequencing), to as specific as possible (also not possible with current sequencing) is the notation medium of choice.

I'll also argue that with technological assistance, there is nothing that a machine can play that a performer will not be able to play someday. Real-time playing technique will always be of great value, as long as human expression remains unquantifiable, and people enjoy and marvel at the achievements of others. The article stops short of asserting that the composer himself can be replaced by programs which can generate music that he/she could never achieve. Thank God for small favors.

The opinions expressed in your article portray the composer as a closet creature . . . desiring no outside influence, completely self-absorbed, yet with a mysterious longing for recognition so great that it must be satisfied even after death. Perhaps the author should change his title from "futurist" to "fatalist." Hopefully, this is not what the future holds for composers (or other artists). I don't think it is.

Mike Williams
Simi Valley, CA

Author Travis Charbeneau replies:

I was writing about options, not absolutes. Today we have the option to cut everyone out of the loop but composer and audience, or notate and pass around the charts - or any combination of the above.

But I see you have grievances beyond the notation controversy, regarding live performance, for example. At one of the recent "aid" extravaganzas, Stevie Wonder gave a performance restricted to just him and his computer. A few purists probably freaked (as they did when Bob Dylan first went electric at Newport) but the audience loved it. I detect techno-phobia, Mike.

But, to the crux of your misunderstanding: "there is nothing that a machine can play that a performer will not be able to play someday." That's highly debatable, but wholly beside the point. Machines neither compose nor play. As the automobile is an extension of the legs, and TV an extension of the eyes and ears, MIDI and related gear are merely extensions of the human musical sensibility. For better or worse, my mind does the composing and playing. The gear merely makes noises. A computer and tone module no more play on their own than do a singer's vocal chords. Until you can distinguish between the aesthetic impulse and the ever-evolving physical apparatus which realizes that impulse, you will be caught in a non-existent conundrum.

It helps to remember that when the piano forte was first introduced, it was widely panned as tinny, dissonant and, yes, "mechanical." ■

NEWS DESK

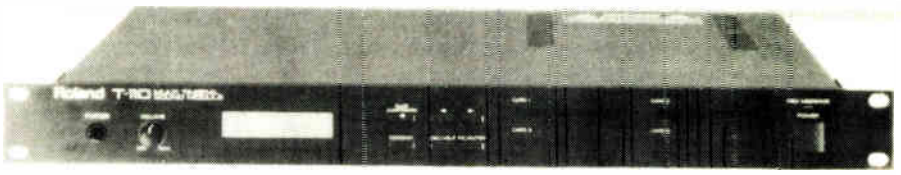
NEWS

Roland Deluge

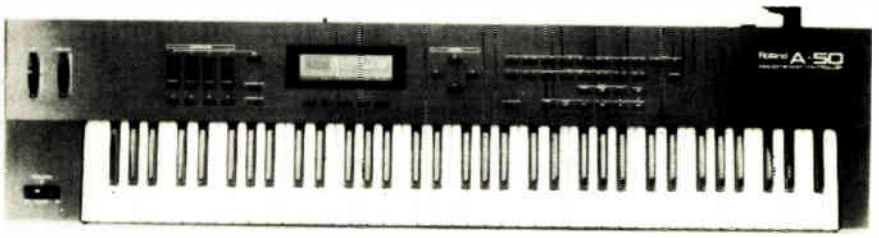
Although they didn't have a booth at the summer NAMM convention, representatives from Roland provided us with lots of press releases on new gear which they plan to introduce over the next few months. The TIIO Multi-Timbral Sound Module ($1095) is a single-space sample playback unit with 2Meg of 12-bit samples in ROM. The unit – which uses technology similar to the S550 – has four ROM card slots on its front panel which expand the online sample capacity to 4Meg, and six individual audio outputs on the back panel (as well as a stereo pair). Most impressive of all is the TIIO's whopping 31-voice polyphony.

On the drum machine front, they let us in on the R8 ($950), their new top-of-the-line model, offering 48 onboard 16-bit sounds, touch-sensitive pads, the ability to alter the pitch and decay of the samples and store those alterations as one of 16 user-programmable sounds, and resolution up to 96 beats per quarter note (1/384 note). Another unique feature intended to improve the feel of the drum patterns is the "Human Feel" function, which provides eight different options for adding slight timing nuances and accents to any pattern.

Also on the percussive side of things is the new Octapad II, the Pad80 ($795), which adds 64 memory locations, an LCD display, Velocity Switching and Mixing, and several other functions to the capabilities of the original.



Holy cow, Batman! 31 sampled voices in the Roland TIIO!



The A50 keyboard controller from Roland combines top-of-the-line controller features with a slick presentation.

The A50 ($2195) is their new mother keyboard. It sports a 76-key JX10-type keyboard with polyphonic aftertouch, the ability to support up to four different zones with independent velocity and aftertouch curves, a 240-character display to see those curves, four independent MIDI outs, two MIDI ins with a built-in merging capability, and memory to store SysEx dumps from connected synths.

The Boss division announced the BX40 ($175), BX60 ($250) and BX80 ($450) portable four-input mono, six-input stereo, and eight-input stereo mixers, as well as the MS30 portable speaker system ($150).

If all you need is a basic MIDI switcher, Roland also covered that base. The A880 ($395) is a programmable 8X8 MIDI patchbay with 64 memory locations and the ability to merge any two MIDI inputs.

Looks like it's gonna be a great year.

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

Making It All Clear

The International MIDI Association is now distributing the new and completely rewritten MIDI 1.0 Detailed Specification, version 4.0, which adds the Sample Dump Standard, MIDI Time Code, Registered and Non-Registered Parameter numbers, the MIDI Inquiry Message and the extensions to the Sample Dump Standard. The latest controller and System Exclusive ID number assignments are also listed.

The new document is available only through the IMA. Price is $35 for non-members and $25 for IMA members. Prepaid and UPS COD orders are accepted.

MORE FROM The International MIDI Association, 5316 W. 57th Street, Los Angeles, CA 90056. Tel: (213) 649-6434.

For Your Reading Pleasure . . .

Wish you had a handy reference for looking up troublesome words? Hal Leonard Publishing has recently released *The Electronic Music Dictionary* by Bo Tomlyn, which contains over 300 definitions covering the basic elements of synthesizers, amplification, MIDI and computers as they relate to making music. The book comes in a tidy 6"X9" size and retails for $5.95.

On the heftier side, A-R Editions Inc. has released the Computer Music and Digital Audio Series texts, a trade and textbook series covering technical and aesthetic issues in digital technology. The five titles available at this time are: *Digital Audio Signal Processing* ($39.95); *Composers and the Computer* ($27.95); *Digital Audio Engineering* ($29.95); *Computer Applications in Music: A Bibliography* ($49.95); and *The Compact Disc Handbook*, written by the highly-respected educator Ken Pohlmann.

MORE FROM Hal Leonard Publishing Corporation, PO Box 13819, Milwaukee, WI 53213. Tel (414) 774-3630.

A-R Editions Inc, 315 W. Gorham Street, Madison, WI 53703. Tel: (608) 251-2114.

Errata

We'd like to apologize to Forte for publishing an incorrect address in the July '88 review of their Music Mentor. The correct address for inquiries is: Forte Music, PO Box 6322, San Jose, CA 95150. Tel: (415) 965-8880. ■

FINALLY, 8 TRACKS IN THE PALM OF YOUR HAND.

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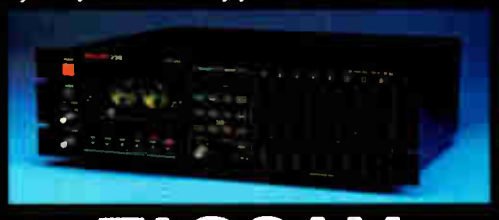
But the 238 doesn't stop there. It's fully synchronizable with MIDI/SMPTE time codes, it's got a serial port for computer interfacing and it's designed with open architecture for software upgrades.

Add to that the advantages of full-function remote control, auto punch in/out, shuttling capability and dbx II noise reduction, and you've got a machine that just about does it all.

If this all seems a little too good to be true, talk to your Tascam dealer. To understand how good the 238 Syncaset really is, you'll just have to try your hand at it.



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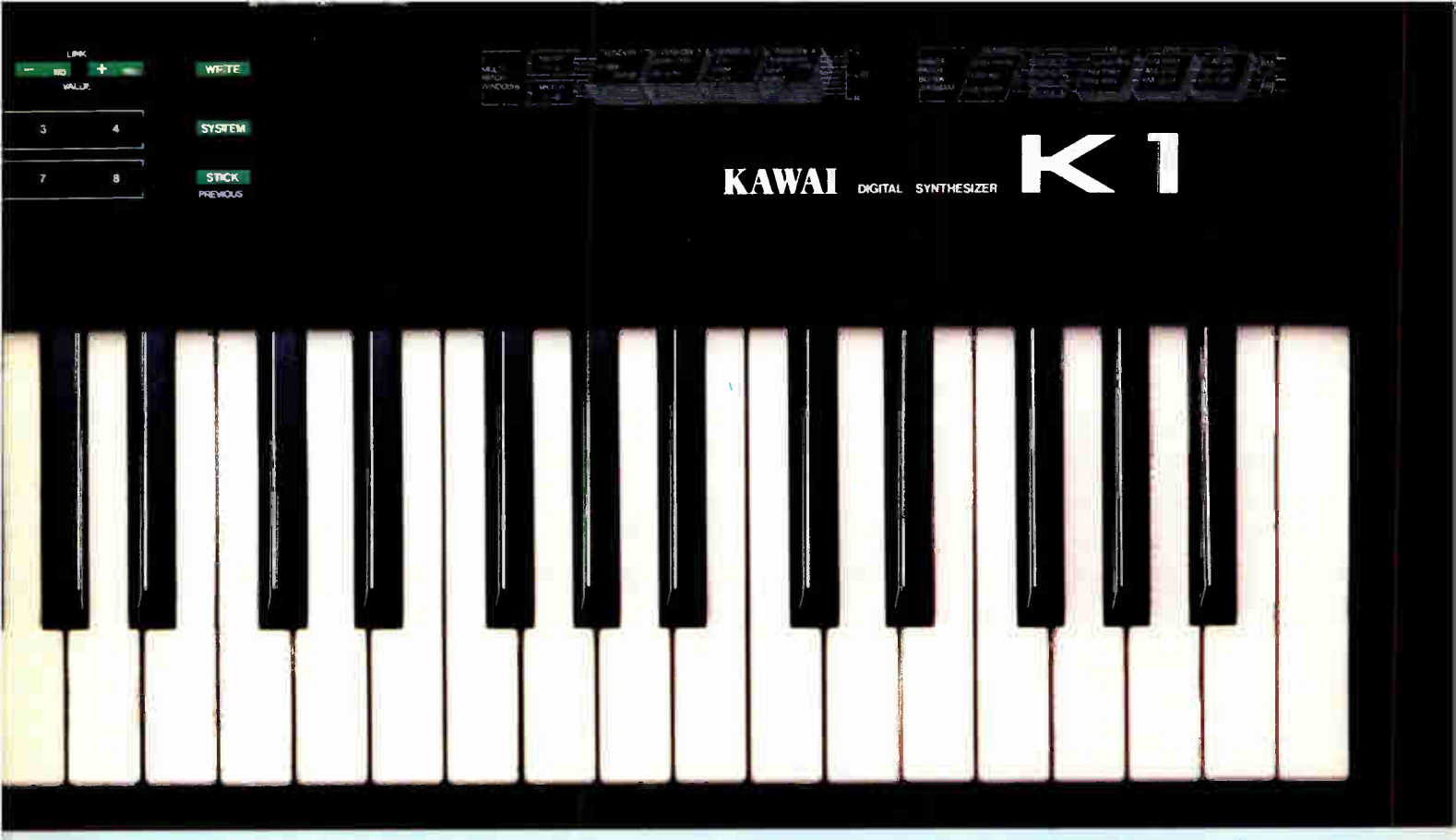

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BASS HOW LOW CAN YOU GO?

Illustration *Sophie Laurence-Jones*

Listening to commercial sample libraries, you'd be forgiven for believing that the electric bass is capable of producing only a handful of different sounds. A session with your sampler could take your music to new depths. *Text by Tom McLaughlin.*

A LITTLE BASS history: the electric bass was designed to replace the plucked upright bass in live performance situations. Using your imagination, its sound almost resembles the real thing – on a bad night. Scope for any other sound was severely limited by simple pickups and passive filters. Even worse is that some basses sported only a single volume control. For all it had going for it, the electric bass might have shared the fate of the do-do bird. Instead it caught on. And in a big way.

Today the electric bass plays a key role in music, not only rhythmically, harmonically and often melodically, but in respect of the energy band it occupies within the audio frequency spectrum. How well the bass "sits" in a commercial track is vital and can make or break a song aimed at the dance floor.

The electric bass relies upon pickups, tone controls and an external amplifier to filter and amplify its sound. It may have a solid or hollow body, single or multiple pickups and may be fretted or fretless. The bass you sample will be determined by what instruments you have available to you and what type of sound you're looking for. If it's an authentic '60s bass sound, hunt down an authentic '60s bass (you can easily find used ones for less than $400 – what you'd spend on an extra reverb – and you can always sell it or trade it in on another later). Pulling the current "bright 'n' snappy" sound from a 15-year-old bass is difficult (they weren't designed for it), but the new wave of basses is capable of mimicking most bass sounds.

Strings & Picks

NEW STRINGS MAKE a difference like night and day, and can give a mediocre instrument a new lease on life. As you might expect, they produce a brighter sound than an old, beat-up set. Strings come in several gauges, metal alloy

combinations and flat or round outer windings. Flatwound strings keep finger squeaks to a minimum but tend to lose their tone in a matter of weeks. Even when new, flatwounds never seem a match for roundwound strings' crispness and clarity. New strings don't come cheap though, they can run from $15-40 a set. Simply cleaning the grime and finger perspiration from a used set of strings with rubbing alcohol or tape-head cleaner, can make a big difference and is the next best thing to buying a new set if your budget doesn't allow it.

How and where you pluck the string has a lot to do with the tone color of the sound. In addition to your fingers, there are all sorts of plectrums (picks) available, made from a wide variety of materials. The nylon variety have the widest range of flexibilities, ranging in hardness from a baby's fingernail to a buffalo tooth. There are also plectrums made from stainless steel, stone and occasionally you'll find them in seashell or hardwood. Try to find a pressed felt zither pick – they're good for a cross between a finger picked and plectrum sound.

It's all a matter of taste, but generally hard picks give a more immediate response, brighter tone, and more pronounced attack than soft ones. Metal plectrums add a "zing" to the front edge of notes with the metal-to-metal contact bringing out upper harmonics "other plectrums fail to reach"; this results in a very aggressive and distinctive bass sound.

The force with which a string is plucked not only gives a brighter sound, it also causes notes to begin sharp. The greater the force used to set a string into motion, the sharper the initial pitch is likely to be, and the longer it will take the string to settle down to its proper pitch.

Where you pluck a string and the effect on its harmonic profile is really a separate issue. Suffice to say here that plucking a string at half its length reduces or cancels every second harmonic. Since only half the harmonics are prominent this is the mellowest and "roundest" plucking position, producing a fruity, square wave-like tone. The further away from its center you pluck a string the brighter the string will sound, mainly, but not solely, because fewer harmonics are cancelled out (and higher numbered ones are directly excited). Finger picking any bass at the 12th fret will make it sound like an old Fender; forceful application of a metal pick near the bridge will make any sound like a Rickenbacker. Starting at its center and working your way to the bridge, listen to the tone color as you repeatedly pluck a string with varying degrees of forcefulness and a selection of pick gauges (not forgetting your fingers) to get an idea of the wide palette of timbres open to you. Don't ignore harmonics, either.

Amping & DI'ing

TO AMPLIFY, OR not to amplify, that is the question. Direct injection of the instrument's output, either into your sampler or after passing through effects and/or a mixer, will give you a crisp sound but may emphasize finger noise and string squeaks. Miking up the sound from an amplifier adds the coloration of the amplifier, tone control circuitry, speaker and microphone, and gives a warmer sound with a natural ambience. Bass amplifiers also have a built in bass boost, which helps round out the tone. Many recording engineers and producers prefer combining both techniques and using the presence of DI'ing and the warmth of amp miking to make a composite bass sound.

Microphones

THE MIC USED to sample an amplified bass plays a large role in the overall sound. Due to its frequency range and percussive quality, a dynamic microphone with a large diaphragm captures the low frequencies of the bass more authentically than your old faithful vocal mic. Most microphones don't have the low frequency response needed. The fundamental of the bass' low E is about 40Hz. Since 95% of sounds you'd want to record have little or no energy present below 80Hz, you'll find many mics go down to only 50 or 60Hz.

Traditional kick drum mics like the AKG12 or Beyer do the job of picking up very low frequencies admirably, but seem a bit lacking in the mid-upper frequency range. (AKG has updated their old "12" with a brighter model to reflect more modern kick drum sounds.) Hefty equalization in the 3k to 8kHz range is needed with these mics for a crisp, modern bass sound. Using a second mic, one with a brighter mid range, is often a more desirable alternative.

If a general-purpose mic is all you have, you'll need to heavily equalize the bottom end to do the bass justice. It takes work but it can be done, often with little more than the bass and treble boost/cut and mid parametric equalization on your standard mixer.

Amplifiers

A HOST OF companies manufacture combo amplifiers designed to flatter the electric bass. The variety of tone control/amplifier circuitry, speaker sizes/configurations and enclosure designs available is a good example of man's ingenuity. Effectively amplifying low frequencies has several physical limitations to hamper a straightforward design approach. In terms of size needed for an acoustic resonator to project the fundamental of a note, low means B-I-G. In the case of very low (40Hz), it means V-E-R-Y big (we could be talking the volume of a small-sized bedroom). The fact that manufacturers ▶

► have managed to bring us bass amplifiers that we can lift, let alone throw in the trunk of a car, is quite an accomplishment.

When miking up amplifiers, give the speaker room to "breathe" by placing the mic 9-12" from the speaker cone. Pointing the mic directly at the center of the speaker will emphasize the mid-upper frequencies but often adds a boomy thud to the attack portion of samples. You may want this. If not, moving the microphone so that it faces the speaker at an angle will prevent its occurrence. The size of the room also makes a difference, because the wavelength of the lower frequencies can be many yards long.

Positioning a single microphone to record a bass amp with multiple speakers is preferable to using several mics. It's true that bass frequencies have less phase cancellation problems than higher frequencies do (you may notice the attack portion going funny), but even low notes may start to disappear. Multiple mics almost always have phase problems. If you do choose to use more than one mic and find that the sound coming through your monitors bears little resemblance to what you hear in the room, reversing the phase of one or more microphones may be necessary. Comprehensive mixing consoles will have a provision for reversing the phase of a channel; it's a simple and inexpensive circuit to install, otherwise you'll have to make up a mic cable cord wired out of phase (swap the wires around at one end) with your other cables, but do remember to mark it.

Equalization

ALWAYS START WITH the instrument's tone controls before resorting to external means. Set any amp, mixer and effects (if you're using them) to flat and fiddle around with the bass' onboard tone controls to see how close you can get to the sound you're after. Take a sample of it and see how it sounds. You may find it needs very little treatment for a straightforward electric bass sound. If not, it'll give you a good idea what needs to be done. If too little top end is coming back from your sampler, try a higher sampling rate before subjecting the top end to equalization.

When using a graphic or parametric equalizer, try cutting frequencies around its middle range a little bit – between 250-500Hz. Cutting in this area helps the bass sound a lot less muddy. Boosting frequencies in the 4.5-7kHz range will improve definition, accentuating the pluck or pop portion of the sound (and string squeak); boosting between 40-200Hz will add "balls" by increasing the fundamental and lower harmonics. Be careful with frequencies under 80Hz – a "sub-basement" bottom end may "boom" when transposed upwards. If need be, a response curve more to your liking can be fashioned easily enough when session time comes, with a graphic or parametric equalizer.

Compression

IF ONE EFFECT or treatment is most responsible for the modern bass guitar sound, it has to be compression. Even though the electric guitar and bass have substantially more sustain than their acoustic counterparts, they're still plucked string instruments and notes are not infinitely sustainable. Compressing a bass won't lengthen its notes (although sample looping will), but what there is of a note will stand more of a chance of being heard. Compression levels out the loudness of input signals, and used to the extreme will eliminate any amplitude variation on the length of a note – the bass will essentially be either on or off. By setting the attack control on your compressor to a few milliseconds, the percussive starts of notes will be kept while the sustained portion comes up solid, giving today's punchy bass sounds.

Effects

FULL REVERB TENDS to confuse the bottom end occupied by the bass (and there are few things worse in life than a confused bottom end). A little added ambience on DI'd bass will help prevent your samples sounding too little; a wee bit will do the trick. In short, go easy. If you can hear it as an effect, it's probably too much.

I'm personally fond of putting a small amount of Yamaha's 'Early Reflections' on DI'd bass and guitar. This is an ambience program on their SPX and REV series of digital reverbs that excels in giving a "live" feel, adding the impression of a sound bouncing off of walls in a room, without the thickness associated with typical room, hall, spring or plate reverbs.

Flanging, phase shifting, chorus, pitch shifting, delays, octave dividing and overdrive all work well with electric bass. Although of limited bandwidth, the richness of analog effects actually seems to complement the range of the bass (chorus in particular). The noise usually inherent in analog effects can easily be filtered out without affecting the bass' tone to any great extent.

Flanging and phase shifting add harmonic movement to sounds by cancelling certain frequency bands and accentuating others. In addition to an internal LFO for modulation, some flangers have an envelope follower circuit that allows the flange to follow the amplitude envelope of the input instrument. This adds an almost vocal quality to the bass. Chorusing adds richness and depth to the bass – anything from the simple chorus with a delay line to the complex chorus algorithms available on digital reverbs and multiple effects units.

Effects are a special case with the bass. To make sure that the fundamental and lower harmonics remain strong, it's wise to carefully balance the treated and untreated bass signals. Some effects units have this provision onboard. If they don't, you'll need to split the bass' output, send one half to the effect and blend them together with a mixer. Fading the effected signal in (with the aid of a noise gate set to a slow attack) ensures a crisp, clean pluck.

Sampling Rates

AS THE FREQUENCY spectrum of the bass is so low, you can often afford to sample it at lower rates than with other instruments (the 32nd harmonic of the bass' high E is only slightly higher than 5kHz). Valuable memory space may be saved when working at lower sampling rates while still capturing the character of the instrument. Use common sense with sampling rates; you should be able to get away with lower sampling rates for a mellow jazz bass than, say, for a crisp, slap sound. Sampling rates giving a playback response of between 4-12kHz are a good starting point to experiment with.

It's important that the trigger at the input of your sampler be set low enough not to lose any attack transients on the pluck portion of bass notes. If you have the choice between automatic or manual triggering, you can hit the manual trigger then quickly play your bass, editing off the blank space before the note at a later time. Leaving a small amount of string squeak or finger noise will help give the illusion of a real bass being played. If sampling a fretless bass, try keeping a little bit of a pitch slide at the start of the note – it's useful to imitate a player changing notes or bending into proper intonation.

Lower sampling rates also mean longer samples, and may enable you to record bass samples so long that a loop is unnecessary; the continual downward harmonic movement throughout the duration of plucked string notes often makes finding smooth long loops a headache. Alternating loops and loop cross-fading sometimes help, but due to the decrease in amplitude and flattening of pitch, there almost always seems to be some tell-tale cyclic movement that gives the game away. Compression helps the amplitude part of the problem. A "short" loop (where the loop is only one, two, four or eight cycles long) may be the best compromise for problem bass samples. If you find the right waveform(s) to loop around midway in the sample, the effect of heavy compression can be achieved.

Multi-Sampling

MAPPING YOUR BASS across the keyboard should be quite straightforward if you sampled a comprehensive set of

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pitches in the first place. If your bass part is not too demanding you'd be surprised at how few bass samples you can actually get away with.

As with any keyboard map, the best place to start is by assigning samples to notes relating to the original sampled pitches, your ear telling you how far from the original pitch a sample can be taken to meet its neighboring sample. For those without velocity switching, soft and loud versions of the same bass sound can be assigned to different keyboard zones.

Some samplers allow you to sample two sounds and switch or cross-fade between them once a pre-chosen MIDI velocity is reached. With velocity cross-fading, the harder you hit a key, the more dominant the "loud" sample will be. This requires two Voltage or Digitally Controlled Amplifiers to carry out. Velocity Switching requires only one DCA, switching to the louder sample once the velocity threshold is reached.

The most obvious application of this facility is to chop between soft and loud bass samples. Loud samples may be slapped or popped versions of a bass note, but don't rule out switching to harmonics or octaves. Loud samples needn't be technically perfect; a small amount of string "rattle" or squeak often adds aggression and realism to a loud bass sample.

Other samplers have a velocity start point feature, that allows soft notes to start playback after the bright, percussive attack (a little velocity to amplifier attack or filter cutoff also helps modulate the bass tone).

Filters

TWO TYPES OF filters are commonly available on samplers; fixed cutoff-frequency low-pass filters, and envelope-controllable low-pass filters. Some samplers have both. Occasionally you'll come across a high-pass or band-pass filter but these are more likely to be found in external software packages than resident in a sampler's internal operating software.

The cutoff of fixed-frequency low-pass filters can be adjusted to set the maximum brightness of a sample. In the case of sampled bass this is ideal for minimizing unwanted tape hiss or amplifier noise. Bass notes are also subject to quantization noise, and lowering the filter cutoff helps to mask that.

Envelope control of a low-pass filter's cutoff must increase a sampler's usefulness by at least ten-fold. Even a simple four-stage ADSR envelope on a filter offers so much more scope to the manipulation of samples that a serious "pro" sampler wouldn't be caught dead without one (hear that, guys?).

As nice as a short loop's tone color might be, the human ear gets bored listening to sounds with no harmonic movement for even short periods of time. Bass samples with short loops benefit greatly from clever use of a enveloped filter. Try setting the attack and decay controls to zero and, while holding down a note on the keyboard, adjust the sustain of your filter's envelope to let only the first few harmonics of your short loop through – the tone color you'd expect to hear just before a bass note dies away to nothing. From there it's a simple exercise adjusting the decay control to start the sample off bright and decrease the filter cutoff to the sustain level over, say, five seconds, simulating the natural decay characteristics of plucked strings.

By manipulating the filter envelope and feedback amount, an effect resembling the old faithful funky bass synth can be superimposed on a sampled bass. Fun stuff. More snap can be added to the attack portion of samples with a filter and by fading in both the filter and amplifier something approaching a bowed string bass can be achieved. Faded-in harmonics border on the ethereal.

Modern instruments and studio techniques allow the tailoring of bass sounds as never before; sampling enables us to store these sounds on floppy disk ready for instant retrieval. So what's your excuse? Get your hands on an electric bass, start sampling and get your bottom end together. ■

REVIEW

Oberheim Matrix 1000

Analog Synth Module

Imagine buying 1000 useful patches for 60¢ each with a synthesizer thrown in for free.

Review by Thomas J. Clement.

YOU CAN'T THINK of the Oberheim sound without the following adjectives: fat, warm, complex, classic, *expensive*. Oberheims have always been out of reach of most mortal musicians who had to content themselves with drooling at the Big O from afar. It was rather like coveting a new, sun-swept Cadillac convertible complete with sun-swept blonde passenger while you made do with a dog-common hatchback.

Long after its competitors dropped the prices of their digital keys below the $2500 mark, Oberheim's machines continued to hover in the rarified region of "pro" gear. If you've been hoping to save enough dinero to buy OB's time-honored flagships, the Matrix 12 and the Xpander, keep saving. They've just been jacked-up another grand apiece (about $6000 and $4000, respectively). Yeesh, the price one has to pay to get a synth with wooden end-pieces!

Fortunately, the folks at Oberheim took pity on we of the smaller pocketbook and released the Matrix 6 and the keyboardless Matrix 6R (see MT September '86 for review) a few years ago – two scaled-down, but still impressive-sounding versions of the big guns. The Matrix 6 is $1700 and the Rack is just over a grand (why can't other manufacturers slash $650 off the prices of their rack conversions?).

But, the masses wanted that Oberheim sound to be even cheaper (we are a demanding lot, aren't we?). Thus comes the Matrix 1000 with a sticker under $600. The 1000 maintains the same voice architecture as the Matrix 6/6R – including two DCOs, a VCF, two VCAs, three 5-stage envelopes, two LFOs, two ramp generators, an FM modulator and a tracking generator – but sacrifices front-panel programmability. If you're interested in tweaking your own sounds, editing is possible via a computer and any Matrix 6 Editor/Librarian program. Most of the hoopla surrounds the 1000's on-board "warehouse" capacity, however, so let's pry into this first.

How Many Sounds?

IN THE PAST, players had to content themselves with skimpy synth memories, usually about 64 locations. If you wanted more room, you could use the dreaded tape backup, or you could pop at least 75 bucks for a cartridge that doubled the synth's memory. Some machines have disk drives built in (Ensoniq's SQ80, Roland's D20, and Yamaha's DX7IIFD), but these are mid-market machines and, while they can store tons of sounds, some juggling is required to get at their libraries.

Enter the Matrix 1000. Picture this: you're in a deadline situation. The producer asks for a different bass sound. No problem. You've got at least 119 to choose from just a patch change away. More attack in the strings? Surely one of the 118 built-in string patches is made to order. How about a plain-vanilla synth? Well you don't just have vanilla. This box has 239 synth flavors! And these are just some of the ROM sounds – *they* make up 80% of the 1000's innards. And though you can't program the 1000 from the front panel, the remaining 20% of its memory can be filled with any Matrix 6 or 6R patch you desire. That's 200 writeable locations. Is there another synth with that much instant-access RAM?

Patches are stored in 10 banks of 100 each (if your controller only sends out the first 64 patch changes, sorry). The first two RAM banks are what amount to "greatest hits" from the other eight banks. Any ROM sound can be moved to any RAM location in seconds.

To change from one bank to another, use MIDI Controller 1, the Mod wheel. By holding the Mod wheel at 50% or above and pressing one of the first 10 program changes on your controller, you can access any bank (each bank responds to the same 100 patch changes). Piece of cake. If a sequencer is in charge, you can also use MIDI controller 31 (a non-standard assignment) to assign bank changes (again, that setting must be 50% or above). If you want to break down and use the front panel, just enter the three-digit number of your choice. Three digits too tiring? Oberheim provides a special button for synth potatoes which allows you to lock on a specific bank so only two numbers need to be entered. Up and Down increment buttons are also on hand. Only complaint here is that the inc/dec buttons don't have a high-speed scroll function. If you're used to relying on a controller's cursor keys or data slider to zoom through patch changes, you'll find that the 1000 is easily left in the dust. A minor annoyance, but when travelling from patch 0 to 99, zoom is the speed of choice.

Eyes and Ears

NOW TO TOUR the rest of the machine. First there's the size: just one rack space (I love these). The front panel is well designed, clearly marked, and intuitive (the buttons have that OB feel, too). A red LED blinks when MIDI signals are received. Then there's a luxurious length of yellow MIDI cable; none of those chintzelburger 3' cables for Oberheim. Finally, there's one detail that took me by surprise. The Matrix 1000 is . . . is . . . mono!

I looked the unit over about a dozen times, turned it upside down, probed for hidden compartments, recited the stereo mantra, but to no avail. It was indeed mono. A sense of overwhelming grief pervaded my inner being. Then I sat down, thought things through, and bucked myself up with the facts. Few synths are truly stereo. You can duplicate the stereo effect on most machines with a "Y" connector, a couple of cables, and some stereo effects (eg. a chorus device). Still, couldn't

Oberheim have put a keyboard tracker on the 1000, like the one Ensoniq uses on the ESQ? Oberheim's Jim Letts put it this way, "a panning circuit would add $200 to the unit and we wanted to keep the price down for musicians. Even the Matrix 6 and 6R aren't true stereo. The L/R jacks work best as direct outs for the splits." Oh.

He went on to say that playing the same patch on two different 1000's that were panned to separate mix locations provides a "must hear" effect. The oscillators from each machine run differently at slightly different times, so they actually oscillate against each other. Sadly, I was unable to test this.

Enough of the tour. How does it sound? Well, it sounds just like a Matrix 6: big, fat and rich. The Matrix series gets its unique sound from Oberheim's "Matrix Modulation." Basically, this is a software version of the monstrous patch-cable synthesizers of yore. Twenty different sources can modulate 32 destinations. If you're familiar with programming in Matrix, you're aware that some of the modulations are produced via controller wheels, footpedals, velocity, etc. For about 5% of the 1000's patches, portions of the sound can be modulated in this way. Unfortunately, Oberheim doesn't tell you which ones, so experimentation is required.

A good rule of thumb for any synth is that half the presets will range from serviceable to excellent; the other half from boring to bogus ("Motorcycles from Mars" anyone?). The ratio of quality patches is greatly increased on the 1000. Naturally, there are some clunkers; a few patches are too similar to others; and some only sound good over a limited range; but over all, this is a library to covet. Some of the imitative patches would give a sampler or wavetable box a run for its money. Some of the envelopes are jaw-droppingly long (you don't so much play a patch as explore it). And some of the sound effects, usually so annoying on other machines, are intriguing and inspiring here. And, of course, the Matrix 1000 does what only a top-notch analog synthesizer can do: soak a track with rich, fat sound.

Written Matters

ONE MINOR COMPLAINT about the 1000 is its patch sheet – a thin-papered, non-laminated, two-sided affair with what appears to be Helvetica 7-point type. More pages, larger type, and coated stock should be *de rigeur* for us beverage-spilling, squinty-eyed users.

Speaking of printed matter, let's cover the manual – usually a bugaboo for most synth makers. Music Technology regularly takes manufacturers to task for the poor quality of their manuals, because they *are* important to either someone learning or in a hurry in the middle of a session. With that, kudos to Oberheim – in about 50 breezily formatted pages, the authors go step-by-concise-step through just about everything you could possibly want to do with or to a Matrix 1000: how to rack mount it; how to control it from a MIDI guitar (mono mode); how to chain it with a Matrix 6/6R (software 2.13 is required); etc. There's also a Quick-Start Page, subtitled, "If You're in a Hurry . . . ," which kind of tells you that the manual is on your side – not a pain in the same. I noticed at least one omission: how to change the bend-wheel setting (it's Ext. Funct. 8, then press Enter and use the up/down buttons), and the Table of Contents is sketchy, but overall this tome gets top marks.

Conclusions

IS THE MATRIX 1000 worth the price? I think so. Sure, you could easily say, "Why spend six bills on a machine that only plays six notes and one sound at a time when I could buy another synth for the same price that plays a zillion sounds at once and comes with a built-in stage lighting system." And if you've got to have polytimbrality for your next purchase, the Matrix 1000 is definitely in the cold. It also has no onboard processing frills (though some of the patches feature delays and effects no processor could mimic).

What it *does* have is impressive patches – and lots of them. You might see it this way. If you've already invested in a Matrix 6 or a 6R and would love to expand that sound, then this is the ticket. If you've always wanted to add quality analog to your digital setup and wouldn't mind gobs of patches at the push of a button, then the 1000 is also an economical way to go.

You can also look at the 1000 as a modular part of a larger system. For about the same price as you'd pay for most other currently hip synthesizers, you could buy three 1000s, yielding 18-note polyphony and three sounds at once. And if your wallet has the horsepower, you can string up to six 1000s together for a 36-note instrument – just the thing for those grand-canyon-deep filter sweeps.

Ordinarily, when playing a machine, the wish list grows with time. Conversely, with the Matrix 1000, my wish list shrank as I came to appreciate the simplicity and quality of the unit. ■

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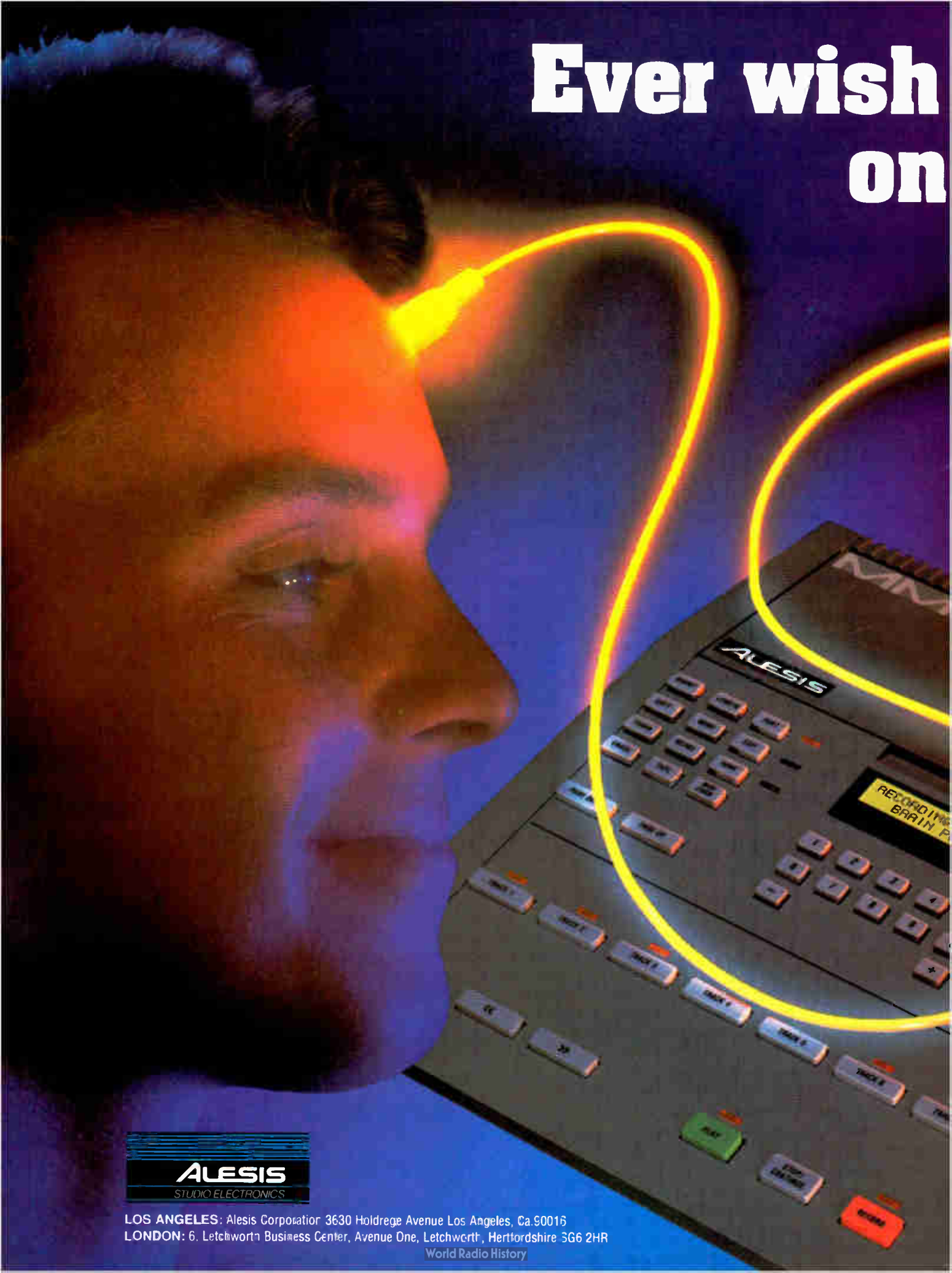

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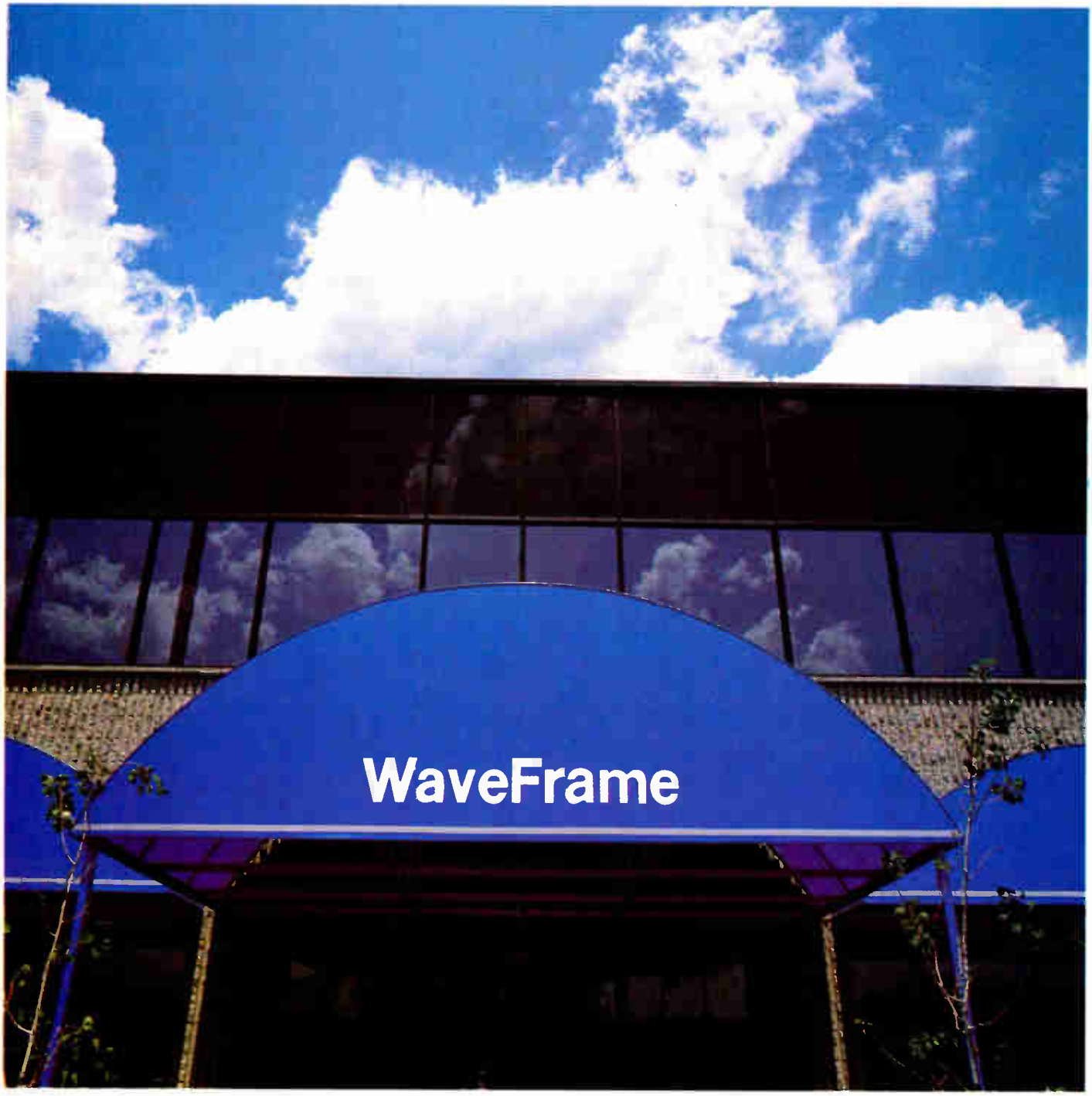
FUTURE POSSIBLE

THE AUDIOFRAME EXPLAINED

part one: the architecture

TECHNOLOGY



Photography *courtesy of WaveFrame*

Nestled in the college town of Boulder, Colorado, a team of hand-picked people at WaveFrame are attempting to build *the* digital audio workstation. Here is what it's all about.

Text by Chris Meyer.

IT'S A DREAM all of us who've had any yearning to design equipment has had: "Let's build something totally new, starting from the ground up. You know, do it correctly – think of a really good basic architecture, use the latest technology, hire the best people, and do it *right*." Normally, the dream ends abruptly and we then go back to our ordinary lives – or reality, if we already happen to work for a manufacturer that designs equipment.

WaveFrame Corporation, however, has been trying to live out that dream. They can't afford to go back to "reality" because they're taking on the well-established New England Digital Synclavier and Fairlight's Series III. WaveFrame is trying to beat these two older systems by choosing a starting point for their hardware that's at a more advanced stage than where NED and Fairlight currently are. At the present time, however, their product – the AudioFrame – is still in its infancy. If you want to talk available-to-anyone, non-beta, debugged hardware and software, its competitors outspec it as of this snapshot in time (late June, when I visited WaveFrame). But I'll reveal some of the plot early – I'm impressed with what I've seen, and feel pretty good about where it's going.

This four-part series will take a close look at the AudioFrame: explaining the foundation WaveFrame's created to build on, looking at where the system is today, and examining where it's going. I'll start from the ground up myself: the hardware architecture of the AudioFrame.

The Two Boxes

THE AUDIOFRAME PRESENTS itself as two units: a Compaq 386 computer (which may be changed to an ALR 386 in the future) and a large rack-mountable black box known as the Digital Audio Rack (DAR). The computer (running Microsoft Windows 2.0 – a Macintosh-like environment for IBM-type computers) provides the user interface and sequencing; the DAR (which is based on removable cards, as opposed to being hardwired, so it can be configured as needed) is where all the sound flows. The two are connected by a 4MBaud token ring network.

The computer deals strictly in commands. Editing commands are passed onto the DAR to be executed; while real-time jobs the computer may be running, such as the sequencer ("Event Processor") and adjusting sliders on the virtual mixing console (mixing will be covered more in depth in the third installment), are communicated to the DAR as MIDI commands via virtual MIDI cables. There are 32 "cables" inside the system (two physical ins and outs are on the back of the Studio Control Processor card on the DAR, along with SMPTE, VITC, SCSI, and token ring connections). Next month's installment will concentrate on the ▶

▶ software that runs on the Compaq.

All of the software that the DAR runs (each card in the DAR has its own processor) is stored at the computer, and is downloaded to the DAR at power-up (no replacing ROMs in the hardware every time there's a software update). Samples used to be stored at the computer, too, but it ended up taking too long to transfer them to the DAR. The new SoundStore subsystem keeps the archived sound data local to the DAR.

The DAR's Studio Control Processor directs traffic. The highway that sound moves around on is referred to as the AudioFrame's Digital Audio Buss (DAB). This buss can route up to 64 monophonic channels of 24-bit audio data around the DAR. A single "Sampling Synthesizer" sample playback card in the DAR can support up to 16 dynamically-allocated voices. A DAR can handle up to eight of these cards, yielding 128 voices.

Wait a minute, I hear you saying, 128 voices, 64 channels of audio. What's going on here? Ain't that too many? No, actually it's not. But in order to explain why, we're going to have to start delving into the intricacies of the AudioFrame's hardware, and admire a bit of the solid foundation they've created.

Constant Sample Rate

AUDIO, IN ITS natural analog state, is easy to mix and pass around - just wire it up, and sum it either by twisting wires together or going through a little analog circuitry. Digital audio isn't as easy to deal with. Remember that analog audio is converted to digital by measuring its voltage level at given intervals and converting that into a number. Digital audio is mixed by adding these numbers together. Further processing is performed by running these numbers through more complex math functions. However, to mix (add) two signals (numbers) together and pass them on to another part of the system, they have to be present at exactly the same time. Once they are passed on, the next two to be processed also have to arrive in sync. The system may be built to survive deviations in this, but severe ones (like two numbers arriving from one source while waiting on the previous number from another source) will result in less-than-perfect audio.

For synchronization to happen, all digital audio streams in a system must measure (sample) their respective analog inputs at precisely the same rate, and all circuitry that processes these signals or creates their own must also operate at that rate. This is what is referred to as a "constant sample rate" system.

Buy why bother keeping things all digital, if mixing them is so much easier in the analog domain? Most functions in these modern wunderboxes are digital in the first place (sample playback certainly is; so are such functions as digital filtering, digital reverb, and mixing multiple dynamically allocated voices to one output jack) - might as well keep the language common. Also, translating at every stage from digital to analog and back to digital gets expensive and impractical. As with any translation, something always ends up getting scrambled to some degree; in audio, that means more distortion or noise.

One of the simplest things a constant sample rate buys you is the ability to do premixes inside a processing card. The AudioFrame's Sampling Synthesizer cards can support 16 voices each. However, they can all be assigned to the same, say, two channels of audio (for stereo). This way, these 16 voices chew up only two channels of the 64 available on the DAB when they finally hit it. Premixing like this also makes it easier to process the group later (such as running them all through a digital reverb algorithm running on a separate card).

Any number of "instruments" can share these voices; the card will dynamically allocate its resources as needed. Each instrument can be assigned a different audio channel, just to keep them separate inside the DAR, with dynamic allocation keeping the voices/instrument/channels distinction transparent. When a flute gets asked for a note via MIDI, the playback card has a voice play the flute sample, mixes it with any other local voices playing the flute, and places it all on the same channel. This is one of the main differences between the AudioFrame and the Synclavier or Fairlight Series III - the others are variable playback rate boxes, and need to be mixed externally.

One thing you can't do with the AudioFrame is have two physically separate cards output on the same channel - there's no way for them to mix themselves. What has to be done is the two cards have to take up two separate audio channels, and then be submixed to a third by a mixer algorithm running on another card. This isn't too far off from having a normal audio mixer combine a tape track with a live vocal before running off to the reverb. One advantage of this closed system is having several voices play back a stack of samples on one channel, and have another part of the system sample *that* (the high degree of segregation of duties inside the AudioFrame means it can do things like play and sample at the same time).

Channels and Bits

AS MENTIONED ABOVE, the AudioFrame's DAB handles 64 channels of 24-bit linear audio. (In other words, each "word" of digital audio information consists of 24 bits.) This word size permits an internal dynamic range of 144dB, which conforms to the highest AES (Audio Engineering Society) standard. A sound

pressure level of 144dB (the extreme of this dynamic range) is also the maximum sound level that can be physically produced (inward excursions of a speaker cone would have to suck air hard enough to form a perfect vacuum), and is well beyond the threshold of pain. The buss is parallel, so all 24 of a given channel's bits are available at a time. Each channel runs at a rate of 44.1kHz, which means that the buss has to run at a speed of over 2.8MHz. This translates to a mere 350 nanoseconds of attention for each audio channel before the next one gets a shot. Although it's easier to think of these 64 slots as rotating in order, in reality the DAB allocates time slots as requested. This means that no pair of signals that are supposed to be grouped together (such as stereo) are ever skewed by more than 350nsec. The output card of the DAR also compensates for these delays, resulting in perfectly locked stereo (for more on the vagrancies of sampled stereo, see the 'True Stereo Sampling' sidebar in the EIII review last month).

Separate input (Analog to Digital – A/D) and output (Digital to Analog – D/A) cards are available for the DAR. The input card (two- and eight-channel versions are available) digitizes signals and places them on the selected DAB channels; the output cards (eight channels to a card) take the premixed audio off selected channels and feed it to the real world. The A/D conversion uses 18-bit parts; WaveFrame puts the quality at a true 15+/16 bits (converter specs alone are worthless if the analog support circuitry isn't up to snuff). WaveFrame also intends to make a "gold card" version available with oversampling that is supposed to be good out to 17 bits or more.

Output is 16 bits, oversampled at a rate of four times the internal 44.1kHz. According to WaveFrame this gives a response that's "essentially 18 bits." Huh?

"To mix two (digital) signals together and pass them on to another part of the system, they have to be present at exactly the same time."

Time to enter back into the world of digital theory.

Chopping an analog audio signal up into little slices of time and back again *does* – shock, horror – create noise. And some of the artifacts caused by representing a normally continuous waveform as a bunch of splices appear in the audio band of frequencies. Another way of looking at it is that the reassembled signal has added noise, and this noise fills the space from half the sample frequency on down.

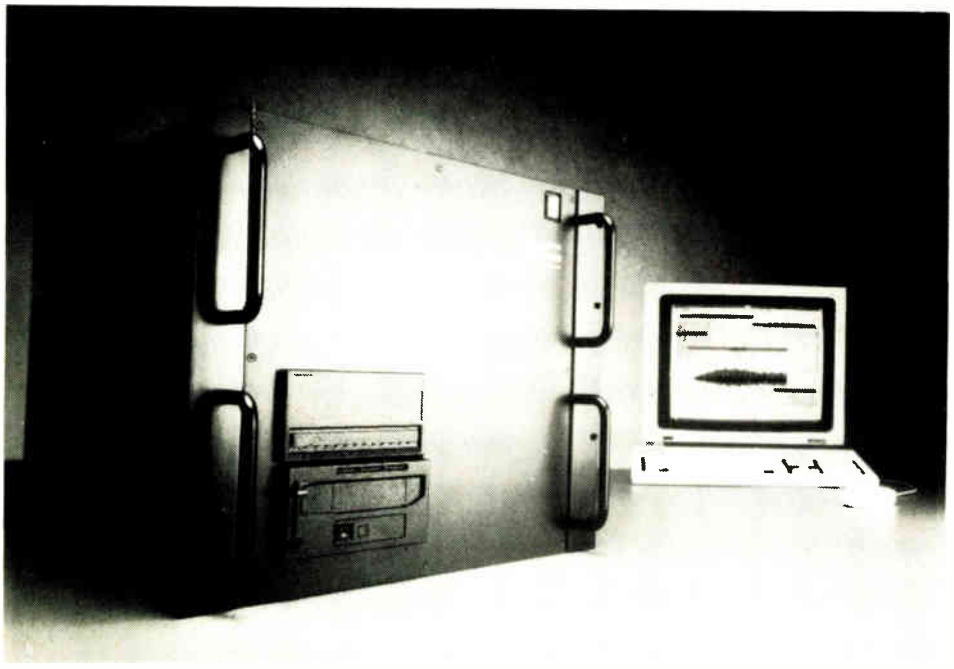
The faster you sample a sound (the finer the slices), the more it will look (and sound) like the original input source. Continuing with our alternative view, this means that a higher sample rate will spread the noise created by reconstructing the signal over a wider range of frequencies – not all of which will be audible anymore. If something is played back at 44.1kHz, there's a certain amount of noise spread from 22kHz on down. If the same signal's played back at 88.2kHz, the same amount of noise is spread from 44kHz on down. Consequently, half of the noise would then be in frequencies out of audibility ("audibility" being from 20kHz on down, assuming you haven't been mixing heavy metal for a living). This yields an effective improvement in signal-to-noise of roughly 6dB.

Increase that playback rate by a factor of four, and now roughly three-fourths of the noise is inaudible – resulting in a 12dB improvement. Since 6dB of dynamic range is the same amount provided by one bit, that explains why WaveFrame can say they have "essentially" 18 bits of playback. Before you get too wowed, most CD players do this, too. And for the record, if you've run across a Synclavier owner who claims to get 6dB better signal-to-noise by sampling at 100kHz, this is also what they're talking about.

In both cases, the higher playback rate means a less severe filter can be used to reconstruct the signal back into smooth ▶

► analog, which means fewer phase distortions in the high end. The AudioFrame uses a gentle third-order linear phase filter which is down only 3dB at 40kHz. On the input, they use a ninth-order elliptical anti-aliasing filter (for those who remember that an odd number of filter orders means the phase gets flipped, three plus nine adds up to twelve, so the signal is correct when it comes back out).

The AudioFrame pulls a couple of more output tricks to optimize signal-to-noise and headroom. For one, dithering may be added to the output. This is a form of noise that is added to the signal before it is converted back to analog. The dither noise helps fake an extra half bit out of the D/A converter (by making a *very* low level signal cross a bit boundary it might not have). Dithering is good to add to anything used in isolation; if several AudioFrame outputs are to be mixed outside the box in analog, it causes a problem – the additional noise per output gets added together and becomes rather objectional. Nicely, the AudioFrame allows you to turn dithering off.



AudioFrame Digital Audio Rack with SoundStore.

The user can also trim the amount of headroom available in the system. If you have just one voice aimed at a single output, of course you'll want to use all sixteen available bits. However, if you have a loaded machine with 128 voices all being mixed to one output, the voices have to be pulled way back in level so as not to clip. This means each individual voice would have to be operating at a maximum of nine of the available sixteen bits (because two full level signals, added together, would add up to ten bits; four would add up to eleven, and so forth). Nine-bit resolution translates to a 54dB dynamic range, which adds up to a lousy signal-to-noise ratio, particularly if you're only using one voice. The user can adjust this trade-off of headroom versus signal-to-noise for their own real-world application.

Sample Rate Conversion

NOT TRYING TO jam too much down your throat at once, I left out an important point up top when talking about constant sample rates. If you sample a sound at 44.1kHz and then play it back at the same rate, fine. But what if you want to transpose it – say, down a semitone?

You have to slow playback down. But the signal still has to be running at 44.1kHz when it hits the DAB. To pull this off, the Sampling Synthesizer card has to "guess" what the signal would look like if converted back to analog and resampled at 44.1kHz. Just how good this guess is directly relates to how good the transposed signal sounds. This is an area into which WaveFrame's engineers put a lot of thought and study.

When you're transposing while trying to keep a constant sample rate, you have to approximate what the signal would have done during the desired time between two existing samples. Most samplers based on a constant sample rate use a linear approximation – they mentally draw a straight line between the two existing samples, and create their intermediate sample from a point along that line. This makes the assumption that all sounds look like sawtooth or triangle waves – which they don't. Waves tend to curve about; therefore, the Sampling Synthesizer cards have to create a curve that fits the contour of the existing samples. Errors in this curve create distortion.

Creating a perfectly fit curve consumes a lot of processing power – too much to pull off practically in real time. Here is where the folks at WaveFrame started throwing out "conventional wisdom" in favor of studying how the ear really works to reach some elegant compromises. Just a ripple in the frequency response – say, a few tenths of a decibel – measures as a ton of RMS distortion. However, the ear won't detect it. White noise doesn't bother the ear that much, either. The ear also won't detect even-order harmonic distortion (ie. second order – a perfect octave above the original signal, like tube amps produce) that stays in tune with the original signal – WaveFrame figures the ear can take a level of −40 to −50dB without getting too upset.

However, the ear hates inharmonic distortion (that which is not an integer or nice musical multiple of the original signal) – particularly if it doesn't track the original as it's transposed (these are the "birdies" or "artifacts" you see us refer to in our product reviews). Those have to be at least −90dB down for the ear to let you off the hook. WaveFrame took this information and decided to make inharmonic distortion the main thing to suppress while trying to keep the trade-offs in other distortions to a minimum.

WaveFrame's curve-fitting algorithm looks at eight samples before and after the one under scrutiny and is optimized to keep inharmonics down. In fact, they have tested the result to be −108dB down, with RMS ripple on the order of a few hundredths of a decibel – less than that in typical audio cables, according to vice president of engineering John Melanson.

The final number in the equation is how many artificial samples are created between the existing ones – these are the ones the system ends up playing back, so the more there are to choose from, the less likely there'll be the kind of distortion (or tuning) problems that result from picking points that aren't quite the ones you really wanted. The Sampling Synthesizer cards create 512 points for every existing sample. The bonus that falls out of this is that these 511 extra points are also used to pick zero crossings from for looping. Therefore, there's a better chance of picking a point where the original signal would have actually crossed zero, and single-cycle waves are better in tune (no need to settle for points just short or just long of where the signal would have crossed).

Road in Place . . .

SO, THE AUDIOFRAME has been discussed up to the point where there are multiple channels of very high quality digital audio floating around inside to play with. Next month, I'll go over the user software and remaining DAR cards that are currently available. From there, the AudioFrame's near-term Future – as a virtual mixer, as a hard disk recorder, and as a synthesis machine. Until then . . . ■

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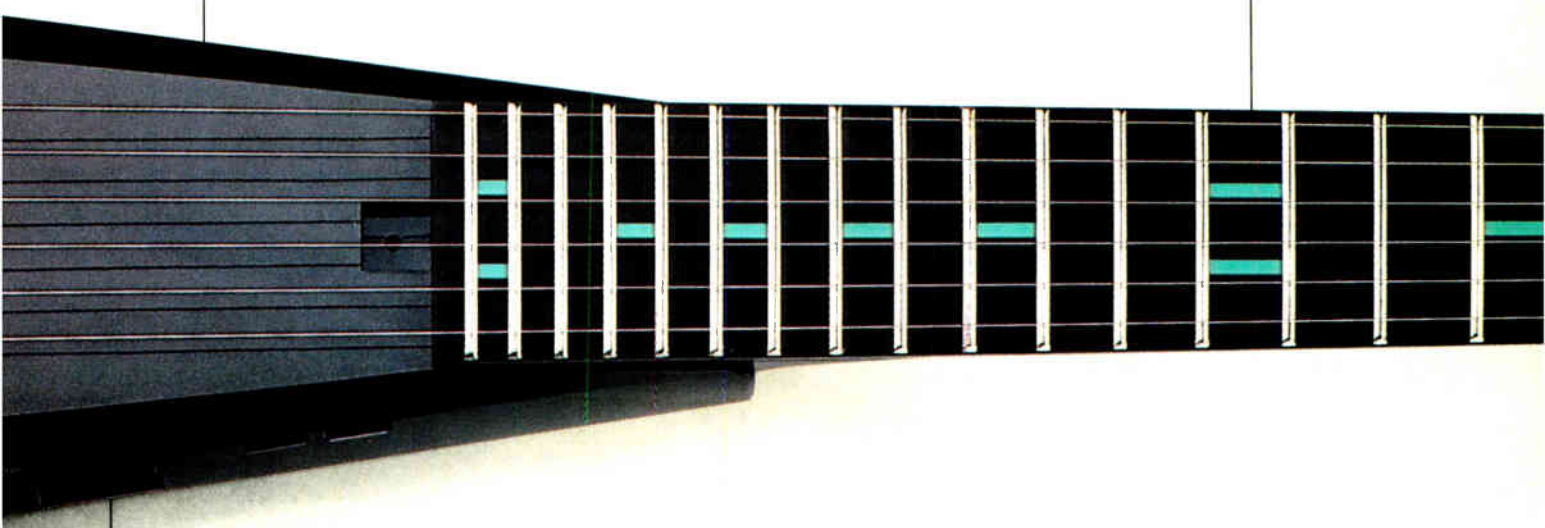

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Casio VZ1
iPD Digital Synthesizer

REVIEW

Photography Rose Rounseville

Big new sounds, a variation on FM and a built-in graphic display are all standard on Casio's latest model Z. *Review by Bob O'Donnell.*

BACK IN 1985, when Casio first released the CZ101, they pulled off a rather impressive marketing coup. They convinced a lot of people that the Phase Distortion (PD) synthesis method found in their CZ series instruments was fairly similar to easily understood analog principles. Remember the Digitally Controlled Wave (DCW) parameter that acted just like opening and closing a filter? It actually adjusted the modulation amount of a "control" oscillator – just like a modulator in FM. But by using familiar terms and explanations and avoiding the confusion that Yamaha started with the introduction of the DX7 and FM digital technology, they disguised the wolf of digital synthesis in the sheep's clothing of analog. The truth of the matter, however, is that Phase Distortion is actually a neat variant of FM.

That makes Casio's announcement of a new instrument utilizing a technology called interactive Phase Distortion (iPD) – which even more resembles FM – a bit less surprising. The VZ1 is not a DX7 clone, however; let's get that clear from the start. Though it does share some basic principles and sonic characteristics with DX instruments, it also maintains several of the features found on the original CZ series, and adds many advanced capabilities of its own. But I'm getting ahead of myself.

The Features

BASICS FIRST: THE VZ1 is a 16-voice, multitimbral synthesizer with a 61-key velocity and pressure-sensitive keyboard. For better or worse, there are no samples, effects, or sequencer built-in – this is a synthesizer, pure and simple (or complex, in the VZ's case). The back panel features two independent outputs and a single mix output, all 1/4". The usual complement of MIDI In, Out and Thru jacks are also provided, as is a handy little MIDI on/off switch which – obviously enough – turns the transmission and reception of MIDI data on or off. Other connections include a headphone jack, and inputs dedicated to a sustain pedal, a volume pedal, a footswitch for stepping through programs, and a programmable footpedal – or "foot variable resistor," as Casio calls it.

Multitimbral *"The Multi Channel mode is for creating just one multitimbral setup for the VZ1 when it is to be used in conjunction with a sequencer."*

The front panel of the VZ1 is where things start to get interesting. Off to the

left side are not two, but *three* performance wheels – the first is dedicated to pitch-bend and the other two can control a variety of different effects that are programmed per patch. The middle control is often used for modulation and hence has no type of spring loading, but the third wheel is spring-loaded and will return to its original bottom position when released.

The VZ1 features internal memory for 64 normal patches and 64 performance-oriented "Operation Memories." (Operation Memories allow you to combine up to four different patches in any configuration of splits or layers that you like.) In addition, the VZ1 comes with a ROM card which holds 128 more of each. By sliding the card into the single slot on the instrument's back panel, you have immediate access to an impressive 192 each patches and Operation Memories (RAM cards which can hold 64 of each are also available). The VZ10M, a three-space high rackmount version of the instrument, offers similar capabilities and connections, but adds an XLR mix output for studio use.

The Organization

THE VZ1 IS organized into four basic modes of operation: Operation Memory, Normal, Combination and Multi Channel. The first two correspond to the types of patches stored in the VZ's memory. To play either of those types you simply enter that mode by hitting the appropriately labeled button, and select from what's available. When you want to edit normal patches you enter the Menu sub-mode by hitting one of the three Menu buttons – each one is dedicated to a particular set of parameters – and proceed to the appropriate software page on the display. Easy enough. Things aren't quite as simple with Operation Memories, however, because they can only be edited in the Combination mode – which is really nothing more than a working edit buffer. Unfortunately, you have to first write the Operation Memory you want to edit into the Combination mode – it won't appear there automatically. It's not a big deal to go through the extra step, but I really don't understand why the Operation Memory mode couldn't have had its own edit sub-mode.

Moving on, the Multi Channel mode is for creating just one multitimbral setup for the VZ1 when it is to be used in conjunction with a sequencer. (I find it hard to believe that more memory wasn't set aside for this function.) The VZ can operate on up to eight channels at a time, playing eight different timbres. Each part (or "Area," as Casio refers to it) can be assigned one normal patch, have its own fixed polyphony (unfortunately there's no dynamic allocation), its own output level, and its own MIDI channel. In addition, the tuning of each patch can be set per Area (over an amazing 10-octave range) so that you can recreate some of the detunings used when combining patches into Operation Memories. As for outputs, Areas 1-4 are all hardwired to output 1 and 5-8 are hardwired to output 2. On the MIDI and performance side of things, each Area can be programmed to respond individually to solo mode and portamento parameters, as well as pitch-bend, aftertouch, the two definable controller wheels and the footpedal. All in all, it's not exactly the most flexible structure I've seen, but it gets the job done.

iPD Synthesis

THE BASIC IDEA behind iPD is that you have eight modules – each of which contains a multi-waveform digital oscillator (DCO) and a digitally controlled amplifier (DCA) – which can combine in various ways to create a patch. The individual modules can either directly contribute to the final sound or can be used to modulate the waveforms in the other modules.

Modules are combined together in groups of two, known as Internal Lines. Within each Line, the outputs of the two modules can be mixed together, or one can be used to ring modulate or modulate the phase of the other. In addition, the final output of each Line can be used to modulate the phase of modules in the next Line. Casio refers to this as "External Phase" (see Figure 1).

For those who may be unfamiliar with ring modulation, the output is a static wave whose frequency content is the sum and difference of the harmonics in the two source waves. The result is usually quite complex, and is often used to create bell-like tones. Most ring modulators throw away the input signals in favor of the newly-created output, but Casio lets you keep one of the sources to mix in. Phase modulation, on the other hand, involves altering the timbre of a sound by frequency modulating the source waveform. The result is determined by the DCO's waveform and the DCA's envelope shape in the "modulating" module. As the amplitude of the modulator increases, the waveform generated by the sound-producing module "grows" more high harmonics (resulting in the sound getting brighter).

On a basic level, you can think of phase modulation as giving you a filter-like control over a single module. The difference between this and a standard low-pass filter is that in the latter case, all harmonics below the cutoff frequency remain unaltered, while in the VZ1 – or any FM synth – the lower harmonics are also affected when the modulation amount increases and the waveform gets brighter.

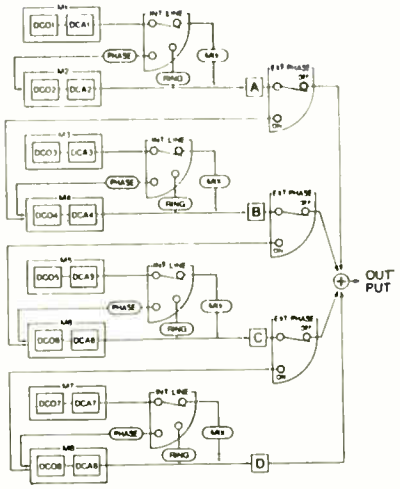


Figure 1. The voice structure of the Casio VZ1.

The various combinations you can achieve with the VZ1's voice structure range from mixing together the independent waveform outputs of eight different modules to having seven modules all modulate the final module. In a certain sense, you can almost think of iPD as 8-operator, multi-waveform FM with user-definable algorithms. Anyway you look at it, though, it's a pretty powerful synthesis system. ■

The Voice

NOW FOR THE fun stuff. The VZ1 gets its distinctive sound from its iPD sound generation system (*see sidebar for more details on iPD synthesis*). With it, Casio gives you control over eight modules –each of which consists of a Digitally Controlled Oscillator (DCO) and a Digitally Controlled Amplifier (DCA). One of the reasons that the VZ sounds as big as it does is because it offers a choice of eight different waveforms for each of the oscillators. In addition to FM's expected ▶

► sine, there are five different sawtooth waveforms with increasing amounts of harmonics and two noise waveforms (one with a fundamental frequency that can track the keyboard, and one that's just plain noise.)

By accessing Menu 1 (the voice editing menu), you'll find that each of the DCOs can be tuned over an amazing *theoretical* range of ten octaves and either track the keyboard or remain at a fixed pitch. I say theoretical because both the high end and low end of the range cannot be accurately produced by the VZ – which makes me wonder why they offered that large of a range in the first place. On the high end, in particular, the VZ ends up producing all sorts of strange alias-like noises and other warbles. On the positive side, one nice touch included in the tuning functions of the DCOs is a Harmonic parameter, which will instantly set the musical interval defined by the selected harmonic (ie. the third harmonic is an ocatve and a fifth above the fundamental). This is just like FM's ratios. It can be set over a range from 1:63 (1/63, in Casiospeak) to 63:1.

The amplifier (DCA) portion of each of the eight modules per patch has an eight-stage rate and level envelope generator (EG) – true to Casio tradition – which contains one major difference from all previous Casio envelopes. The Level amount of the end point does not automatically get set to zero: as on the DX7, the "release" level can have a positive value to produce drones or to increase the modulation amount when a note is released. A single eight-stage pitch envelope is also available to globally affect all eight modules.

Once you've created a sound you like, the VZ allows you to do a great deal of fine tuning for tweaking it into a performance-ready, keyboard length patch. First off, each of the nine EGs in a patch (eight modular DCAs and one global DCO) has a depth control to determine the overall level of the envelope. In addition, the rates *and* levels of all EGs can be affected independently by user-definable, six-point Key Follow curves as well as by velocity. In other words, the keyboard position of the note you play and the strength with which you play it can alter any envelope's shape – either by moving the levels up and down, making the rates longer and shorter, or probably a combination of both. You can program a single six-point Key Follow curve which globally affects the overall rate of any of the nine EGs and program nine independent six-point curves to affect the overall level of any single EG.

In addition, you can choose from eight different velocity curves (actually 16 if you count the fact that you can invert them) and 32 sensitivity amounts for each EG to control how velocity affects its overall level. Finally, you can select one curve to affect the rate of all the EGs – and in this case the VZ1 actually allows you to select which step(s) of each EG you want to be affected. So, for example, you could have velocity affect just the attack and release segments of the pitch envelope and the first four segments of the DCA envelope in module 3. It's definitely one of the instrument's shining points.

Other parameters available per patch include an octave function and independent global settings for vibrato and tremolo – including a choice of four waveforms, rate, depth, delay and multi (key sync) controls. Last, there's an amplitude sensitivity setting per module, which determines how strongly the modules will be affected by tremolo or amplitude envelope bias.

Vibrato and tremolo rate and depth, positive or negative pitch-bend, portamento time and the aforementioned amplitude envelope bias (which affects the overall level of selected modules) can all be controlled independently by aftertouch, the two definable wheels, and the footpedal. These routings may be defined per patch. The pitch-bend range is four octaves in either direction and if you use it at this extreme you can get some slowed down sample-like effects.

On the MIDI front, the VZ1 permits you to transmit any MIDI controller from 12-31 with the second definable wheel. In addition, MIDI volume transmission can be enabled or disabled and you can initiate SysEx dumps from the front panel – which is a big improvement over previous Casio products.

Once you've got some individual patches you like, you can put up to four of them together in the Combination mode (though polyphony will be reduced to four notes if you do). As with the Multi Channel mode, each patch can have independent pitch, and level amounts, and can respond independently to solo, portamento, pitch-bend, aftertouch, definable wheel and footpedal settings. In addition, individual patches can be set to either respond to or ignore sustain pedal messages. Most importantly, patches can be grouped together in eight different keyboard assign modes, with various combinations of splits and layers. Up to three independent split points can be set. If you'd prefer not to have hard splits, you can use the VZ1's impressive positional crossfade function to smoothly spread out the various patches.

If you're working with a layered patch, you can set up to four different velocity splits, with user-definable velocity ranges. You can also inverse the velocity response of any patch for velocity crossfades, and delay each of the four patches by an independent amount. Finally, you can invert the vibrato and tremolo effects independently per patch.

Sound *"The iPD synthesis system used by Casio provides some similar ground to those who are familiar with FM, but gives you the flexibility to do even more."*

Now, this may all sound confusing on paper, but actually working with the VZ is a lot nicer than you may at first expect. How? Read on . . .

The Interface

CASIO HAS USED the "manual on the front panel" approach with the VZ1 and, as a result, I initially found the front panel to be a little busy. I have to admit, however, that having the various parameter options

printed on there did come in handy, once I started to become more familiar with the machine and its operation.

What I *didn't* find particularly handy, was the way you go about moving through the parameters. Once you select one of the Menu buttons for editing you can scroll through the various functions, but to edit any of the parameters nested within those functions you have to hit the appropriate Menu button a second time. When I first started using the machine, as soon as I reached the desired page, I instinctively started using the cursor and value buttons to make adjustments. However, by doing so, I always ended up moving to a different page. I did get better at it, but I don't think it's the quickest or most intuitive system.

I'm willing to completely forgive that small problem, however, because of the

helpful nature of the VZ1's graphic display. In the middle of its front panel is a large (64×96 dot), backlit, LCD display – similar to the one Casio introduced on their FZ samplers. By simply hitting the Display button when on the appropriate pages (envelopes, key follow curves, velocity curves, positional crossfades, etc) a graphic representation of the numbers that are usually present will be drawn on the screen. And yes, you can edit the envelopes while in this display mode and instantly see the results of your changes. You can also zoom in and out to one of several different viewing levels for fine tuning. I personally find this kind of system much easier to work with than a series of numbers – though even the VZ's numeric pages are nicely laid out and easy to understand.

Another helpful graphic on the VZ1 is available by pressing the momentary Line button. Whenever you're programming or editing a patch, you're going to want to constantly refer to the overall combinations of the various modules and by holding down this button the VZ will display the complete Line configuration. As soon as you release it, you're back to where you were.

Casio has also sped up the editing process by offering eight individual module select and module on/off buttons (they double as the patch select buttons), and comprehensive copying and initializing functions. If you're working with a function that can be independently set for each module, jumping from one module to another can be easily accomplished by hitting the appropriate button. Also, if you want to only listen to a few modules at a time when building or editing a patch, the on/off buttons are invaluable.

Last, but definitely not least, in case you do get confused, the VZ1 actually has a decent owner's manual with a complete function index and several helpful tutorials. I would've liked to have seen a bit more information on applications of the various functions, but the unique presentation of the index deserves high marks. To give you an idea, the DCO Envelope page in the index lists and explains all the parameters and their range of values, explains what Operating System buttons need to be pressed to reach it, points out that three software pages are dedicated to the function, that graphic editing is available, and that the function affects all eight modules globally, and finally, lists any related functions.

The Sound

THE FIRST THING you notice about the VZ1's sound is that it's *big*. With the possibility of having up to 32 sawtooth waves going at once in an Operation Memory I suppose that's not a big surprise, but nevertheless it's one of the instrument's most endearing traits. The multi-waveform modulation going on inside the VZ also allows it to produce some great nasty, biting sounds – somewhat similar to things I've heard from the TX81Z. And speaking of FM instruments, yes, the VZ can do some very good electric piano and bell patches, but the VZ1 sounds a bit warmer to my ears than other FM synths. It maintains some of the qualities I hear in my trusty CZ101. The VZ's noise waveforms also enable it to produce some fashionable, complex "airy" sounds. I don't think it's particulary well-suited for imitative patches, but that's not a problem because the synth sounds are great.

To get a good idea of the variety, there's no better place to look than the VZ's factory presets. Many normal patches work well on their own, but of course the really good stuff comes from the Operation Memories. The appropriately titled 'See God', as well as 'VZ EP', 'Ice Horns', and 'Headbanger', for example, should give you a good idea of the VZ's sound capabilities. What they might also demonstrate to you if you listen closely enough, unfortunately, is a small noise problem the VZ1 suffers from. It's nothing horrible mind you, but you can hear the noise floor kick in after you start playing softly and it will continue for about a half second after you release a key. I don't know why, but it seems that lately just about every synth I've reviewed has been slightly noisy – I hope it's not a trend . . .

The Conclusion

ON THE WHOLE, I really like the VZ1. It sounds good (in spite of the slight noise problem) and it's capable of producing a wide variety of different timbres. The iPD synthesis system used by Casio provides some similar ground to those who are familiar with FM, but gives you the flexibility to do even more. In addition, the small details of the voice structure and the numerous options for combinations make it a very musical instrument.

As for working with the VZ1, I can't say enough about the value of its display. I realize these kinds of displays are expensive, but I certainly wish other manufacturers would follow the lead that Casio and Kawai (on their K5) have set. Though they can't replace a computer and editing software with good graphics, they are a *huge* improvement over existing LCD displays.

Casio has a valuable new addition in the VZ1. If you don't need a synth with a built-in sequencer or effects, and don't mind spending some time learning another new synthesis system, check it out. It'll be well worth the time spent. ■

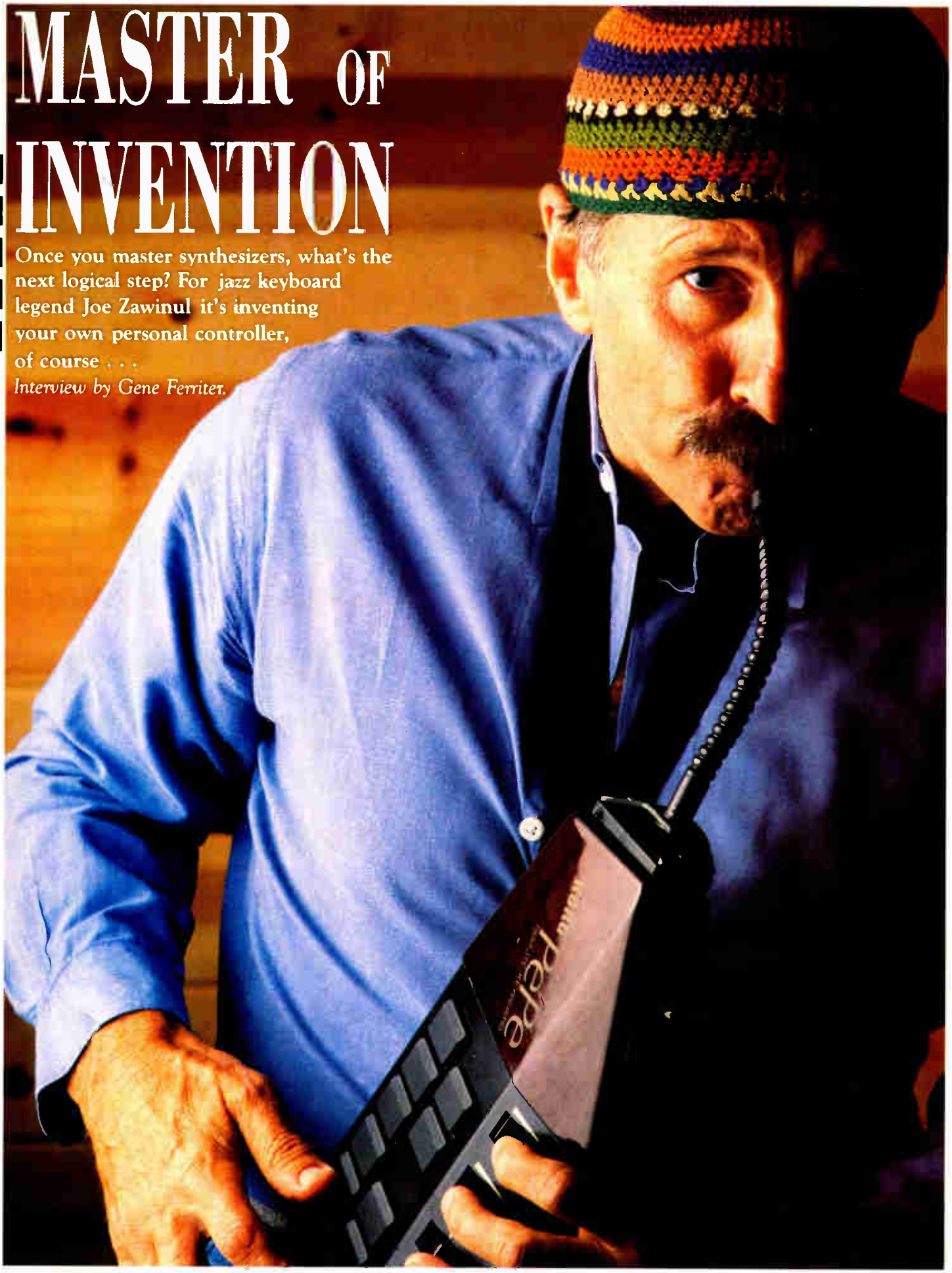
PRICES VZ1, $1499; VZ10M, $1199; RC110 ROM card, $89; RAM card, $59.

MORE FROM Casio, PMP Division, 570 Mt. Pleasant Ave., Dover, NJ 07801. Tel: (201) 361-5400.

MASTER OF INVENTION

Once you master synthesizers, what's the next logical step? For jazz keyboard legend Joe Zawinul it's inventing your own personal controller, of course . . .

Interview by Gene Ferriter.



Photography *Peter Figen*

ARRIVING AT JOE Zawinul's home in the hills of Malibu was as refreshing and cleansing as the spirit of his music, grabbing all of your attention and soothing your insides at the same time. Coupling the essence of good taste in artwork and spare interior design with picture windows that provide a panoramic scope of the Pacific Ocean, the afternoon was a memorable glance into the mind of one of the world's greatest living keyboardists/composers.

Zawinul has long been an innovator, not just as a composer and a player, but in the design, creation and utilization of new instrument technologies. His latest invention is a remarkable sounding device called the Pepe, not quite like anything you've ever seen. Pepe has a mouthpiece and resembles a horn in size but is held at an angle similar to an accordion. "I was an accordion player when I was young, and I always liked to play vertically. I like the vertical approach rather than the horizontal approach. I like the horizontal as well, but to play here," gesturing with the grip of a horn player, "and to play with a mouthpiece where I have control over my tone and everything, that's what I've always wanted."

As a MIDI controller, Pepe is not restricted to one patch or one sound, an idea which sprung from the accordion concept. "On a button accordion you get two notes on one draw and one pull. With Pepe (which is my first name), I have developed it such a way that I get up to six notes by fingering one. I'm not harmonizing the note, but I can get a different sound with a switch that I can control on the instrument. The switch is similar to the pickup settings on a guitar, but with six individual buttons rather than one and a much more diversified range of sound possibilities. If you listen to my solo on 'The Devil Never Sleeps,' all those funny gypsy type of things are a good example. I play it all on Pepe. The trumpet sounds, the saxophone sounds are all played on this instrument."

The nature of Pepe reminds me a bit of the EWI that Michael Brecker uses so effectively. "Yeah, he is playing the hell out of it. It's a much different kind of thing, though. There is only one Pepe in the world (and there will only be one). The fingerboard is set up like an accordion, but different. It is a combination. It is set up like a piano keyboard, but with buttons and with a multiple choice of from one to six notes, totally MIDIfiable to anything."

Another one of Zawinul's trademarks is his work with vocoders, which enable him to embellish his tunes with a robotic sounding voice module. "Well, I kind of helped to develop the vocoder, but I don't want to take credit for something that I did not do. I *invented* the Pepe. But I'm not a technician, so I made a drawing of it and put down all my requirements and then I got together with my keyboard technician, Ralph Skelton, and he put it into technical terms. That enabled the people at Korg to build it for me. Everything on the instrument is totally designed by me, but it took a while to develop."

Never one to rest on the laurels off his vast accomplishments, Zawinul shared some of his future visions. "I have another invention coming up. It's called the Korg Boomerang. It's gonna be shaped like a boomerang, kind of half-rounded. It's gonna have a mouthpiece and a fingerboard that curves around. It will be *bad*," he says, reciting one of his favorite colloquial phrases. "It's brand new, still in the development stage in my head. Korg doesn't even know about it yet. It's gonna be a real funny looking instrument and kind of small so you can just wail on the thing.

"I also want to work on a new vocoder. I want to get a little more out of it. I want it to be a little more flexible. I don't know how I'm gonna approach it, but I'm working on getting a combination between vocoder and a certain keyboard sound. Maybe I can introduce a pitch-bend, a trigger from another keyboard into the vocoder so if I hum one note it will translate another into the vocoder. I can control what it is doing only with my voice at this point, but if we find a way so that the aftertouch only comes into the vocoder and triggers it, then I'll have something else going on. I might be able to even use my six-note system at that point, holding one note on the vocoder. I also have a microphone built into Pepe so I can play the vocoder through Pepe. There are a couple of other things I want to do, but it will take time to develop."

THE ZAWINUL SYNDICATE, formed around a year ago, is a six-piece ensemble that recently released their first album, *The Immigrants* (Columbia Records). The album has received both critical and commercial success, appearing high on national jazz sales charts. "We haven't even started to scratch the surface of the talent of this band," he boasts with exuberance. "Before our first show on this tour we only had four days rehearsal. But what we have already accomplished in sixteen or seventeen shows is phenomenal. We kill people everywhere we go and for me that is the criterion. We played four shows in Fort Lauderdale that were unbelievable.

"At The Palace (in Hollywood), it was mentioned that even (well-known jazz critic) Leonard Feather was dancing around and having fun, and that's the main reason I want people to come to hear my music. I'm not into history-making and all that serious bullshit. I want people to get out there and have a ball, get off their ass and forget about their daily troubles. I want 'em to dance out there and do whatever the hell they want. Throw some stuff. Have some fun. Go home and say, 'Today we spent our money on something really hip.' And if they want to think, if they want to get into the analytical complexity of things, we offer that as well."

"Any idiot in the world can master an instrument, believe me. Then, you must learn music."

Zawinul has a deep history in the jazz world, having worked with such greats as Miles Davis and Joe Williams among many others. His decade of collaboration with Cannonball Adderly, which included nearly twenty albums, and his fourteen landmark records with Weather Report (which he co-founded with Wayne Shorter), have established Zawinul as one of the primary forces in jazz. The inception of The Zawinul Syndicate included drummer Cornell Rochester and guitarist Scott Henderson. When *The Immigrants* was recorded, the rest of the band had not been fully formed, but has since congealed by adding Babatunde Uhuru for the percussion responsibilities, Gerald Veazley on bass, and singer/percussionist Lynn Fiddlemont.

One of the interesting elements of Weather Report was the absence of a guitar player during an era of visible fret wizardry. Scott Henderson is an incredible musician and seems to hold the spotlight as much as Zawinul himself with The Syndicate. "I'll tell you, man, I'm an expert because I've played with a lot of musicians. Scott kicks all of their (guitarists') asses cause he's not automatic. He doesn't play patterns. He plays beautiful chords. He's playing the blues, man. I'm a blues guy myself. I love the blues. Everything we're playing is kind of related to the blues in abstract forms."

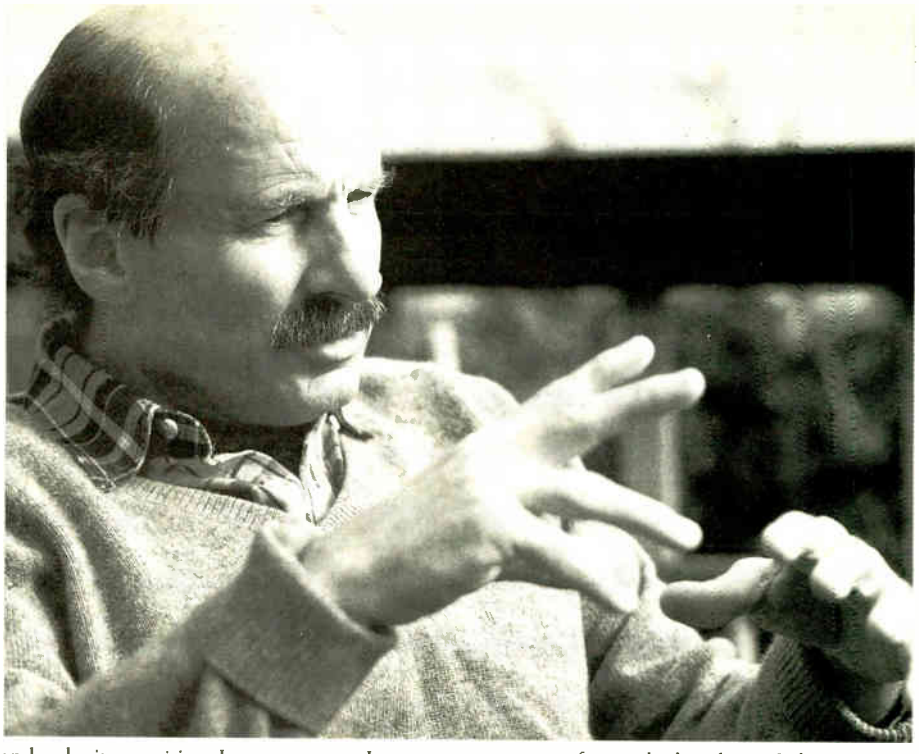
Continuing with conviction, "No one in my band just plays patterns. You play what comes into your head and respond to the other guys around you. That way it's fresh. It's always fresh. When we play five sets they are all different. Most guys just play their licks." Zawinul's verbal phraseology is often as enjoyable and communicative as his exotic keyboard lines. "When it comes to feeling for the blues, Scott is a black dude, man.

"The whole band's gonna be better now that we'll be playing all the time. We're going to Europe for 30 concerts in 32 days. I'm gonna take this band beyond all of my other bands because every member fills the voids with the happiness of playing good music and the fun of creating every day. We go everywhere together. We go eat together, we went to the Tyson fight ▶

►together, we laugh and have fun. That's all this band wants to do. We all love sports. It's great. We used to have that with Weather Report when Jaco first joined the band.

"Weather Report had an image of being such a great band," he reflects, "and it was, in its own right. It was a band with great human beings and I loved it, but it could have been ten times greater. We haven't reached the point with The Syndicate yet that we reached with Weather Report, but the potential is much greater. I can already see it after one month on the road because it took a long time to get that cohesive feeling with other bands that we've already achieved. It's gonna be awesome in about a year from now."

Aside from the sophisticated level of compositions and the calibre of musicianship, sound quality has always set Zawinul apart from mortal musicians. Still riding the crest of the new equipment revolution, Joe has witnessed several generations of change. His current stage setup is an interesting amalgam of the eras of which he's been an integral part. "Right now I'm using the Sequential Circuits Prophet T8. On the top I have a new Korg 707. It's a small keyboard that does little wonders. It's very easily programmed and it has many possibilities. It's digital and I can get many variations on one note by pressuring correctly. It's very touch sensitive and velocity sensitive. I can create scales playing just one note. On one of my tunes I create the whole overtone series playing one F note. The overtones return very nicely and then sometimes I play one other note to blend the overtone series of the two. It creates a very ethnic sound, but I'm not sure from which culture. It has a nice little twang to it.



"Then on my left side I have a DW8000, also by Korg. Underneath I have the Oberheim Xpander which I use for a lot of things when I play with Pepe. That's how I am able to get different notes by fingering

just one. Then I have a DSM1 rackmount module which is an extension of the Korg DSS1. It's fantastic, man. That's how I get that tenor sound. And on the record, the bassline you hear on 'Criollo' is me playing the DSM1. It sounds great. Korg makes great things.

"On the other side, I have a Chroma Polaris, which is an old analog keyboard. A lot of my stuff is analog. Underneath I have an old Chroma Expander, which is also analog."

HAVING GROWN UP in Austria, Zawinul was a boyhood prodigy on the accordion and was sent to Czechoslovakia to study piano at a conservatory during the war. After becoming an accomplished classical musician, he was introduced to jazz at age thirteen. Finding his way into several excellent jazz combos, it was not long after that Joe had his first experience on an electric keyboard.

"The first one ever was a B3 (Hammond) that I played in an American club in 1948. It was in a hotel in Austria, and they had this little American chapel that allowed me to go down every day and practice. I loved it, man. I loved the B3 and I still do.

"Then I played with Dinah Washington. I knew Ray Charles and he was the first one to play the Wurlitzer on 'What I Say' and all those tunes. We had such bad pianos down south that Dinah asked Charles if I could play the Wurlitzer and he let me use it.

"Then later on, when I was with Cannonball Adderley in 1966, we recorded 'Mercy, Mercy' at the Capitol studios here on Vine Street. They had a Wurlitzer as well, and they let me use it, and that's how this whole thing started. After getting my own Wurlitzer I got a Fender Rhodes. One of the first ones.

"Then I met Mike Knops from New Zealand, who had a band they called The Fourth Way. He gave me a ring modulator. So I fooled around with the electric piano with the ring modulator and did a lot of good things.

"I was living in New York and there was a guy living near me who played violin for The New York Philharmonic whose wife taught my children how to play the piano, so I spent time at his house. Together we bought the first synthesizer. It was called Putney. It was nothing but pads. The Putney was a helluva trip, man, all you could get was one oscillator. But at least I started to know what an oscillator was and what it did. Then I went to Boston with Weather Report for the first time, and Todd Rundgren's old keyboard player turned me on to the ARP 2600s. That was a big break. Then things just developed from there. I'd get this keyboard and that keyboard and then I got the Sequential Circuits Prophet 5. I played the Rhodes Quadra for a long time and I still have two of my ARPs and a lot of my old equipment."

Between his absorption with sound technology, composition, recording and touring, Zawinul doesn't choose to teach, but communicates with clarity his insights into developing musicality. "I would have one counsel. Listen to everybody. Copy as much as you can. Then throw all the stuff away and just let it go. First learn your instrument. Master your instrument. Then you learn music. And then let it go."

Somehow, the process doesn't seem quite that simple. "Any idiot in the world can master an instrument, believe me. Then, you must learn music. And while you're doing this you listen to everybody. You listen to all kinds of music and let the stuff soak into your head. Try to copy as much as you can as a youngster, but then let it go and do your own thing. This whole process is no guarantee that you will become a great musician, or even an individual musician or a great composer or an individual composer. But it'll be better than just hanging on one idol and emulating and emulating and maybe becoming a good copy or something. Get as much experience with different art forms within the music spectrum as you can. There's so much great music everywhere in the world. It's not just jazz. Indian music, Hungarian music, Greek music, Russian music, Italian music, classical music. Rock 'n' roll, even. Listen to everything, digest, then spit the stuff out and do your own thing. Something will come out of it." ■

ON STAGE

Dixie Dregs Reunion Tour

NAMM, ATLANTA, GA

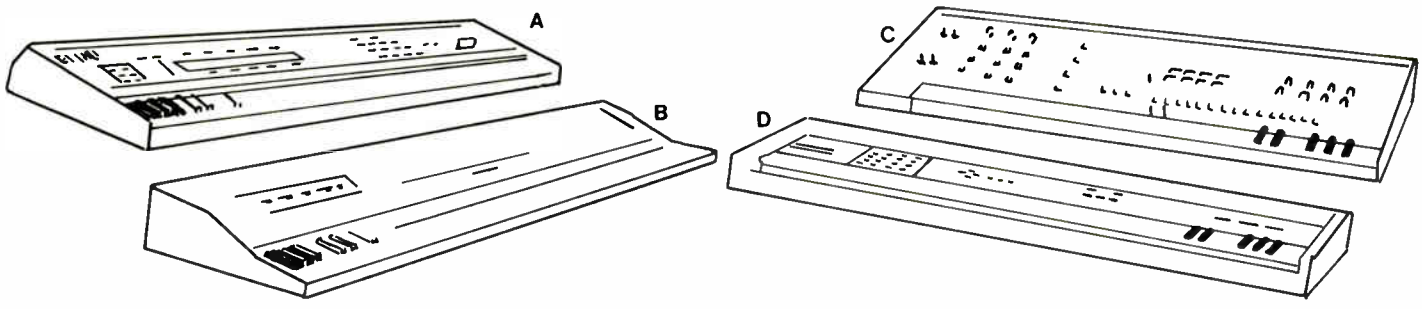
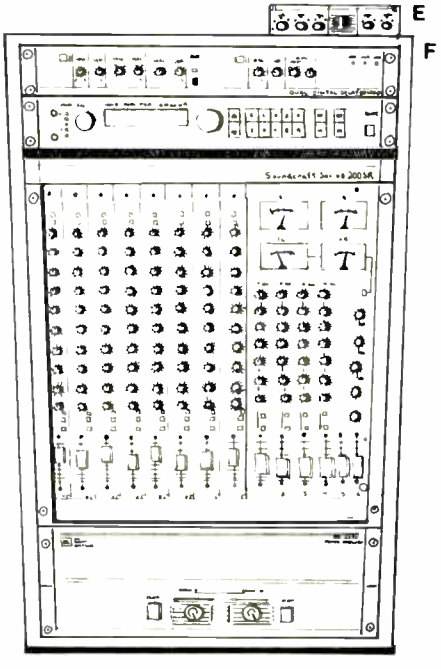


Illustration *Brent Heintz*



T Lavitz

- **A** Ensoniq SQ80
- **B** Ensoniq Digital Piano
- **C** Oberheim OBX
- **D** Ensoniq EPS
- **E** Lexicon LXPI
- **F** **Rack includes:**
  - Ibanez DDI000
  - Lexicon PCM70
  - Soundcraft 200SR mixer
  - Dynacord CLS222

A one-shot promo CD turns into a reunion tour for this critically acclaimed band.

Text by Chris Meyer.

FATE IS PRETTY strange. T Lavitz, well-known jazz fusion keyboardist, decided to add some of Ensoniq's latest gear to his rig. John Senior, engineer at Ensoniq and friend of T, mentions around work what a great product endorsement that might be. Things grow, and all of a sudden, there's a CD single with the smiling faces of the Dregs on the cover. Long-slumbering fans awaken, and there's demand to see the Dregs live again. Suddenly, just another reason for an ad blossoms into a three-week tour by a favorite (and long dormant) musicians' band.

This reunion saw a modified version of the last Dregs band – Steve Morse on guitar, Rod Morgenstein on drums, T Lavitz on keys, and Dave LaRue, who had played with T and Rod, replacing Andy West on bass (who had just taken a gig as head of WaveFrame's West Coast artist relations).

T has slimmed down his rig a bit, using just four keyboards – an Ensoniq piano, EPS, and SQ80, along with a trusty Oberheim OBX – along with a rack of processors. The SQ80, used for marimba and organ sounds, is layered via MIDI with the digital piano, used for piano and clavinet sounds. T related that he uses the piano as the master because he prefers a weighted keyboard, and this duo allows him to recreate lines live that were layered in the studio without having to stretch across two keyboards (only once during the gig does he play just the SQ80 by itself). The EPS (for flutes, rock piano, etc) is MIDI'd to the OBX, but they tend to be on different channels and played independently. The EPS is loved for being able to have several instruments on call at once (including layering sounds like the flute and rock piano, to make a line sound like a tight unison); the OBX provides analog contrast to all the digital sounds.

The Soundcraft mixer was chosen for its generous four sets of effects returns. Those effects include a PCM70 (which "rounds everything out"), an Ibanez dual digital delay (with footswitch control to select between two different delay lengths live), a Dynacord Leslie simulator ("the best thing I've heard, next to a Leslie – and I'm an organ player"), and a Lexicon LXPI (under footpedal

control), used for a long delay. T intends to get more into the LXPI, and remotely control its parameters over MIDI from his keyboards. He monitors the house signal with all the effects, and has one speaker over by Morse so they can hear each other (at the opposite ends of the stage, they sometimes trade or do locking solos).

Both bassist LaRue and guitarist Morse have extensive racks, and both either have or are exploring MIDI foot controllers and mixers to switch effects (Yamaha DI500 DDL and SPX90 for LaRue; an Eventide UltraHarmonizer, DigiTech DSP128, and Peavey line switcher/mixer for Morse). Morse, whose rack has been changing "almost daily," also has the returns from three different delays (Lexicon PCM41, PCM42, and Prime Time) all under footpedals for live mixing. He hopes to eventually have two sets of footswitches – one for amp sound, one for effects sound – so he can pick combinations live, without having to memorize patch numbers or switch connections live.

The NAMM gig, the first leg of a 15-date tour, was held in a ballroom with boomy acoustics. To my surprise, Steve Morse actually apologized for the acoustics of the hall not being able to accommodate the speed of the one number he played with his own Steve Morse Band ('Cruise Missile'). The rest of the night cooked, with the Dregs sounding more powerful (and more "out," at times) than I ever remember them on record. Morse is undoubtedly one of the most tasteful rock guitarists around. Rod was fast and strong while keeping the feel; Dave filled in more than adequately for Andy West (though, really, Andy *is* a unique creature); and T seemed more subdued than when he's fronting a band, which was to his advantage. Spending a good deal of his time on the Oberheim, he continues to be an excellent foil for Morse.

Photography *Rose Rounseville*

They don't rule out another reunion in the future; until then, some of us have got memories . . .

Special thanks to Ken Hirsch for his invaluable assistance.

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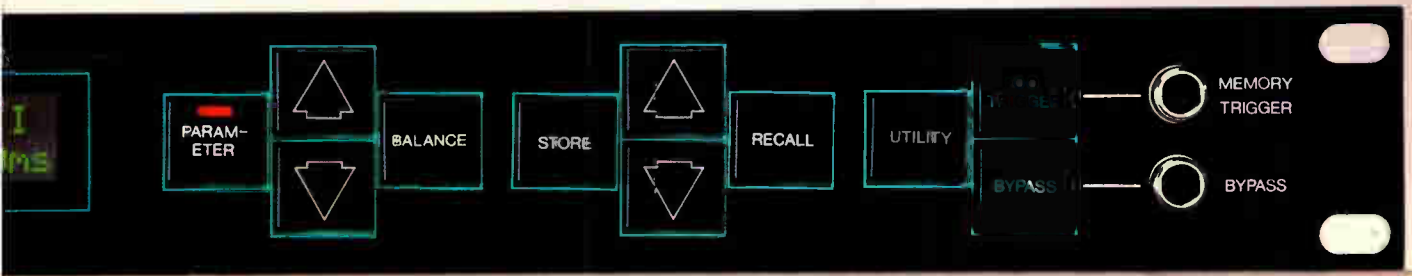

DIGITAL AUDIO 102 *practice*

TECHNOLOGY

Illustration Christine Mallouf

Understanding the theory of digital audio isn't easy, but it's possible. The second part of our series gets a bit more theoretical and a lot more practical. *Text by Peter Bergren.*

I CLOSED LAST month's article by saying that there's an entirely new world opening for those of us interested in music production – the advent of the "affordable" all-digital studio. Prices are still, well, "pricey," but remember when the price barrier for digital reverbs was a thousand bucks? Or when CD players were for the audiophile only? Just wait – the need by general purpose computer and audio consumers for faster processing, more memory, and LSI chips with dedicated functions will surely benefit the recording enthusiast.

This month I'll cover some more of the most essential theory, talk about digital signal interconnection, and attempt to describe some of the new products about to pour into this country.

Moving Numbers

THE PRODUCTS I alluded to above handle all sound storage, transmission and processing digitally, with no analog signals present save at the beginning and end of the signal chain. Such functions as recording, sending/receiving, equalizing, filtering, attenuation, gain, reverberation, delay and the like are accomplished by mathematical manipulation of a stream of clocked, binary numbers.

As we saw last month, these numbers represent discrete measurements or "samples" of analog signal voltages at a particular "frozen" moment in time. Life itself tends to be a continuous, analog function. However, even though many separate measurements (even 48,000/sec) can never completely represent an infinitely changing energy level, it is possible to encode and decode enough information to preserve the fundamental integrity of any analog waveform. After all, a motion picture image consists of many still pictures ("samples") and information is surely lost between picture sampling periods as the subject matter moves. Nevertheless, the illusion of motion is well preserved.

In a like manner, 16-bit digital audio systems can accurately present many different amplitude levels and very fine divisions of time. Taken together, amplitude/time relationships recorded as a string of on/off pulses can do a fine job of both transmitting and recording sound. That's the consensus, at least, among enough people to make this method economically viable.

Binary Background

THE WORD "BINARY" means "consisting of two." In the case of binary numbers, the two quantities involve the numbers One and Zero. As Gottfried Wilhelm von Leibnitz postulated in 1679, it is possible to represent any decimal number with a binary equivalent. The comparison below should clarify how this is possible:

Decimal Number	Equivalent Binary Number	Built From:
0	0000	
1	0001	
2	0010	
3	0011	1+2 (0001+0010)
4	0100	
5	0101	1+4 (0001+0100)
6	0110	2+4 (0010+0100)
7	0111	1+6 (0001+0110)
8	1000	
9	1001	1+8 (0001+1000)
10	1010	2+8 (0010+1000)
11	1011	3+8 (0011+1000)
12	1100	4+8 (0100+1000)
13	1101	1+12 (0001+1100)
14	1110	2+12 (0010+1100)
15	1111	3+12 (0011+1100)
16	1 0000	
Etc	Etc	Etc

Notice that each move to the left from the right-most binary column doubles the value of the next column. Note also that this positional notation scheme yields power increases equalling 1, 2, 4, 8, 16, etc, when expressed in decimal form. To denote intermediate numbers, the binary expressions of components of that number are added together. For example, 3 (0011) = 1 (0001) + 2 (0010).

The neat thing about this scheme is that it's an ideal way to pump numbers through an electronic machine. Ones are On voltages; Zeros are the absence of a voltage, with the difference between each state being designated by a time interval assigned to each pulse or "bit."

Bits can thus be clocked through a digital system, with each one holding fort on the cable for a particular "tick" of the bit clock. As the clock ticks along, each bit (in order) asserts its voltage on the cable. As long as the relationship between which bit belongs to which increment of time is maintained, a serial stream of pulses representing numbers, letters of the alphabet, pixels in an image, etc, can be transmitted and kept track of.

In digital audio, 16 bits (for high quality definition of voltage levels) are assembled by an A-to-D (analog-to-digital) converter into words, or "frames," the boundary of each being defined by pulses of a "word clock." Each word denotes the snapshot measurement of the voltage sampled from an analog signal. The words are propelled by the word clock through the digital hardware, where they can be simply retransmitted, be recorded (after certain types of modulation processes), or be processed in various ways.

One drawback of this digital approach is the frequency of information that must be handled. Just multiply 16 bits times the professional sampling frequency (number of words) of 48kHz, and you'll see what I mean (we're talking 728,000 bits per second, not counting any bookkeeping that might have to go along with the signal). There's also the matter of keeping straight which bit is the most important value-wise in each work (known as the "MSB," or Most Significant Bit), or conversely, the smallest value (the "LSB," or Least Significant Bit). Above that level, the word may be broken down into 8-bit "bytes," and which of the two comes first must be defined. Pinning these down is obviously of extreme importance when trying to get one digital device to understand the "language" of another. Likewise, the clocking rate of the word and bit clock pulses must be kept synchronized between devices, or chaos is the likely result.

Analog-to-Digital Conversion

LAST MONTH I got as far as covering how Sample and Hold (SH) circuits work. You'll recall these are located at the head end of the system, after a low-pass filter. These circuits measure (sample) the voltage present in the incoming analog signal at the start of each sample period. Once this voltage is sampled, the majority of the rest of each sample period (controlled by the word clock) is devoted to holding this sampled value. Otherwise the sampled voltage would vary along with the continuing excursion of the analog signal. Assuming this Sample/Hold function is performed correctly, the result is one "tread" in a stairstep pattern of voltage increments, which roughly follow the contour of the analog waveform at the system input. This stairstep voltage is a hybrid analog signal, which still must be turned into a binary-coded representation of each voltage value of "stair step." How is this done?

One method in wide use is the SAR (or "Successive Approximation Register") design. It consists of an A-to-D converter, a D-to-A converter, and an analog comparison (comparator) circuit in a feedback loop. Each sample from the SH circuit is converted into a binary equivalent, which is then converted back again into an analog value. This voltage is compared to that from the SH circuit (which is the value ultimately to be described in binary code), and if the twice-converted voltage exceeds that of the SH voltage, another word is generated, which goes through the same process of reconversion to analog and is recompared.

This process occurs bit by bit, MSB first. Each trial, or "approximation," begins with insertion of a One in place of a Zero in the interim word. If the analog value of the resulting word is less than that of the voltage from the SH circuit, the One is left in place, and the system tries again. Thus, this design gradually narrows down a finer and finer approximation, in binary form, of the SH value. This all must occur within the period allowed for one 16-bit word, or at the professional sample rate, 1/48,000th of a second! Seems incredible, doesn't it? ▶

► However, this design is widely accepted, even though 16 A-to-D and D-to-A conversions must occur to create the finished, "successively approximated" 16-bit word.

Another popular approach to A-to-D conversion is called the "Dual Slope Integrating A-to-D Converter." In this scheme, a reference voltage is integrated (allowed to drop in level, defining a ramp-like function) through an RC (resistor/capacitor) system with a known "droop" curve until it equals the value of the sample coming from the SH circuit. The amount of time this takes is counted out by a digital counter, and a binary word representing this time measurement is output. This is the representation of the voltage value in the SH circuit. Since the clock rate to accomplish this with just one decay/counting circuit would be prohibitively high, the task is divided between two such circuits: first, one counts the time it takes for a large reference voltage to drop to that near the SH value; the other counts the decay time of a much smaller reference voltage (1/256th of the larger reference value) to finish the measurement off.

The "coarse ramp" counter generates an eight-bit word denoting overall sample value, and the "fine ramp" counter generates another eight-bit word describing the smaller details of the SH sample value. Combined together, these two words create one 16-bit word with both MSB and LSB bits in place. This method allows a much lower clock rate for the counters than would be the case if just one ramp/counter circuit were used.

Obviously the methods one must resort to to change the SH circuit "stairstep" values to accurate binary code are far more complicated than simply handling an audio signal through strictly analog means. It's a wonder that A-to-D (and its mirror process, D-to-A conversion) works at all, let alone well. Nevertheless, it does work; and quite well at that. The trick is to keep from having to do it too often, partly because of the expense of such circuitry. Perhaps more importantly, though, every time you convert from A-to-D or D-to-A, the chances of error increase. Variances and noise in these analog circuits cause the errors.

Another seldom-discussed error in A/D and D/A circuits is uncertainty in the last bit. That last bit can only be either a One or a Zero; in reality, the analog signal was (or should be) somewhere in between there. Therefore, there's *always* a half-bit error (or less) built into any conversion.

Remember: digital audio is a process of approximation of values. Obviously, the longer you can stay in the digital realm, the greater your chances of avoiding error. This is the prime reason that all-digital equipment, with digital ins and outs, is such an important advance.

Space doesn't permit me to describe other elements of a digital encoding and decoding system. That will have to wait for another article. But I hope this month's (and last month's) examination of what I feel are the key elements in such a system will start to acquaint you with the groundwork of digital thinking.

"Digital audio is a process of approximation of values. Obviously, the longer you can stay in the digital realm, the greater your chances of avoiding error."

You've had enough theory, anyway. Time to talk about the real-world applications.

The AES/EBU Standard

WHEN YOU CONSIDER that digital audio is nothing but Ones and Zeros, you should get scared when you think of how different devices try to interpret exactly what each other are saying. A few things have to be agreed upon by the two devices. For example, the word and bit clock rates must match. The number of bits per word must agree. The spacing of quantizing intervals must coincide (linear 16-bit – the current standard – assumes equal voltage intervals between each of the 65,536 quantizing increments). The position of MSBs and LSBs must coincide. And the ways of distinguishing where one bit or word ends and another begins must be in common. Obviously, some kind of standard for interconnection is badly needed. And, fortunately, there is one.

In the early 1980's, committees were formed in common between the Audio Engineering Society and the European Broadcast Union to tackle the digital standards problem. By 1983 they'd arrived at a consensus. The published standard applies to one or two channel digital transmission using balanced audio cable terminated in a 3-pin XLR connector (or 9-pin D connector). This type of connector is readily available – you could even use a microphone cable! The purpose of the standard is primarily to transmit and allow reception of a digitized, stereo audio signal, and not to serve as a channel for other kinds of signals (though some allowance has been make for this, as you shall see). But the prime idea is to allow all-digital units to be chained together.

"The purpose of the AES/EBU standard is primarily to transmit and allow reception of a digitized, stereo audio signal, and not to serve as a channel for other kinds of signals."

Sampling rates of 32, 44.1, and 48kHz can be accommodated by the standard. Per the standard, each "frame" (or "word") of audio data comprises 64 bits which are subdivided into 32 bits for left-channel information, and 32 for right-channel information. Of these 32, 20 on each channel are dedicated to audio data (in other words, to the "word" describing the quantized analog snapshot captured in that sample period). Of the remaining 12 bits per channel, four are synchronizing bits, whose polarity differentiates between left and right channels. There are four "user bits" per channel, one of which is defined but not in common use. Called the Audio Sample Validity bit, it carries either a One or Zero depending on whether or not the audio data associated with it has been error-corrected or not. Apparently there's some debate about whether the "flag" needs to be raised at all.

Of perhaps greatest interest are the last four bits, which are dedicated to "auxiliary audio or other data," which could be MIDI or other information. A pleasant surprise here is that such data will always be in absolute sync with the audio data it is "carried" with. And while four bits per channel may not seem like much, remember that you've got eight bits per 64-bit word (left and right data). Take several sample periods into account (they do go by quite fast), and the possible number of bits add up to quite a bit (no pun intended) of information that can be transferred. Now, before you go running off with visions of MIDI 2.0 in your head, know that there's no standard way to use these bits, and the processing overhead required to handle AES/EBU *is* quite extensive to consider using it for something like MIDI alone. The point is, there's some additional room there.

One criticism here (that I heard from one of my interviewees) is that no provision has been made for some kind of common synchronizing pulse. That would be needed so that the bit rate of each device connected via the standard is synchronized. If it isn't, small data losses can occur. (Imagine someone talking just barely faster than you could write – occasional words would get dropped to keep up.) The losses would be within the range of error correction available in most systems, but (this particular person felt) they could begin to have an effect on the quality of the sound transmitted. In a way, this problem is analogous to the need for "house sync" pulses in a video system, which denote the start of each picture frame and other salient data. However, most digital systems common to one manufacturer (the Yamaha DMP7 Digital

Mixer, for example) make provision for agreement on one device's clock being the master for all elements within the system. No doubt the "common time" question, if it becomes a problem, will be dealt with.

Connections

A MULTICHANNEL STANDARD is also in the works. It will consist of 56 channels, each configured to the AES/EBU specs already mentioned, and be designed to be handled via a coax cable, or a fiber optic transmission line (this due to the very high frequencies that data will be transmitted at). A fiber optic option was considered for the one or two channel AES/EBU standard, but in the early 1980's a standard, inexpensive fiber optic connector wasn't available, and the 1.5 megabit/sec data rate necessary for stereo transmission could easily be carried on standard balanced audio cable. However, if you buy any of Roland's new audiophile digital equipment, you'll see a fiber optic port (very popular in Japan) which Roland tells me conforms to the present AES/EBU standard.

A final warning about connections: there are other digital signal standards in existence! One of the earliest was introduced by Sony. Called SDIF/2, it's utilized by that company's model 1610, and other digital recorder/reproducers and related gear. Then there's what's used on the aforementioned Japanese stereo equipment. Called variously EIAJ (after the Japanese electrical standards commission), SP (Sony/Phillips), or CD-RDAT (for the fact that it appears on CD and R-DAT players), it's very close to AES/EBU, but with some flags different. Virtually every R-DAT player has EIAJ connectors (either in the form of RCA jacks or fiber optic connectors). There's some confusion and concern among manufacturers in this country over whether EIAJ and AES/EBU are compatible, and whether or not both need to be supported on digital equipment, with the anticipated popularity of R-DAT's. More on this as we learn about it . . .

If you're just starting to consider a collection of all-digital gear, on your way to a true all-digital studio, you probably won't have any problems with interconnection, as long as your new gear is either "true AES" or "EIAJ," and the "common time" problem I mentioned doesn't become a thorn in your side. But if you want to ensure the compatibility of equipment with digital ports already in your collection, you might consider a digital converter, such as the Yamaha FMCI Digital Audio Format Converter. Yamaha claims this unit allows its proprietary code, used in the DMP7 digital mixer, to be converted into Sony SDIF2, EIAJ or true AES/EBU formats. And another format converter announced last January is capable of doing dual functions: it will convert Yamaha's proprietary format into EIAJ or SDIF2 formats and then convert that signal back again into the Yamaha format.

Sound a little confusing? Well, not to worry. Most of the new products coming out in the near future from Yamaha will accept any of the most popular digital formats. And a new line from Roland is already configured to AES/EBU standard, albeit via fiber optic ports. Here's some of the gear that's becoming available:

- **Yamaha DA202 D/A Converter.** Converts "true" AES/EBU or CD/DAT formats into analog.
- **Yamaha AD808 A/D Converter.** Converts up to eight analog channels to the proprietary Yamaha digital format used in its digital mixers and other gear.
- **Yamaha DMP7 and DMP7D Digital Mixing Processors.** The well-known 8/2 all-digital boards with moving faders, three built-in effects processors, and MIDI control. The "D" version has digital I/O built in. Both can chain to one or two other DMP7's for a multichannel board. Yamaha also offers three new format converters (the IFU1, IFU2 and IFU3) and a level converter (the IFU4) to use DMP7Ds in conjunction with multitrack digital signals, including Sony's DASH and Mitsubishi's ProDigi formats. (For more specific info, ask Yamaha for a copy of their very informative DMP7D brochure.)
- **Yamaha DMPII Digital Mixing Processor.** An affordable 8/2 digital line level mixer with digital effects. All parameters, including fader positions, are programmable. MIDI capable.
- **Yamaha DEQ7 Digital Equalizer.** A programmable two-channel equalizer with different EQ curve functions. MIDI controllable. Yamaha proprietary digital format in and out.
- **Roland E660 Digital Parametric Equalizer.** Two channels of four-band parametric EQ capable of being used as a single channel eight-band unit as well. Ninety-nine memory locations. AES/EBU standard digital inputs and outputs via coax or optical ports. MIDI capable.
- **Roland R880 Digital Reverb.** Various advanced reverb programs, two channels in, four out, MIDI capabilities. Digital input and output via coax or optical ports. AES/EBU standard.

So, there you have it. This is all just the tip of the iceberg. An all-digital studio may be in your future, maybe even a separate component workstation-type system, if MIDI and AES/EBU data can be coordinated tightly-enough. It should be an interesting next couple of years. ■

"The neat thing about the binary number scheme is that it's an ideal way to pump numbers through an electronic machine."

perspectives

SAMPLING LEGITIMIZED

An argument for why it should be okay to sample others' works. *Opinions by Ron Peterson.*

I FIND IT hard to believe that some people maintain that sampling individual sounds off of albums or CDs is illegal. One wonders how vain modern musicians have become.

First, let's tackle it from a legal angle. I'm not a lawyer, nor have I bothered to retain one for the sake of protecting my vanity, but I know enough about how my *music* is protected to make some extrapolations. The copyrightable element of a song is the melody. Not the bassline; not the drum pattern; certainly not a single sound. One case pending in the English courts centers on whether the bassline can be considered the melody in lieu of a normal melody instrument (M/A/R/R/S versus Waterman). If M/A/R/R/S wins, then this could be taken to an extreme: that a single sound counts as the melody in lieu of any other sounds in the piece that might therefore comprise a "melody." Pretty limited case. The only case I know of where a musician extracted some money from a sampler is when Herbie Hancock stole every other note sung by Jon Anderson in a Yes tune – but in that case, enough of the melody was still intact to make out the song.

Since a single sound cannot be protected as music, the other attempt is to cover it under the same laws as newscasts. You can reproduce your favorite Reagan quote off the 11 o'clock news, as long as you keep it under three seconds. Extrapolating this law to cover any three-second burst of sound, it seems any shorter sample is in the clear.

The only law I can think of that makes sense is the same performance law that BMI, ASCAP, et al use: you cannot re-perform a piece of music in whole or part for profit without paying royalties. Radio stations, dance clubs, and any place where live music is performed are covered, because they pay flat fees to these organizations (that's how modern DJs get away with murder). I think those people who sample off of records or CDs would be more than happy to pay a proportional royalty (based on percentage of running time of a sound sampled from a song) if BMI or ASCAP were set up to collect it. But that still wouldn't satisfy our vain friends up above.

In short, musicians, producers, and recording engineers no longer want to get payed for creating *music* – they want to get payed for just coming up with an interesting *sound*. I think that's really copping out on what music is supposed to be about. Musicians and songwriters have told us for ages that a piece of music should be judged on the song, first, followed by the performance, followed by the sound. I would like to put forth that people who want to sue over having their sounds stolen are admitting that both the song and the performance aren't special enough to make the piece stand up, and therefore it's relying on individual sounds to carry it.

> *"If one wanted money for creating a special sound, that money should go back to the one who made the instruments and effects processors in the first place."*

Enough for legal issues and inflated egos – let's look at some practical standpoints. One, for a sound to be stolen, it has to have already been released, which makes it old news. Anyone who relies on stolen sounds is already behind the cutting edge. If the original artist (or producer, or recording engineer) is actually pushing their art forward instead of resting on old laurels, they're already working on something else.

Two, most people don't recognize stolen sounds. If they didn't recognize the sound, either they've never heard the source (so how can the creator of the source be damaged?), or it wasn't special enough to demand notice. If the listener did recognize where the sound came from, they'll tend to tip their hat back at that source – probably even listen to the source again (remember, imitation *is* the sincerest form of flattery). Beyond tracing the source, the next thought is inevitably a musical judgement – who *used* it better? Which brings us back to performance, a level above the sound itself.

Third, unless the sound was created by a one-of-a-kind instrument owned solely by the maker of the sound, in theory, anybody could recreate it in the studio and use it. If there's no difference to the ears between a sampled Phil Collins snare and one recreated in the studio with a snare, a gate, and a reverb chamber, the fact that one was stolen becomes solely a moral as opposed to a substantive argument. If one wanted money for creating a special sound, that money should go back to the one who made the instruments and effects processors in the first place.

Let's keep in mind that we're talking about a single sound here, not a whole arrangement or musical phrase – I do indeed see something wrong with stealing a Rembrandt and calling it your own; I can't see anything wrong in using one of his *colors* . . .

Music is fun. Sounds are fun. Sounds make music. Some people have pieced together some interesting (or at least, very catchy) music based on others' sounds. But let's not replace concentrating on music in favor of worrying about where the sounds came from, okay? ■

Ron Peterson is an insuronce salesman by day and a gigging musician and songwriter by night who lives in Indianapolis, Indiana.

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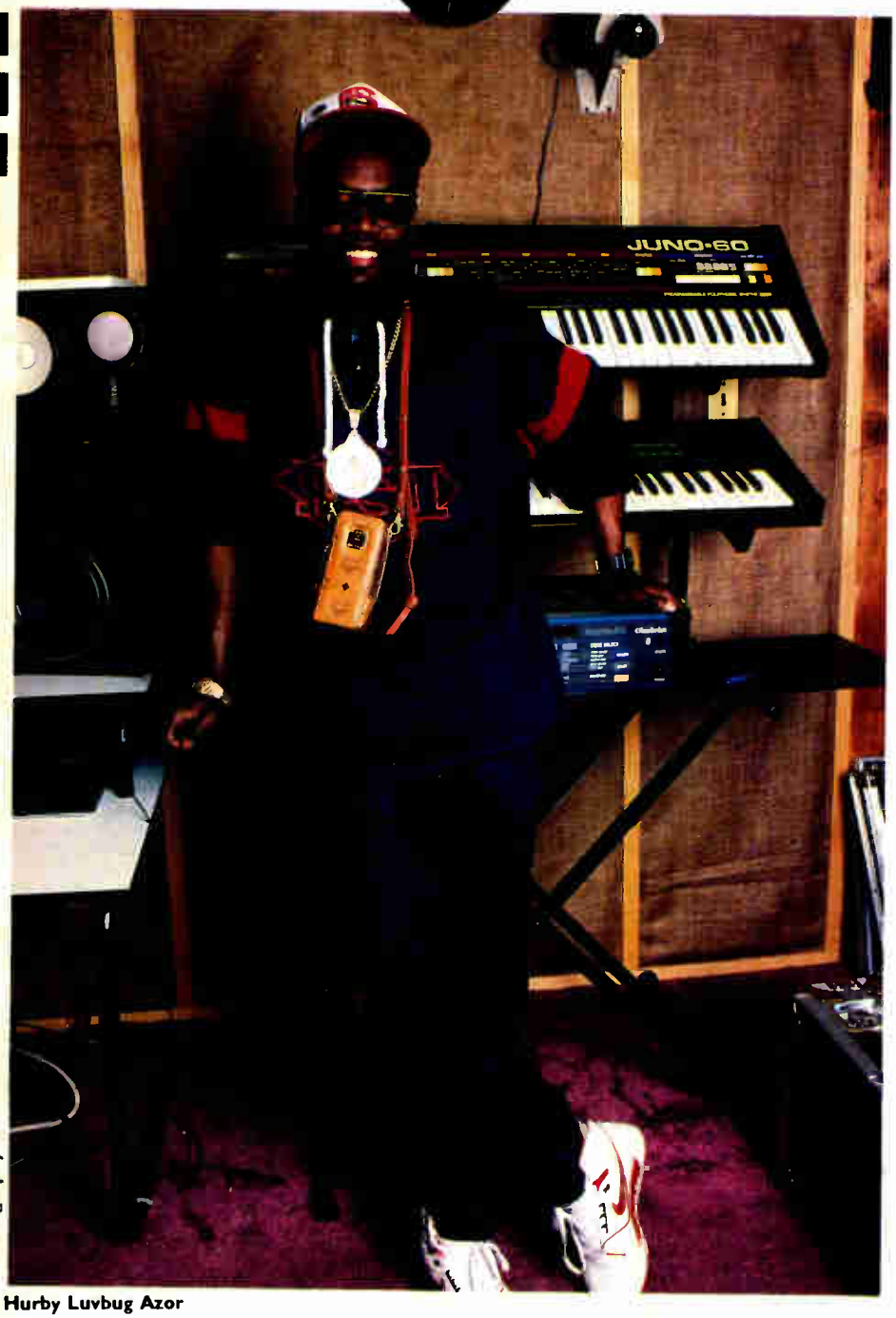


Photography Eddie Malluk

Hurby Luvbug Azor

Marley Marl, Hurby Azor, Chris Barbosa and Jason Mizell may not be household words to you, but their records reside in millions of homes. Switch on the drum machine, hit play on your funkiest drum beat, and enter the world of hip hop producers . . . *Interviews by Deborah Parisi.*

RUN-DMC, SALT-N-Pepa, DJ Jazzy Jeff and Fresh Prince, Biz Markie, and Big Daddy Kane have all made it onto the *Billboard* charts during this week in mid-July. A couple are new entries, but Salt-N-Pepa's *Hot, Cool and Vicious* has been firmly planted for 48 weeks, and they have a new album, *A Salt with a Deadly Pepa* which shipped gold as well. Interestingly enough, these rap groups aren't restricted to the spotlight on the Black Music chart; they're also screaming for and getting attention on the Top Pop Album listing.

Early on in this series of interviews, I came to the realization that no one was really sure what hip hop is. No less than a dozen people asked me just what it was I was writing about, and during the interviews one of the producers said straight out that rap and hip hop are the same thing. Another, while asserting that there is a difference, continually used the terms interchangeably. It was so bad that I began to wonder if I had it right. Luckily, into the midst of the confusion came Run-DMC's Jason Mizell, better known as Jam Master Jay.

"OK," he says, patience exuding from every pore, "rap is what I do; hip hop is a certain kind of sound in music. I rap over all kinds of music. I rap over rock 'n' roll, I rap over R&B, I rap over jazz. And I rap over hip hop. On the new album, 'I'm Not Goin' Out Like That' is kind of R&B; 'Ragtime' is like jazz, my own version of an old Dixieland song. Hip hop would be like 'How'd Ya Do It Dee' and 'Run's House.' It's like a real street sound."

For those not familiar enough with Run-DMC's latest album, *Tougher Than Leather* (Profile Records), let's just say that what he's saying is real. Although this is something of a distillation, hip hop is probably what you think of when you think of rap – the obvious outgrowth of the disco craze of the '70s. Characterized by driving, booming beat box lines and the sounds of records scratchin', it's hard to stand still when it's playing. Whether you love it or you hate it, you've got to admit it's energetic (translate *moving*).

Each of the four guys I talked with started their hip hop careers where it really matters: performing, in the streets or in the clubs. But each has his own unique story.

For Hurby Azor (aka Hurby the Luvbug,

aka Hurb the Supa Def Dope Produsa), it all started when rap was breaking out in his neighborhood. "That was about '78; I must have been 13 when I first heard about it. And I heard about DJ'ing (selecting and mixing the music) before I heard about MC'ing (rapping). It wasn't the way it is now, though; they just wanted to keep two records going continuously. They'd use two turntables and go from one to another – it was a smooth type of thing.

"And then there were people like Flash, who said, 'That's OK, but I'd like to do something a little more exciting.' And he played old records that had just drumbeats, and the drumbeats were real funky, and people would talk over them to make the crowd get up. Next came the rhyming, and it developed into what it is now. I got into it as an MC – I was Hurby the Luvbug, and was in this group called The Super Sevens."

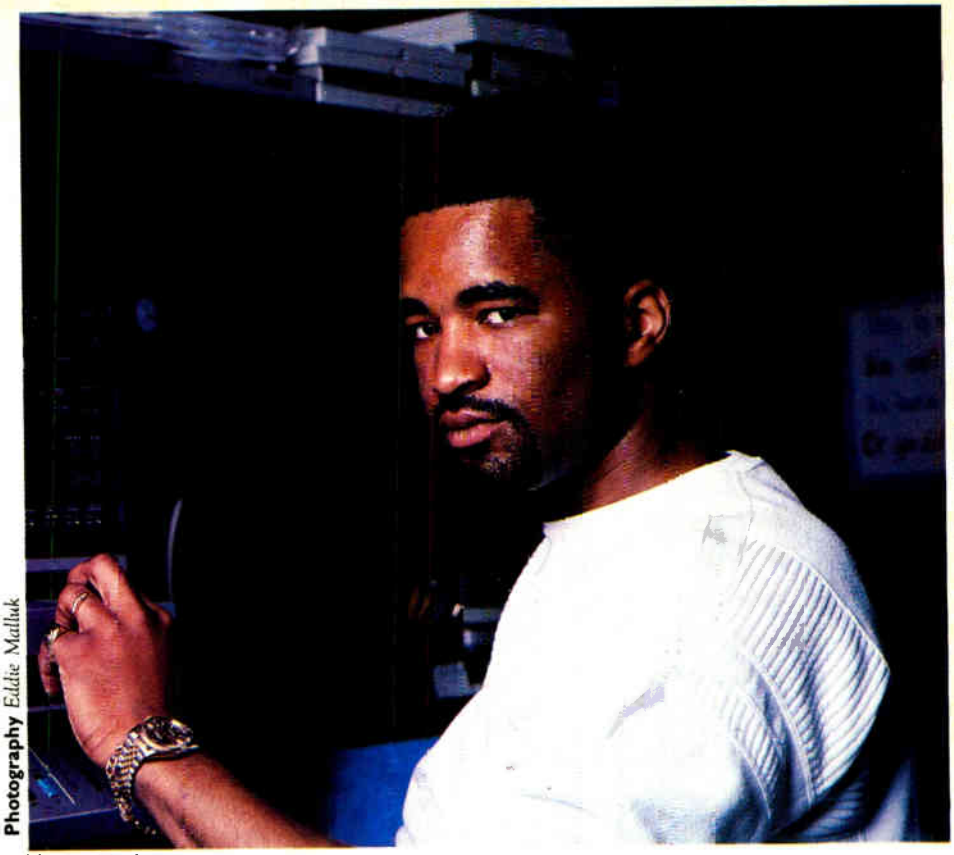
The Super Sevens became the Super Lovers, but they just didn't make the right connections for a record deal. So Hurby went off his own way. "I was working at McDonald's, Sears and going to school – and I was doing all this just to buy equipment," he laughs. "I wanted to buy a studio to make my own records." After a lot of work, and with more than a bit of luck, he put together a girl rap act who hit the mark head on: Salt-N-Pepa. Since then, he's produced a stream of street successes, including Sweet T, Dana Dane and Kid 'N Play.

Chris Barbosa's story is even more of a Cinderella saga. "I was a club DJ, and I bought a synthesizer and a drum machine, just to play along with the dances when I was DJ'ing," he explains. "I had never written a song or anything, but the first day I bought the equipment I wrote a tune called 'Let the Music Play.' I came up with the groove, and then worked with Mark Liggitt, my partner now, and did the lyrics with Ed Chisolm. We put it together with Shannon and recorded it.

"It was a real overnight thing for me," he smiles. "It ended up a Top 10 pop record. And there was no intent. One day I was a club DJ, and a year later I was producing the Bee Gees. Mark and I became exclusive producers at Atlantic Records for awhile, and produced the Spinners, and Robin Gibb, and did some work with Billy Idol.

"We had lots of things at the beginning, but it slowed down for awhile. You learn about the elevator and the yo-yo in the business. And you go back up and down. People who were hot a year ago are not today, and people who are hot today weren't anywhere a year ago." Chris is making waves again in New York on the Ligosa and Warlock labels with his startlingly well-produced releases of singles and albums by Hostyle, Monet, and promising newcomers 3D, as well as working with groups outside the labels.

Few hip hop producers are as hot at this moment as one Marley Marl, who is responsible for Roxanne Shante, Biz Markie and Big Daddy Kane, to name a few. I had been told by a friend that he is actually the inventor of hip hop. "Well, I'm not really the inventor," he says. "Rap was around before I got into it. I never said that I was the inventor. But when it was all going out, I perhaps brought it back. I brought it back with a certain sound.



Photography *Eddie Malluk*

Marley Marl

"I used to be a club jock, and then started doing some remixes for a work-out show," he recalls. "But I wasn't into rap music; I was into dance music and editing and all that other stuff. Then one of the big radio DJs asked me if he could play my rap remix of a song by Buffalo Gal, and I took it down to him on a Friday night. He put it on, and he was getting another record to mix behind me, and I said, 'Hey, that record doesn't mix.' And he said, 'What do you mean? Show me what you're talking about.' So I mixed the record over the airwaves, and he asked me what I was doing on the weekends. And that's how I got into rap radio." Marl still has his own radio show on WBLS FM in New York.

Jasor Mizell, the DJ and co-producer of Run-DMC, is the only one of these four who started as a musician. "When I was nine years old, I had this little group, and we played all kinds of music. We weren't too much into disco, but we did listen to good rock tunes. And that's definitely helped us in Run-DMC. If somebody's playing something wrong, you know it. You don't like it. I still play guitar a little, like I play the wah-wah on 'Tougher Than Leather.' But I want my audience to have the best they can get, so for the difficult guitar parts we hire professionals."

Run-DMC is perhaps the best-known rap act around and have undoubtedly done more for the genre than any other single group. They were the first rap act to earn a gold album (*Run-DMC*, 1984); the first to earn a platinum album (*Raising Hell*, 1986); the first to have their videos aired on MTV; the first to appear on *American Bandstand*; and the first to win that coveted position, the cover of *Rolling Stone*. Their autobiography has been published by the New American Library, their fourth album is said to have shipped platinum, and their second film was released in August. Not bad for a trio of 23-year olds.

ENOUGH OF THIS fanzine stuff, I hear our technical editor scream, where's the beef? OK, here's the truth: if it weren't for technology, none of them would be where they are. Drum machines and samplers, and in some cases synthesizers and sequencers, are every bit as important as the vocals in hip hop.

The sound which Marley Marl added to rap was lean and unprocessed to a great extent – but it originated as much from the equipment he had to work with as it did from intent. "The sound really started with this old 16-track mixing board I've got. It's got like four-band EQ, and I used to just sample kicks and snares from old funk records. I guess I really started all of that," he says. ▶

Photography *Janette Beckman*

L-R: Darryl McDaniels (DMC), Jason Mizell (Jam Master Jay), and Joseph Simmons (Run) of Run-DMC.

►Sampling from old funk records may be the standard of the day (Jam Master Jay takes issue with that), and James Brown is the guy that most often is chosen for sampling. Interestingly, that's what most people thought Marley was sampling on his early productions. "Really, what happened was I was using a record called 'Impeach the President' by the Honeydrippers, but everybody thought it was a James Brown record. It had a real funky kick and snare. Everybody said, 'Oh, he's using James Brown,' and they jumped on the James Brown bandwagon."

Recently, however, Marley's sound has gone through some notable changes, especially evident since Warner Brothers purchase of Cold Chillin' Records. "Well, we're starting to get national exposure, so we're keeping the hip hop sound but adding other elements that the grown-ups may like. But I'll always keep the hip hop drum sound and just add to that."

Marley's drum sound (as well as Hurby's and Jason's) comes from an SP1200 and an S900, but he uses the SP1200 for most of the sampling. "But I don't like using records to sample from for the kick and the snare," he says, "I like to recreate that myself." His synthesizer line-up includes a number of Yamaha and Casio keyboards, as well as the Roland Juno 106.

In his 16-track studio (soon to be moved and upgraded to 24 tracks), Marley has recently added more effects to his productions. "I've really gotten into compressing vocals, especially since we got the Warner Brothers thing," he says. "And you can really hear the difference. And I use a little slap echo, just to give it that street effect, trying to capture the sound of what they used to do in the parks.

"I also don't mind recording noise in the background," he laughs. "You might hear a dog barking on some of the stuff. I've noticed that when I record vocals, especially with Biz, the vocals come out a lot better when he has a little crowd around him. It's a good idea to have vocalists performing, even when you're in the studio."

While Chris Barbosa has produced all kinds of commercial music, he'll almost admit that his specialty is freestyle, previously known as latin hip hop. "Really, it's just dance music in general," he says. "Freestyle has a lot of syncopation and drum programming, a lot of latin-type lines, a lot of sound effects to add an action to the record."

And, yep – there's as much emphasis on drum machines and samplers in freestyle as there is in hip hop; however, there is a different choice in gear. "The 808 rules," he says adamantly. "That drum machine, no matter how old it is or how old it looks or how many machines come out after it, is the one everybody always goes back to. It has that urban sound that started it all. And when you're doing a street record, you want to hear the rim shot, or the cowbell, or that particular decay in the kick drum. That's the first electronic sound that most people's ears heard, and they always get a certain feeling from it.

"Usually we use drum samples in conjunction with it to add a little punch at the bottom," he says, "because a lot of times you can get the ambience of a sound but lose the punch when you sample. So if you combine that with the 808, you get your punch and your sample together."

Chris's sampler is an upgrade, however, from the S900 used by the others. "Well, my partner Mark owns the Publison. It's a really expensive unit, and it's a little hard to use what with all the buttons, but it's really great. It sounds really good."

Chris also relies on effects to create a lot of his records. "I use tons of reverb and effects on the vocals," he says. "Tons. I can't say that I always like it wet, though, because you'll get a mix sometimes where a dry effect will sound really good with a harmonizer. Every time you're in there mixing it's different. But to me and Mark, the background vocals are massively

important. We usually do eight to ten tracks of background vocals, and of course use natural phasing and stuff like that."

The man who has the most equipment at his disposal is, not surprisingly, Jason Mizell, who admits to even using the Fairlight on one song. His last record used those old favorites, the SP1200 and the S900, with the S900 primarily responsible for the samples. "We do all our own sampling," he says, "and really what we do is take a sample here, a sample there, and blend them together, adding on reverbs or what not. But we try to make our own sound, and then we sample that sound. If you want a bass drum, we've got about a thousand bass drums," he laughs. "If you want a snare, we've got a thousand snares.

"Even though you might criticize the sampling and say, 'Well you didn't get on the keyboard and create it yourself,' nobody else has our samples," he says. "I think sampling James Brown is over now, too. That whole '70s sound is in, and he's out, just because he's been played so much."

Jason is in the midst of a new love affair; this time with the MPC60 from the Akai-Linn collaboration. "That's my life now," he says. "I use it for everything. Sampler, yep. Sequencer, yep. Drum machine, yep. You name it; that's what we use it for. It's complicated, and it's got a couple of bugs, but I'm happy with it now. Me and Davy D. make the records, and we keep it in the studio in his basement. The studio where we record our records also has one, so we can make disks at his house and go over the studio with it. I didn't buy it until after *Tougher Than Leather*, but I used it on three of Shinehead's records. And I did one of Real Roxanne's records with a combination of the MPC60 and the SP1200."

Pick a piece of equipment, any piece of equipment, and you'll always get opposing views. Hurby Azor on the MPC60: "I bought the Roger Linn drum machine, but I think they messed it up a lot," he says. "I think it's supposed to be the drum machine version of the S900 (*it's actually closer to a vastly updated Linn 9000 – Tech Ed.*), but there are so many things that the S900 has that this one doesn't."

So while the MPC60 sits off in a corner, the S900 and SP12 get a real work-out (Hurby dislikes the SP1200 for the same reason that he doesn't like the MPC60 – you can't use the entire sample memory for a single sample). "I've stopped looking for a sampler, because I'm happy with the S900. It has multi-pitch and a lot of little things in it that I can use. I do a lot of sampling, only because the people in this rap era are used to listening to old rhythms and old tunes. And to try to recreate, let's say, a James Brown drum pattern, even if you get a drummer out there playing the exact thing, it won't sound the same because of the way they used to record back then. They had this room sound, and the snare was real nasty."

Hurby's recording techniques are a bit different than the other producers. "I don't use signal processors," he says, "it's not happening now in hip-hop. Reverb, hardly ever. If I sample drum sounds, they have everything I need on them. My sound isn't like that dance music, with that gated reverb sound. I don't gate anything, because if there's noise there, the noise is part of the record. I don't compress anything because everything is already there. In fact, I just don't use signal processors unless I have a real dirty sample."

But how about EQ? "EQ, yeah," he smiles. "EQ plays a good part, because if you sample the guitar line that has a drum behind it, you have to take it out. If you take out the low end, it makes it stand out more and makes it sound really old, too. Of course, you lose some of the guitar, too, but it sounds good, so I use it. It sounds like something somebody did in the street – and *that's* what sells rap records."

When I mention to Hurby that he's more than a bit different than other producers, especially pop/rock producers, he laughs. "Right. Exactly. And that's why they're pop and we're rap. Harmonies are always perfect in pop – they have a main bassline along with chords. We never use chords. We don't even know what chords are, except that you put a bunch of notes together and call them chords. If we don't know it, how can we use it?" he asks a bit craftily. "If you want a rap flavor, throw away the music book! Make it raw. It has to be raw."

NEW YORK TIMES; May 16, 1988; Headline: *It's Official: Rap Music is in the Mainstream*. And therein lies the rub – there's a growing debate over whether or not the major labels will destroy rap music with their tendency to homogenize everything they touch. It may come as no surprise that the group which is almost solely responsible for bringing hip hop into a larger audience and gaining greater appreciation is also the group responsible for more than a bit of the commercialization. But Jam Master Jay remains dedicated to the form and maintains his belief in the quality of Run-DMC's product.

"Everybody's out to get Run-DMC," he says. "There's definitely an attitude out there. See, Run-DMC's whole thing at the beginning was that we were the underdogs in the streets, and now everybody in the streets doesn't feel like we're the underdog any more. So we have to win in other ways. If we keep making records the way we do, they can't help but be on our side a little bit.

"As long as I'm thinking of making good street records, the street will always have good ones," he says. "And even though the major labels have it, there's always going to be somebody that doesn't make the majors. There's always going to be somebody in the basement making the same-o same-o."

Photography *Eddie Malluk*

Chris Barbosa

"I'm not going to soften up," claims Marley Marl regarding Warner's purchase of Cold Chillin'. "They do get mad, though'; they go, 'Hey, you've already got this audience, now we're trying to get this other audience.' But we've got to keep the feel; I'm always going to keep the feel."

"I think that what's happening now is what's always been happening," concludes Chris Barbosa. "The majors are always going to be buying something. Independents put out the records first; the independent is really A&R for a major label. Once the record starts happening, the major will want to come in and buy it. Hopefully, they're smart enough to figure that it's working the way it's working and not try to mess with it But unfortunately, it's usually true that when major labels get ahold of something they tend to commercialize it."

Hurby Azor pulls no punches: "The majors will never be able to do it. They can't see it. Hip hop is a taste, and a feeling. It's something that I feel. And you've got to know how the people you're making the record for feel or act. People at record companies don't know how to do it because they're not out there.

"Rap is going to change into something else eventually because of the majors, and when it does we'll have to move on to something else. But nothing's going to happen to me. I just roll with the times. Whatever's happening is what I'm doing," he says, laughing. "I don't create music; I just work with it. I'm not a trend setter; I just make trends better."

Whatever happens to the hip hop genre in particular, or rap music in general, the one thing we can all count on is the fresh and creative attitude coming out of the streets of the cities. Long live rap. ■

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Yamaha MIDI Grand Piano

While MIDI is undoubtedly the performance aid of the '80s, many players still regard the grand piano as the ultimate instrument. Does Yamaha's MIDI grand represent the perfect fusion of tradition and technology? *Review by Simon Trask.*

LET'S NOT BEAT about the bush. Despite the many advances in music technology made in the past few years, ultimately there's still no substitute for a real acoustic grand piano. Not surprisingly, then, many an upmarket recording studio still boasts a grand piano in among its arsenal of hi-tech gear, and the beast has yet to be banished from the touring schedules of the rich and famous. The grand piano may be on the big side, but unlike dinosaurs, it isn't about to become extinct on us.

As MIDI technology continues to invade the lives of musicians and recording engineers, the time is ripe for someone to unite MIDI and the acoustic grand. Retrofits have been available, but best poised to build from scratch such a beast seems to be Yamaha, who have a long history of piano manufacturing to complement their hi-tech expertise.

Seeing the Light

UNTIL YAMAHA'S ENTRY into the field, the only option open to musicians and studios wishing to bring MIDI to the acoustic piano has been the Forte MIDI Mod, a retrofit modification which adds extra mechanics to the piano's existing mechanics in order to achieve its ends. The disadvantages of the MIDI Mod is that it tends to alter the feel of the piano keyboard and it has a limited MIDI spec (which belies the piano's potential as a controller keyboard for a MIDI setup).

Yamaha has avoided the problem of extraneous mechanics by employing a system of optical sensors which "read" light beams emanating from red pinpoint LEDs. Each piano key has a pair of light beams which are located in the path of the hammer. The optical sensors detect when these beams are broken, allowing both the speed (velocity) and the timing of the hammer to be detected just before it hits the strings, while a further optical sensor located beneath each key senses when the key is released. This system (which was first seen on Sequential's Prophet T8 keyboard) works remarkably well, producing unerringly accurate response and no delay.

With the aid of a pressure-sensitive strip located underneath the keyboard (the only "mechanical" addition to the piano), Yamaha have also brought channel aftertouch capability to their MIDI Grand. Not to the piano itself, you understand (let's not get carried away here – we'll leave mechanical pitch-bend bars to the likes of Jan Hammer).

But this is only half the story, because Yamaha has given their piano a thorough complement of MIDI control features. These require a digital-access control panel to be built into the body of the piano, immediately above the keyboard. In case you're shuddering at the thought of major surgery being visited upon your precious grand, I should point out that the only surgery involved will be extricating it from your studio/conservatoire/living room – remember that this isn't a retrofit; Yamaha has built it into their own model C3E and C7E grands.

Yamaha's decision to go with their own piano means that if you want MIDI you'd better like the piano that comes with it. Fortunately, most people are unlikely to be put off by Yamaha's piano, which has a firm, rich tone, and a consistently fine quality over the entire keyboard. The keyboard itself feels very comfortable to play, and has above all a very *musical* responsiveness.

Panels and Pedals

REALIZING THAT PART of the grand piano's appeal lies in its appearance, Yamaha has managed to minimize the visual impact of their hi-tech additions. For a start, the I/O panel is tucked away on the underside of the piano. Here you'll find the power switch (no jokes about "Switched on Bach," please . . .) together with one MIDI In, four MIDI Outs, a breath controller input, two footswitch inputs (MIDI Out on/off and memory change) and two footpedal inputs (volume and modulation). Meanwhile, at the left-hand end of the keyboard are the two wheels

familiar to synth users (namely, pitch-bend and modulation). No, Yamaha hasn't come up with the first undulating, warbling acoustic piano powered by human breath – again, all these controls are MIDI-only.

Foot controllers are nothing new to the acoustic piano, of course, and the MIDI Grand has the usual trio of damper (sustain), soft and sostenuto pedals. The action of the first two can be conveyed over MIDI, but for some reason the sostenuto pedal remains resolutely acoustic (odd, because there *is* a sostenuto pedal defined inside the MIDI spec).

Also on the underside of the piano (but readily accessible) is a lever which allows the piano sound to be muffled. Pulling the lever forward brings a row of felt pads up against the strings. Assuming that you don't want to layer acoustic piano over *all* your electronic sounds, the muffler is a very good compromise – with even modestly-amplified MIDI instruments, the piano is barely audible.

The most visible sign of hi-tech influence on the MIDI Grand is the aforementioned control panel. Though Yamaha has made it as unobtrusive as possible, there's no getting away from the centrally-located 2×40-character backlit LCD window. Operationally the panel is well thought out, with many dedicated function buttons and never more than two parameters per button – all of which helps to lessen the potential headache of digital access.

Grand Performance

THE MIDI GRAND'S parameters can be stored in 64 Performance memories which are selected from the front panel (in bank/number format) or remotely via MIDI patch changes. These memories can be SysEx bulk-dumped to an external MIDI storage device (Yamaha's MDF1 MIDI Data Filer, for instance) for subsequent recall if required.

The MIDI Grand uses two processors running in parallel to handle MIDI Outs A+B and C–D respectively. For each processor you can define volume, pitch-bend, modulation, sustain, aftertouch and breath controller on/off status together with values for low/high level scaling (+/–15), low/high key limit (A-1 to C7), transposition (+4/–3 octaves in semitone steps) and MIDI Out channel (1-16).

Additionally you can define portamento time (0-127), volume level (0-127), patch number (1-128) and on/off state for each of the four Outs. The control panel has a dedicated on/off button for each Out, together with a global on/off button. In addition, as mentioned earlier, MIDI transmission can be turned on/off globally from a footswitch. Fortunately, Yamaha has successfully avoided the potential problem of MIDI drone notes.

The key-limit settings allow you to define two MIDI zones – one for each

processor. Each zone can cover any range from an individual note to the whole keyboard (so of course it's an easy matter to overlap zones). Level scaling allows further variation by letting you determine how and where MIDI'd sounds will fade in/out across the keyboard.

With an instrument hanging off each Out you can layer two MIDI'd sounds per zone, with independent patch selection on each instrument and the ability to preset the volume balance. Of course it's also an easy matter to limit MIDI'd sounds to selected area(s) of the keyboard, leaving the rest piano-only, while further textures can be created if your slave MIDI instruments themselves allow split textures.

Surprisingly, Yamaha hasn't included velocity scaling over MIDI as a means of tailoring the responsiveness of MIDI instruments to the touch of individual musicians.

To take advantage of the MIDI Grand's zoning features you need to use at least two of the piano's MIDI Outs, a consequence of the piano's "hard-wired" system architecture. Another consequence is that you can't take advantage of multitimbral instruments, while if you want to record into a MIDI sequencer then you have to forgo the piano's zoning features altogether.

Incidentally, it's worth mentioning that, although the piano itself is 88-note polyphonic, for MIDI transmission purposes polyphony is reduced to a more modest 16 notes (again, odd – sure, a DX7 has just 16 voices, but the usual philosophy over MIDI is to send everything and let the receiver deal with it as it chooses).

If you're into MIDI diagnostics, then you can summon forth the MIDI Monitor display, which allows you to read the output of the two processors and the input to the MIDI In socket (in hexadecimal, I should add). Courtesy of the User Program function, you can also enter your own MIDI messages in hex, up to a total of 40 bytes, for each Performance memory. The same data sequence is transmitted over all four Outs, though transmission can be turned on/off for each Out. The uses (and abuses) of this feature depend on your ingenuity, but obvious examples include sending a Start code to a drum machine, and sending additional patch changes to MIDI'd signal processors.

"The piano is 88-note polyphonic, but for MIDI transmission polyphony is reduced to a more modest 16 notes."

The logical next step for Yamaha would be to turn the MIDI Grand into a late '80s player piano – with MIDI sequencing substituting for the traditional piano roll. This would allow the Grand to be fully integrated into a modern-day sequencing setup. In fact, Yamaha already produces an upright player piano (the MX100R Disklavier) which offers limited MIDI features coupled with internal sequencing and storage to a built-in disk drive. Currently, however, there's no definite plan for a playback option to be added to the MIDI Grand, so owners may have to wait a while for those capabilities.

Conclusions

IT'S HARD TO overestimate the scale of Yamaha's achievement. The bottom line is that Yamaha's system *works*, and that it affects neither the feel of the keyboard nor the sound of the piano. There are a few oddities to the MIDI implementation (like the splits being actually physically split among two output jacks), but I feel the MIDI Grand represents a significant merging of old and new technologies, with neither side losing out in the process.

Quite simply, it's in a class of its own. ■

PRICE Neither the C3E (6') or C7E (7'4") models have list prices but you can expect to pay something around the range of $25,000.

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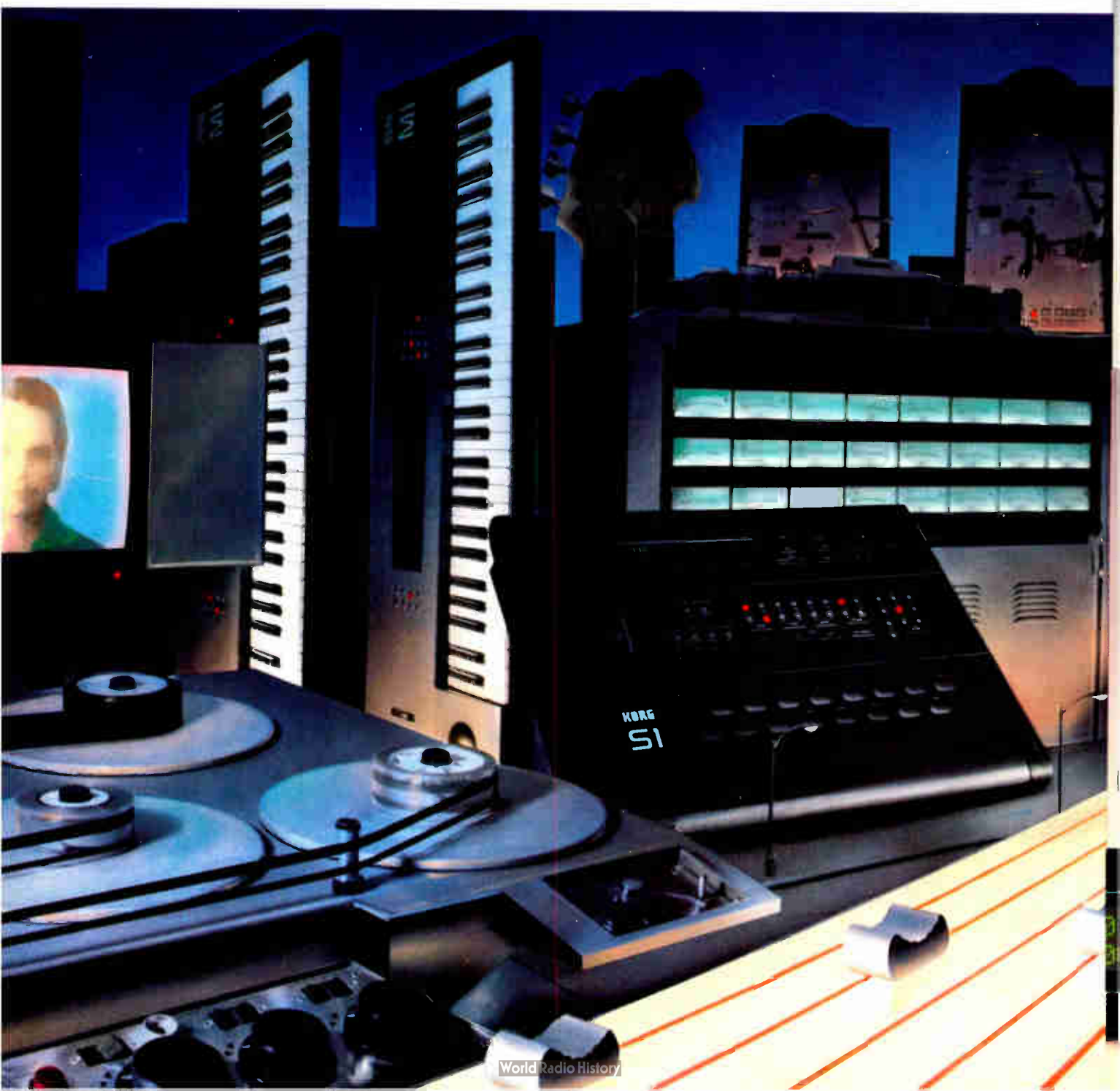

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TAKING • CONTROL

THE SUMMER NAMM REPORT

With the absence of several major manufacturers, this summer's NAMM convention was nearly the show that wasn't, but there were some interesting developments in controllers, accessories and software, as the following report reveals. *Text by Bob O'Donnell.*

NORMALLY, THE SUMMER NAMM (National Association of Music Merchants) show is one of the major launching-grounds of new products in the music industry. This year, however, it seemed like the hi-tech side of the music industry decided to slow down a bit to catch its breath. Contributing to this was the fact that several hi-tech manufacturers (Roland, Korg, Akai, Ensoniq and E-mu amongst others) announced prior to the show that they wouldn't be attending for a variety of different reasons. As a consequence, things were slow.

Not that there wasn't anything new or exciting introduced in Atlanta (where the show was held). Major developments in hardware seemed to center around controllers and accessory-type products, while innovations in computer software centered around notation and algorithmic composition packages. If you're looking to refine your MIDI and recording systems, there was (and is) plenty of stuff worth investigating in the following pages.

The Noisemakers

AMAZING HOW WE get used to the world getting reinvented every six months, isn't it? In spite of that expectation (or perhaps because of it) probably the biggest surprise for me at the Atlanta Show was the utter *lack* of new sound producing instruments. Sure, there were a few repackaged machines, but not a single new synth was anywhere to be seen (or heard). **Kawai** introduced the KIR ($629), which is a single-space rackmount version of their hot selling KI synth (see MT June '88 for review). Similarly, **Casio** showed their VZ10M ($1199), a three-space rackmount version of their VZ1 synth (see review elsewhere in this issue) which adds an XLR mix output for studio applications.

If you're looking for inexpensive velocity-sensitive MIDI keyboards, Casio also introduced two new 12-bit digital pianos – the 61-key CPS300 ($499) and the 76-key CPS700 ($799). Intended to replace their ill-fated CDP digital pianos, the new entries each offer five different sounds: acoustic and electric piano, vibes, harpsichord and pipe organ.

Kurzweil's announcement of Sound Blocks for their 1000 series products should be good news for existing (or prospective) owners. For $495 ($595 for K1000 owners), 1000PX users will be able to add EPROM chips which give the instrument new samples, including drums, electric bass, electric piano and flute among others. Best of all, these new sounds are simultaneously available with the existing ones. (See *Patchware* this issue for more details.) Additional sound blocks for the PX and the other 1000 series modules will be forthcoming.

Yamaha formally introduced the PF2000 ($2395), an FM digital piano with 12 preset voices that has the equivalent of two DX7II tone generators. The very sleekly designed instrument also includes a basic two-track sequencer, a built-in power amp and speakers, and the ability to play DX11 patches via its cartridge slot.

On the sampler front, **Dynacord** displayed

KTI's new GZ1000 keyboard controller was nearly destroyed by the rampaging Godzilla dolls that attacked the company's booth.

more finished versions of their ADS 16-bit stereo sampler in both rack-mount ($4895) and keyboard (slightly higher) formats as well as their ADD-two drum sampler ($4995). **Steinberg/Jones** officially unveiled the UK-created Lynex 16-bit stereo sampler (no price yet) for the Atari ST. Housed in a single-space rackmount, the Lynex can be either eight- or sixteen-voice polyphonic, has eight independent outs, and comes with 1Meg of its own RAM (expandable up to 32Meg) as well as its own sampling and voice editing software. This looks to be competition to Hybrid Arts' ADAP system (which Hybrid was showing with hard disk recording). **Simmons** demo'd their integrated SDX 16-bit sampling system and some impressive acoustic piano sounds with their new keyboard software (a free upgrade to existing owners of the $7960 system).

Of course the company is normally associated with innovative electronic drumming products, and to that end they showed a prototype of the SoundStation (no retail price yet), an electronic drum brain with built-in 12-bit samples which uses some of the technology developed for the SDX. Simmons also showed the Trixer ($1350), a handy little rackmount box designed for acoustic drummers who want to trigger sampled sounds. The Trixer combines a trigger converter, a sample player, a 6X2 mixer – to combine the acoustic drums with the built-in sampled sounds, and an integral digital reverb to finish off the mix. Acoustic drummers looking to beef up their sounds may need to look no further.

The Playmakers

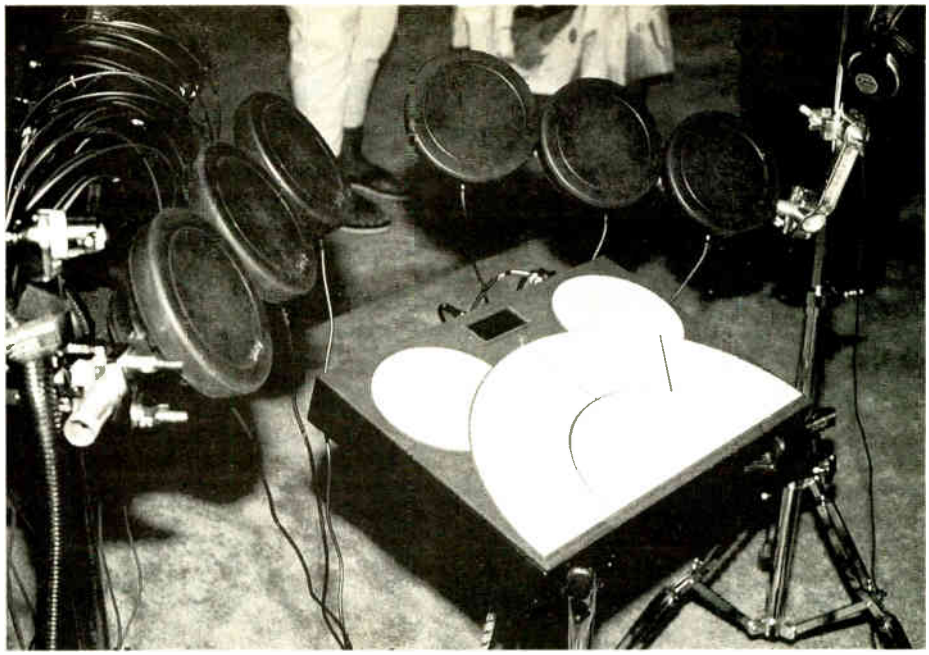
IN CONTRAST TO synths, new controllers were in abundance at Atlanta. **Keyboard Technologies, Inc.** (KTI) formally unveiled the GZ1000 ($3995) MIDI keyboard controller with real piano keys and adjustable action. One unique feature of this instrument – which is being built by American piano manufacturer **Baldwin** and was also shown retrofitted to a Baldwin grand piano – is its ability to store SysEx dumps from the synths being controlled and actually display the names of the patches for the various synths. Pretty cool.

No new drum machines were introduced at NAMM but two new drum controllers should be of interest to those with a rhythmic bent. **Simmons** displayed a more finished version of the Portakit ($999), which includes 12 differently shaped pads, six trigger inputs and a built-in polyphonic sequencer. The real star of the show, however, was the new Drum Kat ($995) from **Kat MIDI Controllers**. In addition to having 10 pads which can be grouped into any type of zones, this sturdy little box has nine trigger inputs that can each have their own gain and mask settings, two mergeable MIDI inputs, four MIDI outputs, four footswitch inputs and a click out. Each pad can transmit up to three MIDI note messages at once – each with an independent delay time if you want – or send any type of other MIDI message, and the strength of the pad hits can dynamically affect velocity, pitch or gate time. Finally, the Drum Kat also has an internal sequencer and a built-in function which can control the speed of external MIDI sequences depending on the tempo at which you play, à la the Kahler Human Clock.

Other controller news included the official unveiling of **Yamaha**'s GI0 MIDI guitar system ($2499) (see MT August '88 for preview), demonstrated by Lee Ritenour and his new band. A journey to the hinterlands of the exhibit floor also turned up the Synthophone ($3450), a MIDI saxophone made by **Softwind Instruments** in Switzerland. Consisting of a modified Yamaha alto sax and a small power supply, the Synthophone produces no acoustic sound of its own, but instead generates MIDI information based on the key, lip and wind pressure sensors put into the horn during the modification. If you don't like the feel and action of the other electronic wind controllers, this may prove to be the answer. If, on the other hand, you're a vocalist who's interested in MIDI control, you'll probably be glad to hear that the Imagine Music Group has picked up the US distribution of **Digigram**'s MIDIMic ($349) – a basic pitch-to-MIDI converter in the shape of a microphone.

Studio Stuff

DIGITAL SIGNAL PROCESSORS were quite the rage in Atlanta, as they have been for the last few NAMM shows. Like clockwork, it seems, Alesis, ART, DigiTech and Yamaha all managed to top their previous work with yet more bang for the buck. The



One of the most impressive new controllers introduced at the show was Kat's new Drum Kat.

latest wonder from **Alesis** is the Quadraverb ($449) a stereo input, stereo output programmable signal processor which can produce up to four different effects at the same time, including reverb, chorus, delay, parametric EQ and pitch-shifting. The unit features 20Hz-20kHz bandwidth and its MIDI features include the ability to alter program parameters in real time via MIDI control. Alesis also introduced the Micro EQ ($125) and the Micro Cue Amp ($125), the latest additions to their one-third rack space Micro Series.

New from **ART** is the MultiVerb ($575), a 16-bit programmable signal processor capable of doing four effects at once, including – does this sound familiar? – reverb, chorus, delay, EQ and pitch-shifting. This directly competing unit only offers a 15kHz frequency response, but it has twice the memory space of the Quadraverb for storing programs (200 locations). Neither the Multiverb nor the Quadraverb were completely functional at the show (though the ART unit was much closer), so direct sonic comparisons will have to wait for the reviews . . .

DigiTech announced several new goodies, including the impressive sounding IPS33 Smart Shift ($799), an intelligent harmonizer. Like Eventide's long-awaited Ultra-Harmonizer, the IPS33 can produce harmonies that fit the scale that you're playing, instead of just the parallel harmonies that most pitch shifters produce. The 99 available preset locations can be recalled via MIDI – which is where the company's new PDS1280 footpedal ($169) comes in. With it you can increment or decrement your way through MIDI program change messages which can be sent on channel one to any MIDI device.

The GPO16 Guitar Processor ($899) – it stands for Guitar Players Only (really) – is DigiTech's answer to Roland's GP8. Offering up to six effects at once, including distortion, digital reverb, chorus or flange, delay, parametric EQ and compression, the single-space rackmount has 128 presets which can be selected via MIDI.

For those of you who prefer the Tom Scholz sound, but want MIDI control, **Scholz R&D** has solved your problems with the MIDI Octopus ($239). The half-rack size unit connects to all the footswitch inputs on the back of the Scholz products and allows you store on and off settings for them in up to 100 locations which can be recalled via MIDI.

Yamaha's DGA (Drums, Guitars and Amplifiers) division also had a new multi-effects processor intended for guitarists, the GEP50 ($695). The unit looked and sounded remarkably like an SPX50D ($695), one of the Pro Audio division's newest toys, but apparently its factory programs and operation are geared more specifically towards guitarists. The other new signal processor from Yamaha Pro Audio is the half-rack space R100 ($295), which has room for 60 programs, including reverb, delay and reverb, early reflection, stereo echo effects and more. The little unit, which is intended to be a companion to the MT100 ($495), the latest in their line of four-track cassette recorder/mixers, is programmable, with four different parameters available per program.

Peavey also introduced some new and updated signal processors: the UniVerb II ($299), the AddVerb II ($399) and the UltraVerb ($449). The UniVerb II greatly expands on its predecessor by upping the program count to 128 (from 32) and adding delay and echo effects. Except for the fact that it's completely non-programmable, it seems to be somewhat similar to the original AddVerb. Speaking of which, its new version adds additional memory for more programs and longer delay and reverb decay times. Finally, the UltraVerb adds complete programmability to the features of the AddVerb.

Peavey Audio Media Research's new Autograph AEQ2800 ($549) is a programmable, MIDI-controllable 28-band EQ

with 128 memory locations. The single-space rackmount features a large display (2×40 characters) and the ability to analyze a room and instantly create a flat response. Once it's done that, you can then superimpose one of your preferred EQ settings over the top of it. The unit should work well in conjunction with the company's PLM 8128 Programmable Line Mixer ($1199) and PLM8128E Expander Modules ($999), which they displayed in more finished form. The eight-channel mixer offers complete programmability over level, panning, monitor level and three effects sends per channel and can be completely controlled via MIDI automation. In addition, users can store "snapshot" settings in 128 locations. Peavey AMR's other new MIDI-controllable studio device is the MAP 8×4 Programmable Audio Patchbay ($429). The single-space rackmount features eight channel inputs and outputs (all ¼") as well as four send/return loops and can store 128 routing programs.

The big news at **Yamaha** had to do with increasing their presence in the MIDI programmable mixer field via the US introduction of the DMP7D ($5995) and the worldwide first appearance of the DMPII ($2395). The DMP7D is functionally identical to the original DMP7, but it has digital inputs and outputs. Thanks to several different conversion boxes also being offered by Yamaha – the IFUI, 2, 3 and 4 (priced between $525 and $875) – and a slew of different types of connectors on its back panel, it can support any popular digital audio format. *(For more details on digital audio, see the Digital Audio 101 and 102 articles in MT August '88 and this issue).* Yamaha also announced the availability of stand-alone high-quality A/D and D/A converters – the AD808 ($4995) and the DA202 ($1195) – to connect conventional analog signals to and from the DMP7D, as well as the PLSI Programmable Line Selector ($795). The three-space high rackmount has four groups of eight inputs and eight outputs and permits you to select various combinations via MIDI program change messages.

The latest low-cost wonder from Alesis is the Quadraverb, which gives four effects at once for under $450.

The DMPII is a slightly cut down but much less expensive version of the original DMP7 housed in a four-space rackmount. It lacks the motorized faders of its older brother and has only two bands of parametric EQ and two built-in effects processors (as opposed to three of each), but its graphic display allows you to see the position of each fader via a bar graph, and its signal-to-noise ratio (92db) is slightly better than the DMP7. I want one.

In the analog realm, **Passac** showed the Unity 8 professional stage mixer ($395). Squeezed into the single-space rackmount are eight inputs, two stereo effects sends and returns, left and right balanced outputs and two monitor outputs. Each channel provides control over volume, panning, effects send level and selection of effect 1, 2 or both. The mixer boasts a signal-to-noise ratio of 115db.

Moving over to tape recorders, **Tascam** introduced the MSR16 ($7000) a half-inch 16-track. Even more interesting, though, was the company's new MIDiiZER ($1999), a combination SMPTE/MIDI converter, synchronizer, and autolocator for the MSR16 and the 238 eight-track cassette recorder. By virtue of its tempo map functions and other capabilities, it allows you to synchronize two audio tape decks (or one audio and one video with an optional conversion unit) and a MIDI sequencer and reference everything to bars and beats, instead of just SMPTE frames and subframes. Tascam also showed a Macintosh Hypercard application remotely controlling their 238 cassette deck, and mentioned that they hope to see the day where control of tape transports is built into MIDI sequencers.

Answering Tascam's Model 238 eight-track cassette deck, **Toa** finally announced the US introduction of their eight-track cassette recorder, the MR8T ($2149). Available in Japan for over a year, the MR8T has basic onboard mixing facilities, including insert points for effects, in its four-space rackmount housing. ►

► To connect all this and your other equipment together in a very high quality fashion, **Monster Cable** would probably recommend their new Prolink Series 4 Instrument Cable ($24.95 for 20-foot cable) and Prolink MIDI cable ($19.95 for 12-foot cable), both of which feature 24K gold-plated connectors. Finally, to organize all those cables, a new company appropriately called **Get Organized** offers their Cord Control Kit ($34.95), which includes a variety of ties, clips and labels, as well as their Snake It plastic tubing.

MIDI Boxes

OVER IN THE MIDI effects and control-oriented hardware area there were several interesting innovations. First off is the impressive-looking Feel Factory ($650) from **Future Lab**, which is being distributed exclusively by **Filmsonix**. Subtitled a "MIDI Algorithmic Feel Composer," the Feel Factory offers eight sliders with which you can adjust the timing of MIDI note information from sequencers and drum machines so as to produce subtle (or overt) changes in the music's feel. Developed by the same person who designed the Human Clock, the box starts the sequencer ahead of the track it is supposed to be slaving to, and allows moving events backwards or forwards in real time against the target time by individually delaying tracks. The unit will also allow you to apply preset or user programmed "feel algorithms" to sequenced material, and with the addition of an optional port, will permit you to analyze "feels" from existing audio sources and place those timing nuances onto your sequenced music. Last but not least, the Feel Factory can read and write SMPTE and includes a built-in Macintosh MIDI interface.

In a related development, a new version of the Human Clock, Studio E ($299), was also announced at the show. The new box, which is to be distributed by **Imagine Music Group**, is said to operate better and be easier to use than the original.

MIXI ($999) is a new type of MIDI mixer and MIDI controller developed and produced by **Blue Sky Logic Corporation**, a new entrant to the music market. The plain looking box contains one switch, two sets of MIDI ins and outs, and 17 faders – each of which can be assigned to send any type of MIDI message. One basic mixing application is to use 16 faders to send MIDI volume messages on each of 16 MIDI channels, but you can also assign some sliders to send program changes, while others send out MIDI panning messages and volume messages on several different channels.

Some new performance oriented products come to us via **Oberheim** and their new PerF/X line of MIDI Performance Effects. The first two of what promises to be several similarly shaped and priced boxes are the Systemizer ($249) and the Cyclone ($249). The very handy looking Systemizer organizes a MIDI system by giving extensive control capabilities to any MIDI keyboard or other controller. In any of the 32 available memory locations – the number can be expanded to 100 via a RAM card – you can set up a patch which will send multiple program changes and controller setup information to numerous synths. It will also give the main keyboard up to three fixed or floating splits (which can also be permanent or temporary – while a footswitch is held down) with four instruments per split. In addition you can crossfade between several instruments, filter and scale controller information, and combine different instruments into one with Group Mode.

The Cyclone, on the other hand, is basically a sophisticated arpeggiator. You can record up to 32 notes into it and then separate the pitches from the rhythms and manipulate them or combine them with other sequences in a variety of ways, including transpositions, inversions and doubling.

S-S-Software

ORIGINALLY REFERRED TO as the "MIDI Ghetto," Software Alley has expanded to be one of the largest regions on the show floor. Given the lack of major hardware introductions this time around (as a matter of fact, one of the biggest hardware introductions was Yamaha's CI MS-DOS computer we snuck word about to you last month), all of the other new developments at NAMM were in software.

For those of you who didn't read about **Yamaha**'s new computer, however, here's a quick overview: the CI ($4000 w/20Meg hard disk, $3000 with two 3 1/2" floppies) is a laptop IBM compatible – with a 640×400

backlit LCD display, an 80286 CPU, and 1.2Meg of RAM – that has been designed specifically for electronic music applications (though it can also run any standard MS-DOS programs). Specific music features include 11 MIDI ports (eight Outs, two Ins and one Thru), a built-in SMPTE reader/generator, a ROM-based music font, two front panel sliders which can be assigned to control any function, and two special chips within the machine which are dedicated to timing and the handling of large amounts of memory respectively. The CI also has one expansion card slot, a printer port, two RS232 serial ports, and an output to hook up a larger external display. It comes standard with MS-DOS 3.3 and a MIDI monitor/bulk data manager program.

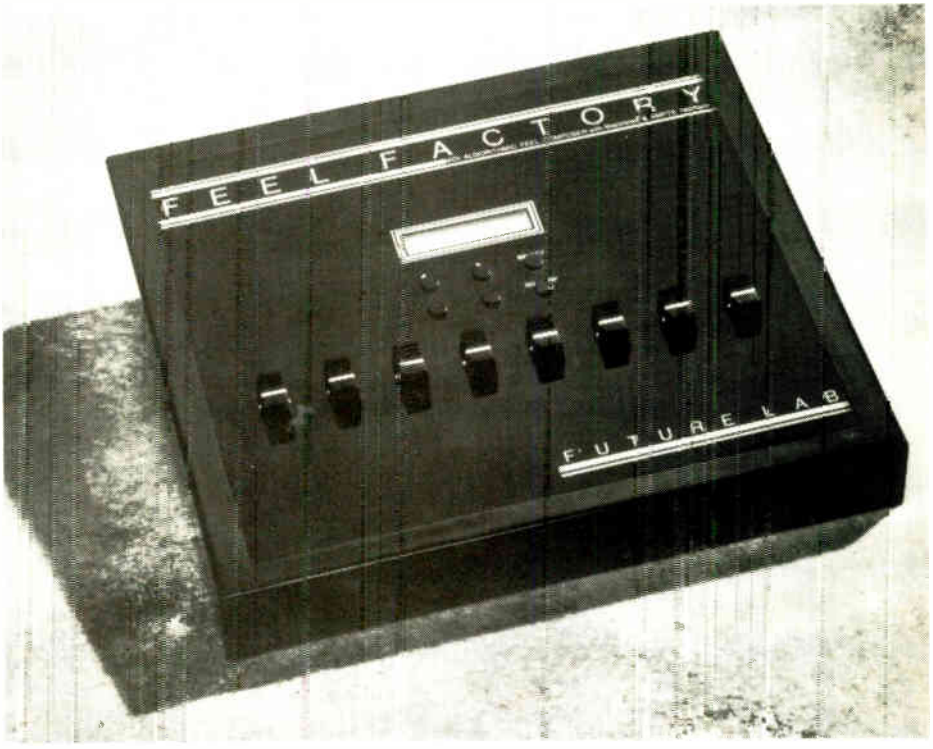
Sequencers and Notation

THE BIGGEST SPLASH was undoubtedly created by the formal introduction of **Coda Music Software**'s Finale ($1000), an extremely impressive music publishing and composition program available on the Macintosh and soon to be released for the IBM. (Versions for the ST, Amiga and Apple IIGS are apparently in development.) Finale combines the capabilities of a sophisticated MIDI sequencer with intelligent real-time music transcription and extensive editing capabilities for music, lyrics and any types of symbols. You can print out the publishable-quality notation on any Postscript laser printer (making use of the program's own Petrucci font or the standard Adobe Sonata font), on a Linotronic, or on a basic dot matrix printer. The program's only pitfall is its hefty price, but representatives from Coda mentioned that less expensive versions of the program with less features will be forthcoming (and apparently they'll also be available on all major computers).

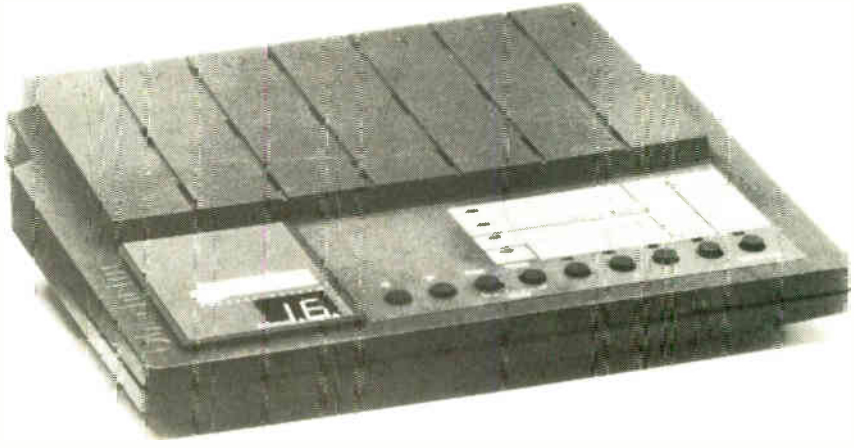
Passport also introduced two new notation programs for the Mac. Encore ($395) works as a companion program to MasterTracks Pro and Jr, converting sequence files from those programs as well as standard MIDI Files into notation, or as a stand-alone music composition package. It can transcribe real-time and step-time MIDI input and offers graphic editing of MIDI controllers such as pitch-bend, etc. Notewriter ($295) is a non-MIDI notation program that's intended specifically for music copyists. Its graphics-oriented approach allows fine tuning of the placement of different musical symbols and permits it to create graphics files to be used in desktop publishing and other types of applications.

A few combination sequencing/notation programs for the ST were also on display in Atlanta. **Digidesign** was showing a nearly completed version of **C-Lab**'s Notator ($595), which combines all the features of the company's Creator sequencing package (see review in MT April '88) with extensive notation capabilities, the very hip multi-tasking features of this program allow it to print one score while playing a different sequence. **Sonus** was displaying Superscore ($299), which is reviewed elsewhere in this issue. ▶

The DMP11 is the newest digital mixer from Yamaha's Pro Audio division.



Future Lab's Feel Factory gives you minute control over the timing and feel of sequenced music.



New from Oberheim is a series of MIDI control boxes, including the very handy Systemizer.

Yamaha's new CI computer was one of the highlights of the show.

▶Two other ST packages which include some notation functions are Studio 24 ($225), a French program being distributed in the US by **Imagine Music Group**, and Symphony Orchestra MIDI ($349 Canadian) a full-blown sequencer package with some basic notation functions. An optional piece of hardware from **OCC Informatique**, the French Canadian company who produce the Symphony Orchestra software, is the SPMX ($250 Canadian), which reads and writes SMPTE and MTC, can read VITC (vertical integral timecode – which is written to video tape) and has four MIDI Ins and Outs.

For you IBM-types, **Erato Software** demonstrated their Erato Music Manuscriptor system, or MUMS ($5800), which combines a graphics tablet and a large plasma display along with the IBM software. Notes and symbols are entered by choosing them with the stylus connected to the graphics tablet – it has an overlay that contains all the available choices. Playback over MIDI is supported and printout can be to a laser printer, plotter or dot matrix printer.

Among other sequencer developments, **Performance MIDI Systems** announced Pro MIDI Player ($149), a live performance-oriented sequencer for the ST. The program has no editing capabilities, but it allows you to instantly access any of 26 songs in available memory. Interestingly enough, it will send the name of the selected tune to the LCD displays of whatever synths you have connected to it, so that you don't need to use the Atari's monitor onstage. The company also announced the ML2412 (no price yet), a companion 24-channel, 12-scene MIDI lighting controller.

Dr. T's showed a working version of their KCS Level II sequencing program for the Macintosh ($325) as well as a version for the Amiga ($325) and announced Phantom ($249), a software program which generates SMPTE and FSK and sends it out the supplied cable (which also has an additional MIDI Out). On the Amiga, **MicroIllusions** introduced their MusicX ($299) an integrated sequencer and user-definable MIDI librarian. New from **MIDIMusic** is the inexpensive MIDIMusic MIDI System ($149), which combines a basic 64-track sequencing package and plug-in MIDI interface for the IBM PC.

Composition and Performance

ALGORITHMIC COMPOSITION, ONCE the playground of universities and the odd hobbyist, got an even stronger foothold at NAMM. Leading the charge was **Intelligent Music**, who unveiled and/or announced several more of their unique compositional programs. MIDIDraw ($95) for the ST, will create music based on drawings you create with the mouse inside the program's drawing window. RealTime (no price yet), which is also for the ST, is an interactive sequencer with some of the same functions found in the company's UpBeat, M and Jam Factory programs for the Mac. Speaking of M, the company also announced that they were developing a version for the Commodore Amiga. Finally, Riff and OvalTune (no prices yet), both of which run on the Mac, combine graphics with algorithmic composition-like music making features.

Hybrid Arts also announced a new real-time composition package called Ludwig ($149) which runs on the ST. In addition to creating musical lines based on preset or programmed scales, Ludwig will take sequences or patterns and create variations on them based on user-defined parameters. Ludwig can also record its performances and transfer its files to any of Hybrid's sequencing programs. **Dr. T's** also showed TuneSmith ($149), an algorithmic composition package that runs under their MPE or works as a stand-alone program on the ST.

Another interesting (and really fun) package for the ST which can create music is Big Band ($295), a French program being distributed in the US by **Imagine Music Group**. Big Band will take a basic melody and create a complete arrangement with one click of the mouse, including chords, drum patterns, countermelodies, etc, in a wide variety of musical styles.

New from **MIDIMouse Music** is MousterPiece ($229), a MIDI controller and compositional tool software program for the ST. Mousterpiece allows you to generate any type of MIDI event, including notes and controllers, by moving the mouse or selecting keys on the computer keyboard. The program also allows you to process various types of controllers and notes through inversions, retrogrades, etc.

Yet another interesting ST package comes from new company, **MusicSoft**. MIDI-Drummer ($149) is a generic pattern and song editor with a large visual display for any drum machine which responds to MIDI note information.

Sounds, Editors, and Miscellaneous

ON THE SOUND producing side of things, **Dr. T's** unveiled SampleMaker ($299), a synthesis package for the ST that allows you to create samples via FM and other synthesis techniques and transfer them to almost any available sampler.

Digidesign announced Sound Designer for the ST ($349) and pointed out that it will be a universal program that should work with almost any available sampler. The new Macintosh version ($395) will do the same and upgrades will be available for registered owners of other earlier single machine

versions, Digi was also showing a considerably further along version of Sound Designer 2.0 for the Macintosh (with multiple-channel window and phase vocoding, among other things) along with a stereo version of their Sound Accelerator card ($995).

New editor/librarians (E/L's) were in absolute abundance at the show. **Opcode** demonstrated new Mac editor/librarians for the MT32 ($150), TX81Z ($175), SPX90 ($125), and REV5 ($125), along with librarians for the KI ($100) and the JL Cooper MSB switch box ($100). They also announced price reductions of $50 on the Studio Plus Two Mac interface ($225) and $100 on Sequencer 2.6 ($150). **Digital Music Services** announced DMPII Pro ($295) mixing, editing and automation software for the new Yamaha DMPII and the Macintosh. Finally, **Resonate** introduced Patchworks (no price yet), a Mac E/L for MIDI patchbays like the JL Cooper MSB+, the MX8 and the KMX MIDI Central.

Over on the ST, **Digital Music Corporation** demo'd a preliminary version of their own patch editor for the MX8 ($79) (IBM and Mac versions are said to be in the works), **Drumware** displayed a KI E/L ($119), **Intelligent Music** introduced Cartographer ($100) an E/L for their Mapper, and **Dr. T's** had E/Ls for the KI and the D110 ($129 each). **Performance MIDI Systems** demo'd Designer Series – Roland MT32 ($125 Canadian). **MusicSoft** unveiled E/L programs (all $129) for the MT32, all four-op DX synths (including the DXII and TX81Z), the D50, Roland Alpha Juno synths, and the KI, and announced that they will be distributing the British Iconix sequencing program (no price yet). **Dr. T's** also announced an MT32 E/L for the Amiga ($149).

IBM owners can look forward to new Master E/L's from **Sound Quest**, including versions for the DXII ($175), DX, TX81Z, MT32, DI0, D50, SQ80 (all $150), and CZ ($125). **Dr. T's** also introduced IBM voice editors for the MT32 and D110 ($129 each) and **MusicSoft** had programs for the SQ80, four-op DX synths, and MT32 (all $129). Finally, the **Imagine Music Group** introduced D110 Master ($99) for Roland's new line of L/A synths.

Educational software is making more of an inroad into the market. Take Note ($79), from **Thinkware** is ear training software for the ST while the similarly priced Keys, also for the ST, is a basic, educational step-time sequencer and composition program from **Dr. T's**. Also catching eyes was **Hip Software's** Harmony Grid (distributed by Resonate; no price yet) for the Macintosh, a combination music theory and mouse performance program.

Closing Comments

IN RETROSPECT, THE Atlanta NAMM show actually offered a good number of interesting little innovations – at least enough to keep us all busy through the summer. To be honest, though, I can't wait for the big innovations that I'm expecting to see next January in Anaheim. Until then . . . ■

MAXI MIDI.

MIDI equipment is the best thing ever to happen to music—if you can use it the right way. Too bad that most equipment owner manuals read like a wiring diagram and are about as tough to follow.

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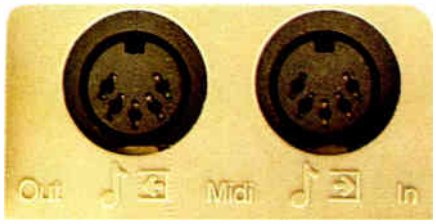
COMPUTER NOTES

ON THE ROAD

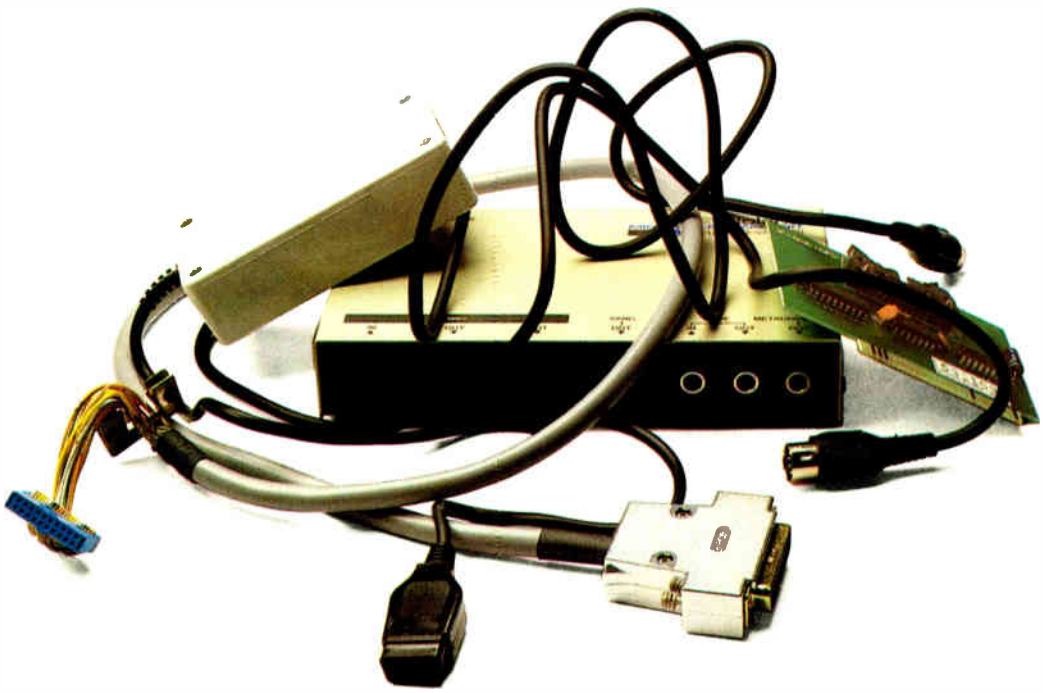
Using Computers Live

REVIEWS

Digidesign Turbosynth
Sonus SuperScore
HMSL



THE DIFFERENCE BETWEEN A COMPUTER THAT MAKES MUSIC, AND ONE THAT MAKES TROUBLE.



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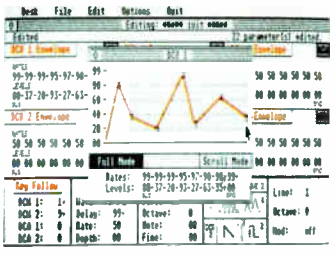

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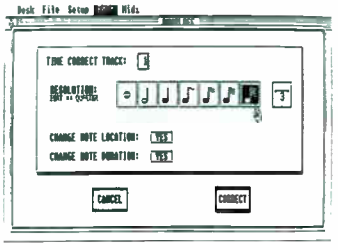

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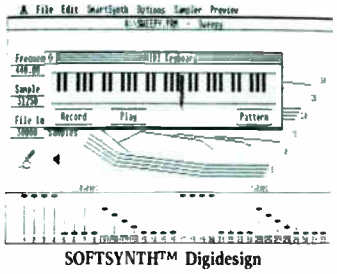

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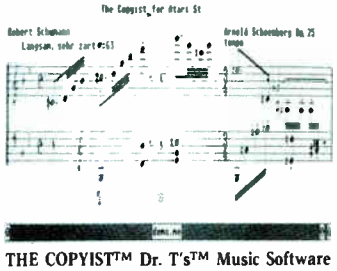

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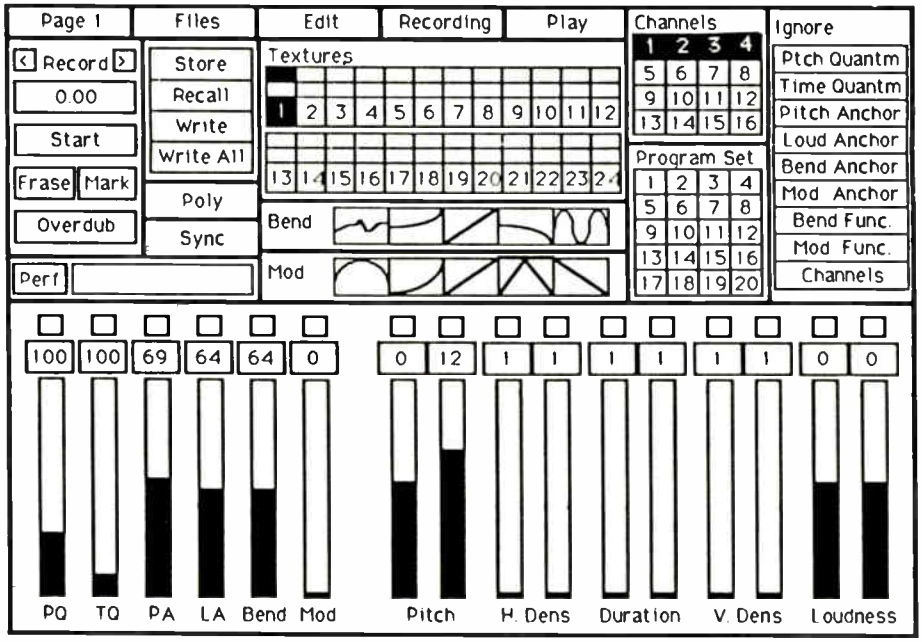
COMPUTER NEWSDESK

Globs o' Sound

Out of CARL (Computer Audio Research Lab) at the University of California/San Diego (UCSD) comes Sound Globs – an "intelligent instrument" program for the IBM that deals in "textural spaces." The user/performer can create textures with particular macroscopic qualities, and then manipulate and orchestrate them live from a screen with icons and software faders. Texture parameters include constants of pitch quantum, time quantum, pitch anchor, loudness anchor, and bend and modulation anchors; range and probability distributions for pitch, horizontal density, duration, vertical density, and loudness; and bend and mod functions for each note. You can control one or multiple textures at once. The performance is saved as a sequence file that's compatible with Cakewalk or Standard MIDI Files. The program is scheduled for release October 1; price will be $150 (a demo program will be available "for a minimal cost").

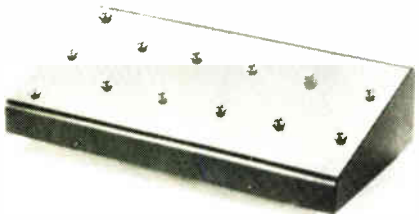
MORE FROM Twelve Tone Systems, PO Box 226, Watertown, MA 02272. Tel: (617) 924-7937.



The Sound Globs Performance page.

NAMM Hardware

In case you thought part of our NAMM report was missing – namely, what new computer hardware was announced – the answer is it merely escaped to Computer Newsdesk. Several companies introduced new computer hardware products, including **Passport** who unveiled the P4001 ($199), a basic MPU401-compatible IBM MIDI interface, and the previously announced MIDI Transport ($395), a combination one in, three out MIDI interface, and SMPTE/MIDI converter for the IBM and the Atari ST. **Computer Music Supply** introduced



Look ma, no hands! CMS' new FT13 foot controller lets you run your software with your feet.

two new IBM-based hardware products: the CMS401FSK ($199) MPU401-compatible interface card, and the FT13 ($199) foot controller. The latter comes bundled with Smart Keys macro software and allows you to control up to 10 different macros (within any IBM sequencing program) with your feet.

Go ahead – treat yourself to Christmas early.

MORE FROM Passport Designs, Inc., 625 Miramontes St., Half Moon Bay, CA 94019. Tel: (415) 726-0280.

Computer Music Supply, 382 N. Lemon Ave., Walnut, CA 91789. Tel: (714) 594-5051.

Two New O's

Opcode Systems has announced a couple of new small releases for the Mac that may pique your interest. If you want to figure out what Opcode software is all about in the first place, they now have a HyperCard stack (Opcode Demo 2.0) that is in essence a software catalog. Actual screen dumps from the products help show how they look. Also, they've completely rewritten the old "Timecode Panel" for their Time Code Machine SMPTE-to-MTC (MIDI Time Code) converter. Running as a deck accessory, the new program will allow you to stripe tape with any SMPTE format, will test incoming timecode, and will show the state of all Timecode Machine DIP switches and the firmware version number. The demo is free to anybody; the new Timecode Panel is free to all Timecode Machine owners.

MORE FROM Opcode Systems, 1024 Hamilton Court, Menlo Park, CA 94025. Tel: (415) 321-9034.

More Potions

Blank Software has just announced version 1.1 of their Alchemy sample editing software (first reviewed here June '88). This version adds support for the Roland S550, Casio FZ1 and the FZ10M. In addition, this release is compatible with Apple's CD-ROM drive (the AppleCD SC) as well as Optical Media's CD-ROM digital sound series. Both access the disks via SCSI. This update is available to registered Alchemy owners "absolutely free of charge."

MORE FROM Blank Software, PO Box 6561, San Francisco, CA 94101. Tel: (415) 863-9224.

Model A

It's an unfortunate fact – if you own an Amiga, you just don't have the choice for software that others have. Dr. T's is making an effort to change that, however, with their release of the MIDI Recording Studio sequencer. Emulating an 8-track tape recorder in function, music may be input from either MIDI or the computer keyboard and played back via MIDI or the Amiga's internal sounds. Individual MIDI events may be edited: cut, copy, paste, transpose, expand, and scale velocity are supported over a definable range (down to a single event), and note timings and durations can be auto-corrected separately. Timing resolution is 384ppqn. Both the mouse and multitasking are supported, and the program (which lists for a scant $69) can be used with the Amiga 500, 1000, or 2000 and "all standard Amiga serial MIDI interfaces." They have a free catalog, so be sure to write for that too.

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

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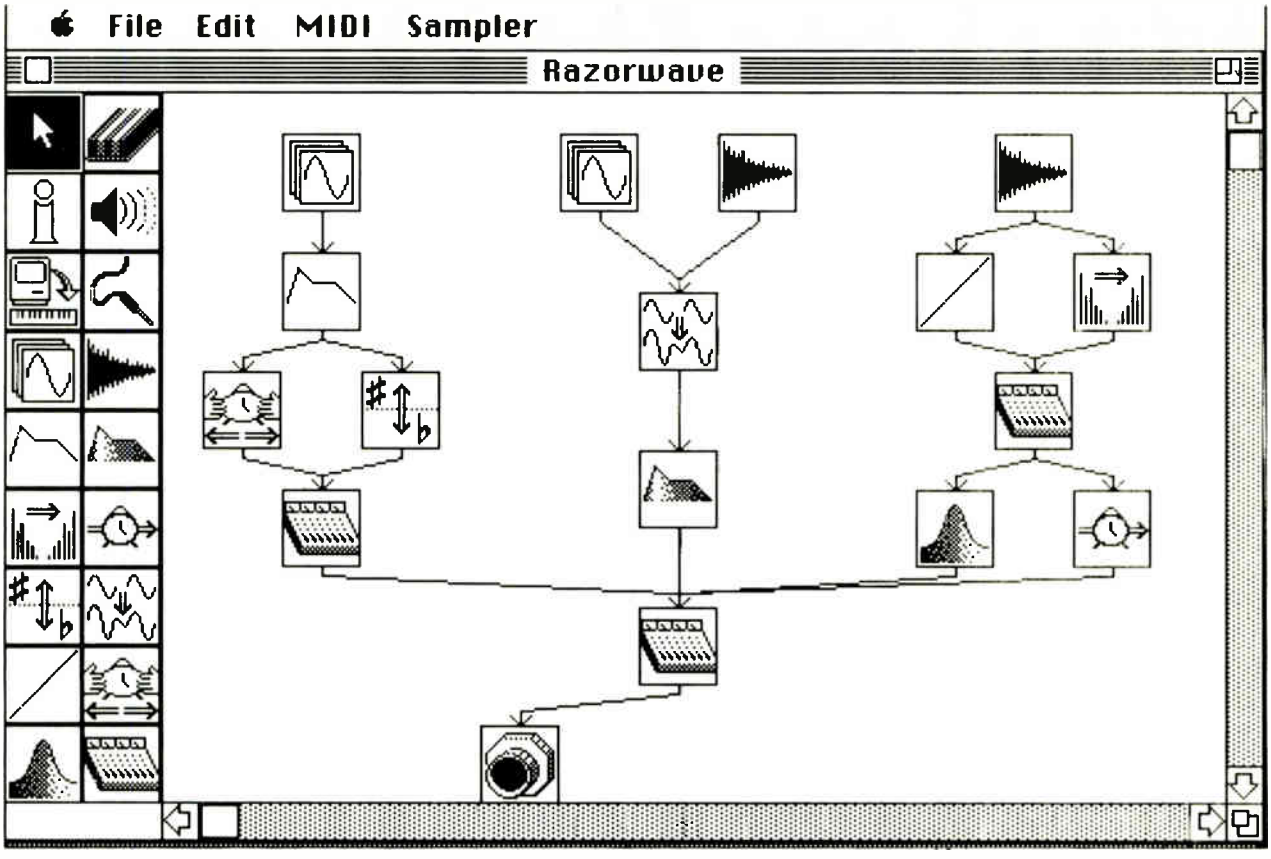
Digidesign Turbosynth

Synthesis Software

With extensive synthesis options and a flexibility that hasn't been seen since the days of modular instruments, this new program for the Mac demonstrates that serious software synthesis is an idea whose time has come.

Review by Robert Rich.



Turbosynth's main patch screen.

I MUST CONFESS, I never really liked the sound of samplers. Images of chipmunks and Darth Vader have poisoned my ears; orchestra hits and shakuhachis; endlessly repeated stuttuttuttering political snippets. Yawn . . . Surely there is more to sampling than this?

Apparently, the folks at Digidesign agree. They rallied to save our jaded ears with a new Macintosh program called Turbosynth. Turbosynth lets you patch together various digital signal processing modules – all in software – to create your own sample generating algorithms. These patches can incorporate digital samples, additive, subtractive, FM and AM synthesis, waveform crossfading, filters, and several different kinds of modifiers. Turbosynth then lets you export the results to your sampler. If you're starting to salivate, read on. It's as cool as it sounds.

REQUIREMENTS

Turbosynth runs on a Mac Plus, SE or Mac II, with a minimum of 1Meg of memory. You'll want at least 2Megs to do anything really ambitious, though. Even some of the example files were too big for my 1Meg SE.

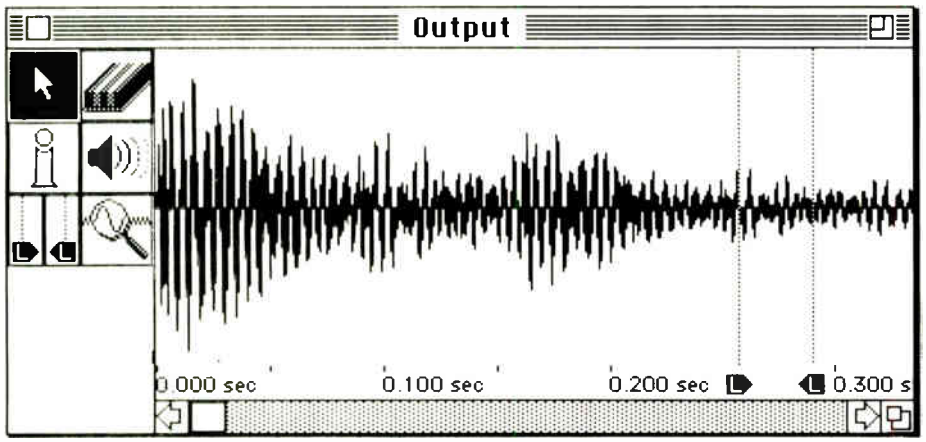
I tested version 1.0 of Turbosynth with no problems on Apple's System 4.2 (Turbosynth is shipped bundled with 5.0). However, the folks at Digidesign said that Turbosynth has some problems running on Apple's new System 6.0 (see accompanying sidebar for more information). Digidesign soon hopes to ship patched versions of Turbosynth for registered users who have the new system.

Turbosynth works with many different samplers. The current list includes: Akai S900, S700, X7000; Ensoniq EPS and Mirage; Roland S50, S550, S10, S220, MKS100; E-mu Emax and EII; Korg DSM1 and DSS1; Casio FZ1; and Yamaha TX16W. Turbosynth also speaks the MMA Sample Dump Standard, adding the Sequential Prophet 2000/2002, Dynacord ADD-one and Simmons SDX to this list. Turbosynth defaults to the most commonly used sample rate for each sampler, but the user can change rates if desired. Unfortunately, Turbosynth cannot send stereo samples. What a shame.

BASICS

For all its complexity, Turbosynth is mercifully easy to understand. It works like a modular synthesizer, with little boxes representing various components in the signal path. The screen shows a large pallet of module icons. You simply click on the icon of the module you want to use, then click on the screen where you want that module to appear. A patchcord icon lets you connect these modules. You can change this "wiring" or move modules around at any time. Always at the end of the chain sits an Output Jack icon, which represents the final sample emanating from the patch.

Before you can hear the sample from the



The sample screen with loop points indicated.

Macintosh speaker or send it to your sampler, you must first tell the Mac to calculate the results of the patch by clicking on a Speaker icon. The Mac takes a few seconds to calculate any new changes in the patch. Another click on the speaker icon will let you hear the sound from the Mac. Turbosynth earns high marks for its speed. It seems that Turbosynth can calculate a complex additive algorithm at least as fast as Digidesign's own Softsynth (probably faster).

Turbosynth works with three kinds of files. Document files store the Turbosynth "patches," including module parameters, interconnections, and sample data desired. These document files are very compact if they contain no samples. Turbosynth can also read and save sound files in the Sound Designer format, and waveform files containing sound data representing regular repeating waveforms. You can create waveform files from samples within Turbosynth by opening a Sound file, creating a small loop, then saving that loop as a waveform (or you can draw your own waveforms by hand.) Turbosynth can then use these sampled waveforms in its oscillators.

MODULES

Turbosynth contains thirteen DSP modules, counting the Output Jack (which automatically appears at the bottom of each new patch screen). The other twelve icons sit in a pallet at the left of the screen.

The program lets you cut, copy, paste, delete, or move modules around, just like MacDraw or any other object-oriented graphics program. Double-clicking on one of the modules opens up a window showing the parameters for that module, with an appropriate assortment of tools. These windows vary with each module. The simplest windows show sliders, letting you adjust certain basic parameters. Other windows provide highly flexible envelopes similar to those found in Softsynth, while modules containing wavesamples provide wave editing environments similar to Sound Designer.

Every module window contains a speaker icon, letting you hear a signal's development anywhere along its path. Also common to every module is an info box, accessible by clicking on an "i" icon. All module info boxes let you name the module, and also let you change the sample length (in number of sample points) and sample rate (in Hertz). A similar info box in the main patch window lets you set default values for the length and rate, but you can specify different values for each box if desired. Also in each module info box, a "normalize" button adjusts the signal level so that the highest peak will be as loud as possible without clipping, thus improving the signal-to-noise ratio. Two other buttons in the info box select the module's playback rate from the Mac's speaker. Choosing the button for the Mac's rate will let you hear the module's output at highest fidelity on the Mac, but will change the pitch if the module's sampling rate is set to something other than the Mac's 22kHz.

Here's a brief description of each module:

• Oscillator Module

Turbosynth distinguishes between Sample modules and Oscillator modules. While Sample modules play sample files, Oscillators work only with repeating waveforms. But the oscillators in Turbosynth can do a lot more than just generate static waveforms.

The Oscillator window consists of a timeline representing the length of a sample. You build sounds by pasting waveforms onto this timeline. The oscillator automatically crossfades between these waveforms, allowing for very natural sounding timbral changes. You can select one of 15 preset waveforms by clicking on its icon, then clicking on the point in the timeline where you want it to play. Presets include sine, triangle, sawtooth and square waves, some harmonic multiples of the above, and a few more complex waveforms. If you want to use something more unusual, you can open previously stored waveform files and select from those. Each waveform icon shows a little picture of the wave it represents, making everything blissfully easy to figure out.

To edit a waveform, just double-click on its icon in the timeline. A window opens showing a single cycle of the selected waveform and a number of editing tools. The same 15 preset waves appear here as before, along with the icon for opening waveform files. A pencil icon lets you draw waveforms by hand, and ten waveform modifying tools let you alter them globally. These icons adjust the amplitude of different parts of the wave (or induce clipping if desired), skew the wave left or right, add randomness (noise), or smooth the wave (reducing overtones). Considering how hard it is to hand-draw a waveform, I think the smoothing tool is a particularly wonderful addition to this set of modifiers.

Turbosynth lets you name custom waveforms and save them as waveform files by selecting "Save waveform" from the Mac's file menu. Any waveforms you create can thus be used in any Turbosynth patch.

• Sample Module

Data from any of the supported samplers can be imported into a Sample module, as well as any Sound Designer file. Turbosynth can also create samples from any other module or group of interconnected modules in a patch. You can save any sample as a Sound Designer file, which means that you can further tweak these sounds in Sound Designer or Blank Software's Alchemy.

The Sample module window shows a waveform representation of the sample, with a tool pallette including an eraser and loop icons. The eraser can only erase loop points. There is no pencil tool, nor can you create crossfade loops. The loop icons select the start and end points of a loop, which you can then fine-tune in the Loop Window. The Loop Window shows the adjacent start and end points of a loop, with the end portion on the left and the start portion on the right. To move the loop points to a particular point in the wave, just click on the desired point and the loop point will jump there. Scroll bars let you nudge the loop points to remove glitches. Vertical and horizontal zoom buttons let you scale the amplitude and frequency resolution of the loop editing display, but do not affect the sound.

Once you have found a loop that sounds good, you can save it as a waveform file for future use in an oscillator, or you can fill out the remainder of the Sample module's buffer with copies of the loop. Turbosynth will not otherwise use the loop. In other words, a Sample module represents a one-shot sample – the loop within a Sample module is not the same as the loop used to sustain a note in your sampler. This latter loop is created in the Output Jack module. This difference allows many Sample modules to play at once, each with different loop points.

As you may have noticed, the Sample module provides only the basics for editing. I would complain about its inadequacies if I felt they really mattered, but these limitations get lost in the overall flexibility of Turbosynth.

• Amplifier and Filter Envelope Modules

These modules do exactly what you would expect, acting like analog VCAs and VCFs, with envelopes built into each module. Unlike their analog forefathers these envelopes have unlimited segments. Both modules provide several preset envelopes, which you can bend around with the mouse by clicking on any point and moving it. Four envelope modifier buttons create global changes by altering sustain levels or skewing the envelope parameters in time. ►

► Unfortunately, I could find no way to copy envelopes between the two kinds of modules.

The filter module provides only low-pass filtering, which seems unnecessarily limiting. I would love to see band-pass filtering and resonance. The filter offers a cut-off slope of 6db/octave, which can be increased by daisy-chaining several copies of the same module. The cut-off frequency only goes down to about 500Hz, but this is generally sufficient. When chained together these filters sound quite good – very "analog."

C O M P U T E R

• **Spectral Inverter and Waveshaper**

The effects of these two modules are a bit harder to describe. The Spectral Inverter module inverts the harmonics of a waveform around the halfway point of its bandwidth. If the sample rate is 20kHz, the bandwidth will be roughly 10kHz, and the halfway point is 5kHz: the spectral inverter would then move an overtone at 6kHz to 4kHz, or an overtone at 1kHz to 9kHz. In practical terms, this will make most waveforms sound hollow or metallic. It generally works best for very bright sounding waveforms, where the results can be quite pleasant and unusual.

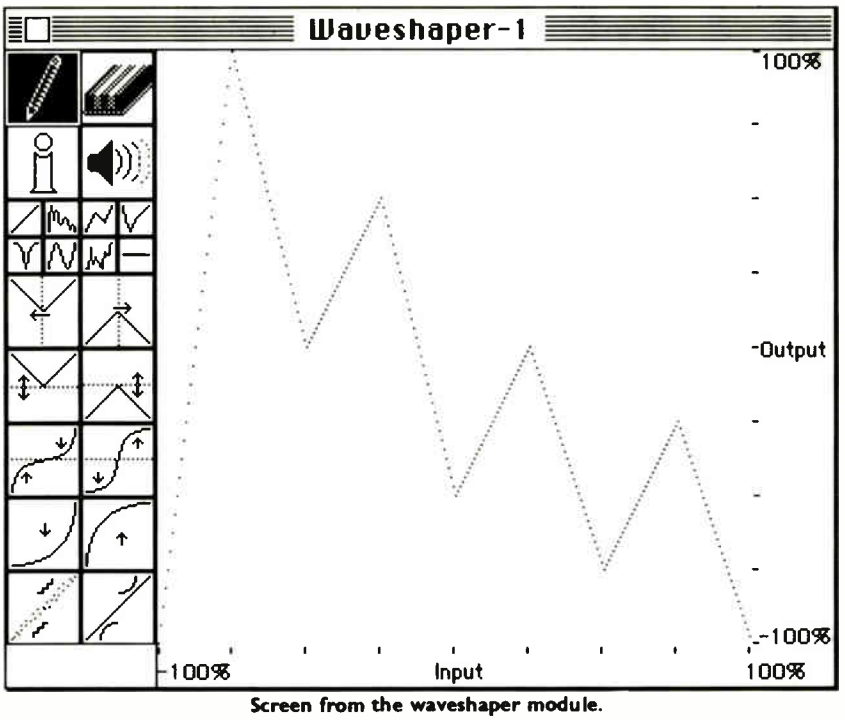
The Spectral Inverter window provides an envelope just like the amplitude and filter envelopes, only this envelope controls the mix between normal and inverted signals. Nice touch.

Waveshaping (a technique used a good deal in Buchla's digital synthesizers) is one of the more interesting modules in Turbosynth. This module provides a transfer function which maps the amplitude of incoming sample points onto different amplitudes according to a graph which you can edit. This graph shows the input and output levels on horizontal and vertical axes, with its range covering the dynamic range of the sample. For example, if the graph shows a straight line from lower left to upper right corners, it will leave the signal unaltered because every input level will map to its identical output level. If a line goes from upper left to lower right, it will invert the signal by 180 degrees. If the transfer function is bumpy, then it will add lots of overtones to the signal. More complex curves get pretty hard to predict. The editing window provides several shaping tools, preset curves and a pencil to draw your own functions.

• **Modulator and more . . .**

The Modulator module lets you create more than just the FM sounds made famous by Yamaha. Any two signal inputs can modulate each other, including samples from the outputs of any Turbosynth module. Two sliders provide the only controls for this module: one for modulation amount, the other for mixing modulated with dry signals. One input can modulate either the frequency (FM) or the amplitude (AM) of the other. AM synthesis sounds different from FM, and it is a refreshing addition to Turbosynth. The Modulator module also allows pitch modulation, which is essentially FM optimized for chorusing and vibrato effects.

The **Delay** module simply creates echoes out of the signal. Controls include coarse and fine delay time, feedback, polarity inversion and wet/dry mix. The one thing missing is a frequency sweep for flanging effects.



Screen from the waveshaper module.

Similar to the delay module, the **Resonator** module works with short delay times to create ringing delays. Instead of delay time, its slider controls show resonant frequency (again, no sweeps).

The **Pitch Shifter** module transposes sounds by using linear interpolation (thereby extending or shrinking their duration proportional to the amount of shifting). It can range over +/−2 octaves, with slider controls for pitch and wet/dry mix.

The **Stretcher** module increases the length of a signal by scanning through it with a loop of definable length, copying the segments within that loop as it goes along. This only sounds natural with harmonic waveforms where you know the fundamental frequency. Otherwise it induces some interesting distortions. Its sliders control coarse and fine loop frequency, predelay before stretching occurs, and the number of times each segment gets duplicated.

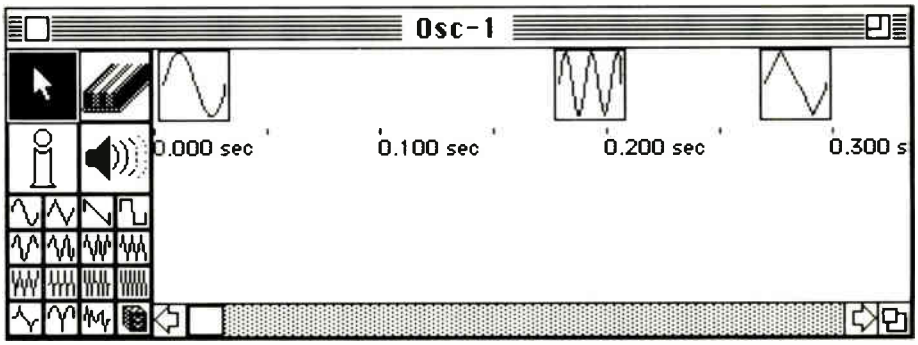
The **Mixer** module contains as many sliders as there are modules entering the mixer. Since most of the modules described above can only process one input, the mixer serves to merge many sources into single module inputs.

Last of all, the **Output** module has a window much like the Sample module, only here the loop point really is the sample loop. You can create both sustain and release loops, but the distinction only matters with the Prophet 2000.

CONCLUSIONS

The folks at Digidesign said that Turbosynth is a fairly open-ended program, and they hope to add modules in the future. Indeed, I have a short wish list of features I would love to see. Pitch envelopes would really help, although the Modulator module lets you work around this (draw your envelope as a waveform in an oscillator, set it to a very low frequency, then use it as an FM modulator). I would also love to see a module that "convolves" two samples like the Emax SE. The Filter module could be beefed up, and the Delay and Resonance modules could incorporate frequency sweeps. An envelope copying function would also prove helpful. Since one of the main philosophies of Turbosynth was having processing modules that could calculate quickly (and not keep the user waiting), we'll have to see what all they can add before compromising that goal.

As a final comment, I should mention that synthesis for a sampler is not the same as on a ►



The Oscillator window lets you piece together waveforms to create a sound.

The "Bugs" in System 6.0

Apple has been evolving the operating system for the Macintosh since Day One. Rewriting the system while keeping older software compatible can only be done one way: by guaranteeing how outside software tries to use the system. To do this, Apple specifies to their developers how to interface with the system. By doing so, they can keep consistent the layer that programs "see," while changing anything they like inside the system itself.

The problem comes when software developers attempt to use the System in undocumented or unrecommended ways. Such a problem has come up with the Mac II sound driver portion of System 6.0. Apple suggested in their documentation that the sound driver not be accessed a certain way. *Violating* Apple's suggestion caused no problems up through System 5.0 – but it did in 6.0. Some developers will assert that Apple was ambiguous in their warning; either way, they will have to patch their software to work with System 6.0 and beyond. There *are* some small bugs in 6.0 that Apple would not elaborate on aside from saying that "there's nothing catastrophic," and that the vast majority of existing software runs with 6.0 just fine.

The other problem Mac users have with System 6.0 is that it has evolved to the point where it needs 1Meg of RAM to run, and will not run on older 512 or 512e machines (even those with non-Apple memory extensions up to 1Meg). As Apple says, "We told them it wouldn't work, and lo and behold, it doesn't." ■ *Report by Chris Meyer*

► synthesizer. Any time-variant synth effects will suffer the same problems that plague acoustic samples. Envelopes will expand and contract across the keyboard. Chorusing will speed up as you move up the keyboard and will also make looping much harder. By creating certain effects within the sampler rather than in Turbosynth, you can avoid many of these problems. Nevertheless, Turbosynth does not overcome the inherent shortcomings of the sampling process.

Despite my criticisms, I can't really complain about Turbosynth. I really like this program. I had so much fun working with it that I could scarcely tear myself away to write this review. Like any good tool, Turbosynth is easy to understand, yet it has enough depth for years of exploration. It can create a huge variety of sounds, from simple to complex, ugly to beautiful. If you have the equipment and you want new sounds, take a serious look at this program. It's not cheap, but I think it's worth it. I can't wait to start patching again . . . ■

PRICE $349

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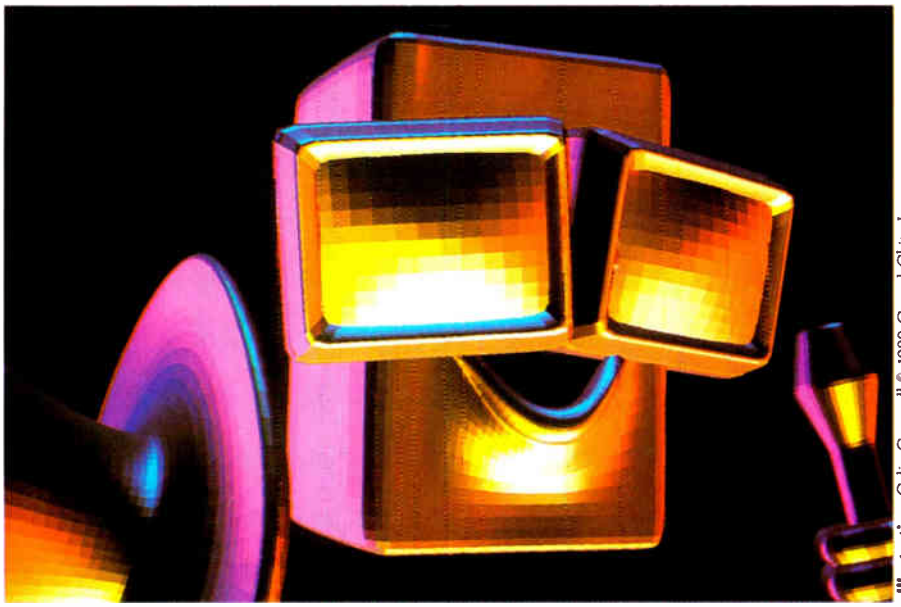


Illustration Colin Cantwell © 1988 Crystal Chip, Inc.

Using computers in a live performance can add flexibility and complexity to your set . . . really!

Text by Greg Whelchel.

IF YOU'VE MADE any recent excursions into the local "live music" scene (if you can still find one these days), you may have noticed the rather out-of-place glare of a computer monitor on stage. If not, there's no doubt you will . . . sooner or later. The use of a computer on stage is becoming more and more common-place as computers become an integral, if not expected, component of a keyboardist's basic setup. Using a computer in a live situation will not only help fill out the "missing parts" as well as make patch data transfer a breeze; used in a musical, creative fashion, it can take you to places you could not go without the aid of this marvelous machine. It can also save you a fortune during the mandatory break-time-video-game-diversion!

OKAY, WHICH COMPUTER?

Making a decision as to what type of computer and software to use for a live show requires special consideration. You'll have to make a choice about operating systems. Do you prefer the no-nonsense, cursor-oriented IBM look, or the point and click, mouse-oriented Mac, Atari ST and Amiga systems? For playback, I can't see that it would make that much difference. In fact, where to put the mouse can be a little bit of a problem at times. The line starts to blur, anyway – mouse-oriented graphics shells like Windows, Presentation Manager, and DesqView are all available for the new IBMs and compatibles; so the prospects for more Mac-style programs on the IBM should prove promising for the future.

Of more importance than the computer is the software it runs. It might be a better idea to take a look at the different software packages available and buy the computer that runs the program you find best suited to your purposes.

Another criteria to ponder is size and portability. Let's face it, most computers were designed to be set up once and stay where they were put. Leave it to musicians to want to drag the things all over the country (especially to bars). There are quite a few laptop IBM compatibles out there, but most lack the necessary expansion slots, etc, required for the MPU-401 or similar MIDI interface. The recently announced WonUnder (see MT August '88 Computer Newsdesk) can solve that problem and Yamaha's new CI MS-DOS computer makes the whole point moot. Along those lines, I just heard two fairly substantiated rumors today – that both Atari and Apple may be releasing laptops of their own quite soon. Hooray! Having one of these babies might just be the ticket for most people. You could just carry it on the plane or bus or whatever and not have to worry too much about it. Easy to take to the room for some late night computing, too.

Surprisingly, there are several full-blown rack-mount PCs to be found, and even a rack-mountable Mac Plus or Mac II and retrofit rack mod kits available from Julian Systems of Concord, CA. Current Music Technology of Malvern, PA also makes a Mac Plus in a rack or will retrofit an existing Mac and throw in Opcode's Studio Plus Two MIDI interface and a 20Meg hard disk as well. If your computer is used for predominantly music-related applications, a rackmount might be a sensible solution to the 'Where do I put the monitor?' dilemma and the resultant 'Who the devil tripped over the cords?' mishaps you might wish to avoid; although the fact of rack-mounting may make it inconvenient for other uses. Being that these guys run anywhere from $1200-$6000, that's an inconvenience many of us just can't afford.

WRAPPED IN SWADDLING CLOTHES

This brings up one of those highly necessary, yet all too often disregarded, elements of preventative gig medicine: cases. True, they cost a pretty penny; sometimes more than their contents. And, of course, we'd all rather be drooling over that endless toy list, trying to decide which contraption to get next, than buying a case for some old outdated thing we bought two weeks ago. But computers are frail creatures and need to be protected from us humans and our quirks. Besides, you've got better things to do than trying to recall, last minute, all those parts you sequenced a year and a half ago and immediately forgot, now that the 'puter isn't cooperating.

No matter what type of computer you end up with, it will be money well spent to keep it working. I've sent an Atari ST all over the world in ATA cases (even forgot to park the Astra hard drive, once, on a flight back from Paris,

with no ill side-effects . . . gasp!!) and have had very few, only minor problems, like loose connectors and nuts, or cracked plastic pieces.

POWER AND TIME

How much computing power do you need? Do you just want to do no-frills sequencing, or do you need to get in there and edit, shift, modify, transmute, or maybe even deracinate or extirpate each and every little MIDI event? Once again, that's in the realm of software. Most any computer on the market is plenty powerful enough to do anything you could imagine. The question should actually be, "What kind of features do I need?" Using a computer live is somewhat different than at home or in the studio. You probably won't be starting and stopping too much. Hopefully, you hit "go" and don't stop until it's over. All those nifty editing and mapping features are great for the recording side of sequencing, but showtime playback requires a whole different set of capabilities.

One of the main problems to consider is song load time. More than about three seconds of silence is way too long, as anyone who has ever performed on stage will vouch. There are several ways around this. The easiest (and most expensive) solution is to buy a hard drive. The average song loads from hard disk in about a second. But, it's just one more delicate device to cart around and protect from all the well-meaning, but completely oblivious beverage-spillers.

A few programs, ie. C-Lab's Creator and MidiSoft Studio Advanced Edition for the Atari ST, and Texture Live by Roger Powell for the Amiga and IBM/clones (which requires Texture 2.n to create the sequences), allow disk activities while the sequencer is running. You can actually load a song to execute while another one is playing. I wouldn't recommend, as a normal operating procedure, saving files to disk while running a sequence, however. You can never tell what might happen, no matter what the manual says! Other software options include Sonus' Masterpiece for the ST and Cakewalk Live from 12-Tone Systems which allow multiple songs to be loaded into memory at once so that you can immediately access any of them. A similar idea is used in the new Pro MIDI Player software for the ST from Performance MIDI Systems; it's a sequencer playback program with no editing features.

If you have sufficient memory, a RAM disk for the song files works quite well. (For the beginner: a RAM disk is a small program that reserves a definable portion of memory and sets it up to store files in the same fashion as data is stored on a disk drive. The computer is then "fooled" into seeing that memory area as an additional disk drive. Loading from a RAM disk is extremely fast because the data is already in memory and, really, only has to be copied from one area of memory to another.) The drawbacks are that the capacities of both the RAM disk and the sequencer data areas are severely limited; and if power fails, for some reason, you've got to recopy all those files into the RAM disk all over again.

Something along the same lines, but less volatile and without the same memory consumption problem, is a battery-backed-up RAM cartridge. They are available for most computers now, in sizes plenty large enough to hold a whole set's worth of tunes. The advantages are: 1) the files are protected in case of power failure, 2) they can even be temporarily stored in the cartridge, as long as the battery remains good, and 3) the usual amount of free memory is available for the sequencer and any other accessories or programs you may wish to have resident and switch in and out of. (We'll get to the exciting prospects of switchers later on!)

Another solution is to load several songs into memory at one time. If your sequencer allows importing individual tracks, you can load more than one song into the same sequence, and then save that as one composite song file. Some programs, like Hybrid Arts' SMPTETrack ST, also have programmable mute and tempo registers that make it possible to both select the tracks comprising each song and set the tempo, with a single keystroke or mouse click. By mixing all tracks of a song onto one track and making a composite song sequence by importing each song-track, I've been able to have all the music for an entire set in memory at once. If your song is looped, this may be somewhat of a problem. SMPTETrack has a really great feature called 'Flatten Chain' that basically converts a looped song into a single-track linear sequence, by automatically duplicating and assembling the selected sections. All data and channelization are retained, but are now compiled to one linear track. This makes putting together a whole "sequence" of songs quite effortless. If your sequencing package allows multiple MIDI channels on one track and you can get two computers in the room at the same time, you can do the same thing by playing one into the other.

If you have a limited number of tracks and they are linear (not patterns or looped), another way to do this is to put several songs back to back, sharing the same tracks, with a little dead space between the end and start times. You then just store, if possible, unique locate and stop times for each song. It takes more work to set things up this way, but it's real fast and convenient once it's done. If you change your repertoire or song order often, however, I would recommend the prior method of putting each song on a separate track as the better choice.

What are some other features in a sequencer that would make live usage more effective? How about remote control? There are many sequencer programs that allow remote sequencer control from a keyboard or controller without having to use a separate mapper to send MIDI start and stop commands. How about MIDI control over repeat or vamp sections? Assign an unused MIDI switch (like yes/no, portamento or sostenuto) to hold on one section until pressed again. If you don't have a mapper, some programs allow real-time mapping and/or adjustment of controllers, notes, switches, velocity, etc, during playback. The computer can easily map your incoming parts and merge them with the sequenced parts to be output together.

Not all programs have these features. Examine each program thoroughly before buying and give them a chance before making harsh judgements. It takes time to get to know a program before you can really know its strengths and weaknesses and be effective with it. If it was too easy to learn, it probably won't have the degree of sophistication you might need. Look how long it took to learn how to play.

SWITCHING KEYS

With the typically larger amounts of free RAM available in today's machines, one upshot of the inevitable question "What in the heck do I do with all this memory?" is the switcher. There are many programs in this realm to choose from. Dr. T's has the Multi Program Environment (MPE) for the ST, Hybrid Arts has HybriSwitch for the ST, there's Multi Finder for the Mac, module-style programs for the Amiga, and several "SideKick" style hot-key programs for the IBM. Be they accessories, switchers or multi-tasking kernels like Multi Finder, the basic intent is the same: to be able to have several programs resident in memory at one time, so you can quickly move between them without disk access or the continual loss of data every time you quit a program. This is particularly useful if there are facilities for data sharing (sending a sequencer file directly to a scoring program, for example).

Because of the nature of the beast, not all programs are compatible with any or all switchers. Generally, it is limited to running programs developed by the same company or others that implement the same procedures to switch. You can pretty much be assured that all or most of a particular company's programs will run successfully under their own switcher; but, try before you buy, just to make sure. As companies release their developer kits to more outside programmers, there may be more prevalent implementation of a particular method, but as it stands now, there are about as many switchers as there are developers.

Aside from all that, however, what a switcher can allow you to do is a dream come true. With sufficient memory, you can have, for instance, two librarians, a sequencer and an algorithmic composition tool (as they call them) in memory at once. Load up the synths with the patches for the show. Before the rest of the band gets there, quickly spice up a rhythm part you find a little drab with 'M' from Intelligent Music or Dr. T's KCSII with PVG (that's Keyboard Controlled Sequencer II & Programmable Variations Generator, whew!). And, then load the set's worth of tunes, and you're ready.

If that sounds great, imagine how useful it would be on the recording side. How many times have you been in the middle of sequencing something and you've got to hear that part with this other patch, except it's not in the synth? Or you need to write out a part real quick for the ▶

guitar player. You could play it into the sequencer, switch into a music scoring program, and print it out, without having to quit the sequencer. Now, anyone who has messed with scoring programs knows how temperamental they can be, but for a quick and dirty, fairly straightforward part this can be real fast once you know the ins and outs of a particular program.

COMPUTER

FEIGNING HUMANNESS

With all this fantastic technology to become immersed in, it's sometimes necessary to remind ourselves that what we're really trying to do here is play music. I was recently in Toronto, in a lounge that shall remain nameless, because I don't remember the name. There was a four-piece band on stage with a Mac running Performer from Mark of the Unicorn. I thought, "Oh, this might be interesting." It wasn't. In fact, it wasn't much of anything. Everything was so mechanical and stiff. The keyboard player had sequenced so many of the parts, there was nothing left for him or the drummer to do. All this automation is fine and dandy, but what good is a live show with nobody up there playing?

If you're sequencing live, the ultimate thing you can do is to do it so well, so musical, that the thought that you might possibly be sequencing never enters anyone's mind. It's got to still communicate and be emotional. Is it a tool, or a crutch? How do you view it? Do you use it to help you communicate, or do you think it's supposed to do that on its own?

How do you get a sequence to not sound so mechanical? First off, don't sequence anything unless you absolutely have to. If you can play it, play it! That's what live music is all about. Try to sequence only repetitive parts, keeping the solo or lead parts and more fluid parts that require interaction and subtlety for yourself. Also, whenever possible, don't quantize. Make three or four attempts at a line, or punch in. The results are usually much more musical, unless you happen to be trying to achieve a very robotic, mechanical sound intentionally.

Try playing linearly all the way through a song instead of looping the same sections over and over. Record several different takes of a part that may be slightly different. You can choose which track to use from night to night if you get tired of one. Some sequencers have a 'humanize' feature that can actually give a rather stiff part some life. I wouldn't have believed it until I tried it for myself.

There are also several programs that implement variation algorithms that create somewhat controlled variations on what you've played. The results can, at times, be quite amazing . . . even usable. And, then again, sometimes not. Three that I know of are Intelligent Music's M for the Mac and ST (and soon for the Amiga), M/pc from Voyetra for the IBM, and Dr. T's KCSII with PVG (which I mentioned earlier). Give them a try to help alleviate that "same old lick" problem or just to throw in a new, wild hair here and there.

If it just doesn't feel right, fix it. Play it again and again until it does. Or, if you're syncing to a drum machine, offset tracks one way or the other a few ticks until things gel. This makes an amazing difference. If the sequencer you choose doesn't allow time shifting on an individual track basis, you may never experience what it feels like for a sequencer to really lay a serious groove.

Experiment. Most people I know don't use but about 10% of the capabilities of their sequencers. Go through each menu item and experiment until you really understand what each and every function does and why. Then, you'll be able to "play" it like your instruments. Sounds funny, but it's not that far fetched. You'll find ways to do things that you supposedly can't do with that software. In fact, I would welcome any input in the way of helpful tips and undocumented features that anyone may have discovered for their particular sequencer setup. (Maybe we could do a reader/sequencing tips/article every so often if there were enough people interested.)

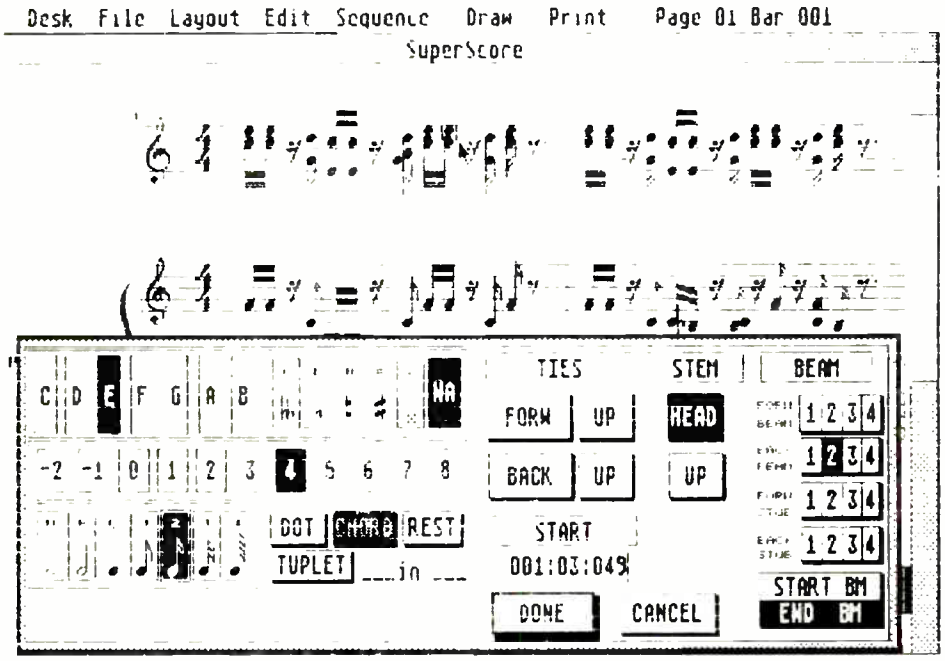
Above and beyond any of the fancy tricks and astonishing feats of our new technology, is our ability to conceive and appreciate music. Let's use this stuff to make beautiful music and make a contribution to life. Why? Because we can. ■

In addition to being a keyboard player with the Pointer Sisters for the last five years, Greg Whelchel plays on numerous TV and jingle sessions and once served as musical director for the Hughes Corporation.

Sonus SuperScore

Music Notation Software

Combining notation capabilities with a built-in sequencer, this Atari ST package may help answer the question, to score by software, or not to score by software. *Review by Chris Many.*



Main notation screen for SuperScore.

THE BASIC CONCEPT with which a musician, or anyone for that matter, must approach the computerization of his or her craft is simply stated: will it make my work easier? Does the computer hardware/software symbiosis created by others allow for greater freedom of creativity, or at least reduce the effort required to make music? By providing a service facility to the artist, computer assistance in the field of music has changed forever the way professional and amateur musicians alike create their art. And yet there is still plenty of ground for software authors and companies to cover.

For example, take the subject of music notation and scoring programs. Over the last year, between 10 and 20 packages have been released for a variety of computer systems, all promising to rid the composer/musician of the trials and tribulations of putting one's music to paper. Each package contains the basics, and provides their own strengths (and weaknesses). But despite the proliferation, no one package has established itself as a standard. This means the market is still wide open for any contender, large or small, to create this standard. Consequently, we'll keep seeing a lot of scoring programs until the wheat has separated from the chaff, so to speak.

Which brings us to this review: SuperScore from Sonus. Not only is this a scoring and music notation program, but a fairly complete sequencing module as well. Requiring 1Megabyte of RAM and a monochrome monitor, this offering is currently available for the Atari ST series of computers. Although the current version (1.2) does not support laser printer output, the next update will (as well as color monitors).

SEQUENCER INPUT

Having read the advertisements running in the music mags, one of the features that initially attracted me most about SuperScore was the conversion routines available not only for Sonus sequences, but files made using other companies sequencing products; ie. Dr. T's, Hybrid Arts, and Steinberg, to name a few. Sonus, thankfully, has incorporated a conversion program that will allow you to use your favorite ST sequencer, load the file into the converter and create a file which SuperScore can work with. Unfortunately, the conversion program in v1.2 only converts Steinberg sequences at this time. All other major software formats will be supported in the near, if not immediate future, including Standard MIDI Files. Supporting everybody's sequencer format is no easy task for anybody; the "Standard MIDI Files" spec is finally on the verge of being upgraded to *official* 1.0 status, which will hopefully make life easier. We can only hope every score program supports it – the current truth of the matter is that compatibility across different manufacturers' programs is a spotty thing (buyer beware).

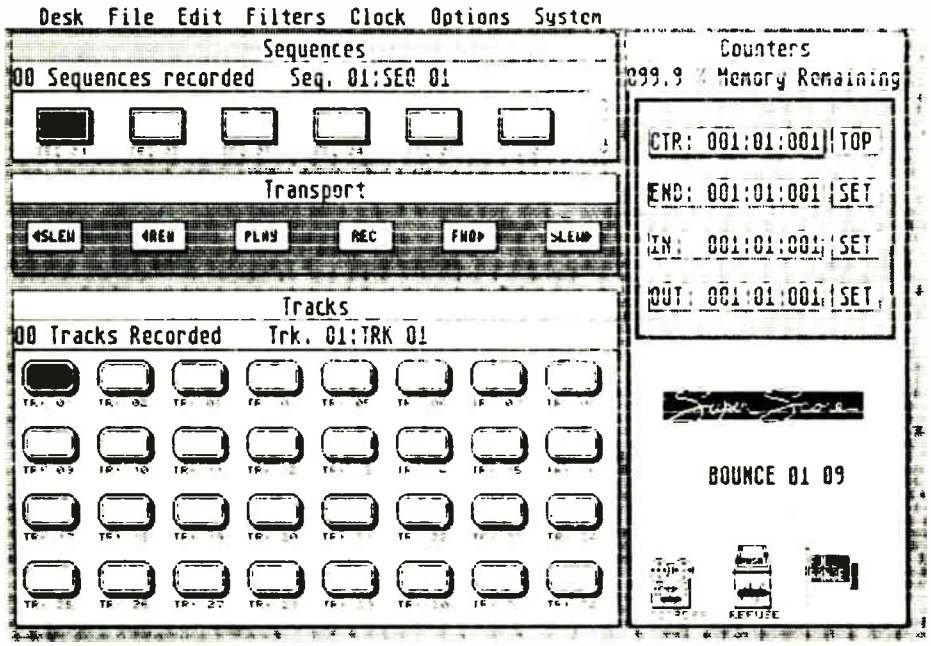
The fact that the program has paid a good deal of attention to sequencer files is this: the only way to enter music notation into SuperScore is from a sequence file. This doesn't mean you can't edit your music, or any such thing, but it does mean that this is truly a MIDI musician's product. There's no blank staves to confront, or entering notes or bars from the keyboard (in text format or by pointing, clicking and/or dragging symbols) without *first* having created the music you want to print out using a sequencer product of some sort. This is quite a reverse from some other scoring programs which ignore MIDI altogether – here's one that requires a usage of MIDI, no ifs, ands or buts. Of course this also means SuperScore can play its finished scores out over MIDI (with unlimited polyphony per staff).

TAKING INK TO QUILL

The scoring portion of the program is straightforward and easy to use. Before you load in your sequence, you'd be wise to set up the format of the score, although this can be done at any time. There are seven score layouts built in; Piano w/Vocal, Piano, Solo, Duet, Trio, Quartet and Choir. The appropriate labels are included with them, or left blank in the case of the four generic formats. You're not bound to these formats, though; included on the disk is a format library which allows you to load a variety of other score formats. String Quartet, Big Band or Orchestra are just a few of the formats contained in the library, each one correctly labeled with instruments, with staves joined as necessary for each instrument grouping. If any of the library formats doesn't work, you can set up your own and label it how you want, up to 32 staves. So if you've got a quintet of Flute, Bassoon, Trumpet, DX7 and Percussion, you can create a format that reflects your needs. You can edit the format library if you wish, but it will require a word processor to enter it and change the ASCII text to meet your needs.

Next you'll want to set the size of the score to Small, Medium or Large. Small will cram more of the score on the screen at a time; Large will generate large notes and staves. Medium is a comfortable setting in between, but it's just a matter of how you want to interface visually

with the program. If you're setting up your own score you can also select specific clef symbols for each line of the staff as well as individual labels for each line. An Instrument library is included, so you can either step through a lot of choices



The sequencer screen from SuperScore.

to find the instrument you want, or just type it in.

Next, set the key signature you'll be working with. You can change this at any measure in the score, so you're not bound to the first one you select. The same goes for meter settings. Because you can set multiple keys and meters, you'll need to press a Set button for each one, and then exit the routine when you're complete. Bar numbers can be displayed at a determined frequency (every bar, every five, every ten, never, etc).

You'll also want to enter the score titles, which include the Main Title, Subtitle, Composer and Lyricist credits, Tempo and Expression markings. The fonts for each of these sections are built in and can't be altered, but that's OK as the ones chosen are appropriate.

Once these basics are set up you'll be ready to load your score in, but for the most part you're not particularly limited about whether you do your setup first or after. It's just that when you do it after you load your score in, the screen will redraw the whole score after you input each of the setup changes, which can make for some long waits. Done prior to loading in your sequence means no wait time as there's nothing to draw.

TAKING QUILL TO PAPER

To begin working with a score, select the sequence you want to load in and click. Your sequence appears on the screen, Track 1 on the first staff, Track 2 on the second and so forth. Now comes the real work: turning your MIDI sequence into something intelligible. If you've never used a scoring program to convert a sequence before, you'll immediately see one of the problems with using a computer to score your music – the literalness with which a computer translates your performance to music notation. You wanted a quarter note but played a dotted eighth; the notation reads as a dotted eighth plus a sixteenth rest. You wanted four eighth notes played staccato, but your score reads four 32nd notes with dotted sixteenth rests in between each one. If you think it's a pain to go through an entire score to correct stupid computer conversion errors, you're quite correct. It's one thing to clean up your own mistakes, but having to fix timings because the computer is too literal in its translation is something else entirely.

This problem is inherent with all scoring programs that convert sequence files to notation, not just SuperScore, and there is a feature that helps take some of the sting out of it. By selecting a Score quantize function (which is separate from the sequencer's quantize value), you can select what value of notes you want to quantize your notation to for individual staves. This way you can keep down the number of 32nd rests you have to erase. Still, this is a broad fix and isn't really that helpful. For example, if you set the score quantize to eighth notes, you can forget the triplets you played, not to mention the 16th note sequence section. If you could selectively quantize individual measures or sections of the score, this feature might be more useful. *(According to Sonus, this feature is planned for version 1.3, which will be a free update. – Ed.)* The bottom line is that it gets to be a chore to go through a score and correct computer timing conversions.

TAKING ERASER TO PENCIL

Once you've got your sequence loaded, you'll probably want to make a few other adjustments aside from correcting mistakes as described above. SuperScore has spacing functions which allow you to spread out the score to make room for lyrics, chord symbols or any other symbols you plan on using. Using the horizontal and staff spacing features, you can resize your score and open it up. After doing this you'll want to repaginate the score (have the computer redraw the score based on the spacing commands you've just issued – just like in word processing). It doesn't redraw the staves in real time, unfortunately, so you need to give the commands first and then redraw the screen to see the result of your handiwork.

This kind of editing in non-real time is how SuperScore works when you get to the point and click editing stage as well. It was annoying at first to make edits, and then have to tell the computer to display the changes for me. However, the trade-off is that if the whole score had to be redrawn each and every time I made one tiny edit, I'd really be annoyed at having to wait three seconds (or more depending on the size of the score and number of notes and symbols being used) to get on with the next edit. So, you edit a series of notes and rests, or all of them if that's the way you want to work, and when you're ready to see what you've done, redraw the whole score.

Editing the score involves pointing at the note or symbol and clicking the mouse button. You're there if you want to edit it, move it, erase it or set the stem length. By selecting edit you're provided with an entire edit screen, where by pointing and clicking you adjust the note pitch and value, beam parameters and a number of other limits. When finished, the note or symbol you edited retains a small square box around it, letting you know you've changed something about it.

Although this function is pretty straightahead, I must say it's not very well documented. Point and click editing is not really covered in the manual itself, and the one page addendum enclosed doesn't really explain the quirks of the feature. For example, if you have four 16ths beamed together and want to change two of the 16ths to an eighth note, the 16th beams don't change; what you see on the screen appears to be three 16ths. What you need to do is readjust the beam partitions on the 8th note in the edit screen to get it properly formatted.

Once you've got your score in order, you'll want to add dynamic markings, slurs, trills, etc. By calling up the palette, you can point to and pick up any number of these symbols, as well as standard notes, stems, rests, etc. And if you don't find what you need on the palette, you can always enter draw mode and sketch it in yourself. A crosshair grid appears when you've picked up an item in the palette, making it relatively easy to place the symbol within the score. There are other options within this

Conversions *"Sonus, wisely, has incorporated a conversion program that will allow you to use your favorite ST sequencer, load the file into the converter and create a file which SuperScore can work with."*

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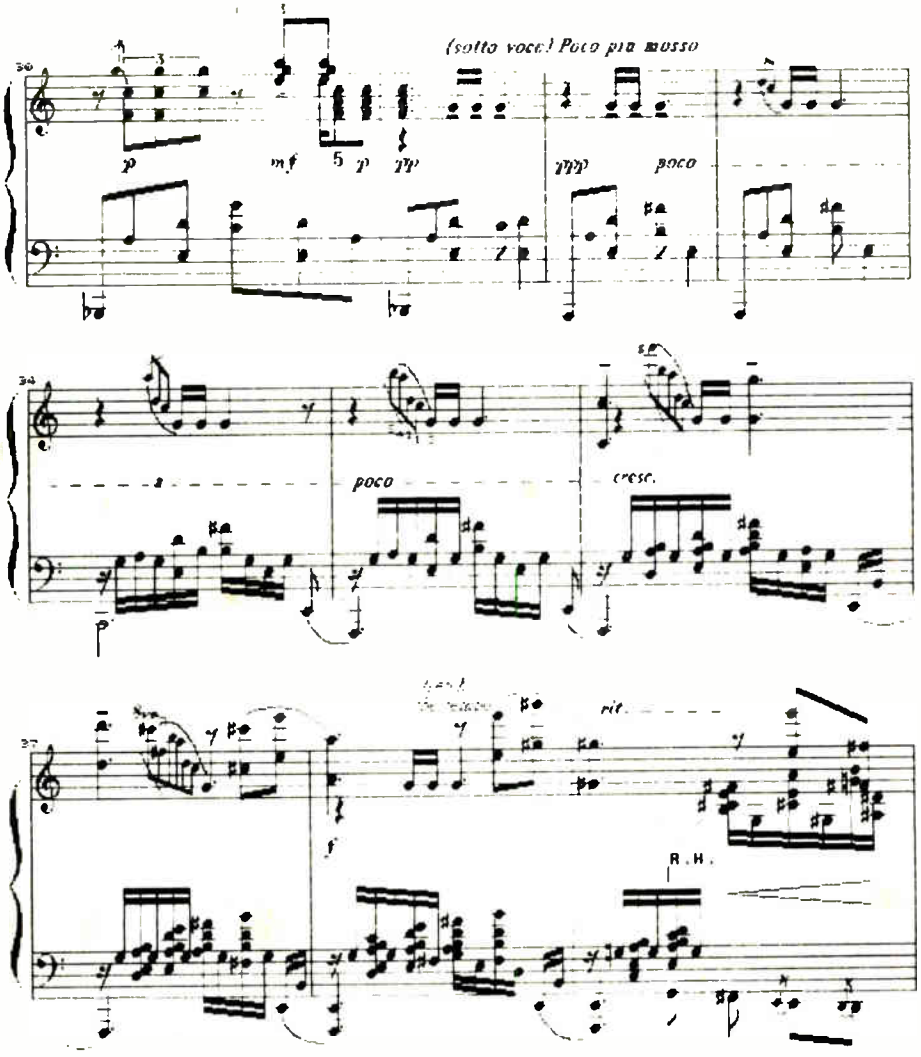


drawing mode, which helps you to draw good-looking crescendos, tuplets and other items. Suffice it to say you can control the final steps of preparing your score fairly painlessly.

There are other features, such as copying, moving, inserting and deleting measures, updating meter by bar, adjustments of barline styles, inclusion of lyrics, chord symbol placement including guitar chordings, and other functions that give you greater control over your music.

SEQUENCING

As mentioned earlier, SuperScore also includes a rudimentary sequencer for use in creating music. It's not hard to learn and doesn't contain many of the frills and features we see on the stand-alone sequencer packages, but then its function is different. The sequencing mode contains 24 separate sequences, each with 32 tracks. Sequencing is done drum machine-style, but there's really no limit to the length of the sequences you can make.

Standard quantize, transpose, bouncing, copying and basic functions you'd expect are all here, as is individual track shifting. What's missing, though, is a detailed edit page. (Sonus assumes any of this type of editing will be done in the Score Mode.) Once you get used to working with a sequencer that allows you to fine tune notes and dynamics with pinpoint accuracy, it really is a hard act to follow. Yes, you can punch in and out to correct the offending wrong note; but it is so convenient to be able to just find the note and adjust it in a MIDI edit screen. Tracks can be named, velocity scaling is supported, MIDI channelizing is a given, and overall, it provides you with more than enough features to get a piece of music recorded.

There is one thing to keep in mind, though: SuperScore will only notate the first sequence. Although all the other ones are active and have MIDI information within them, and they're all chained together to form the full song, you're still going to have to append each sequence to the end of the first one if you want to be able to turn the sequence into notation. Now why it's necessary to make the whole process include this one little awkward step, I don't know; but it seems to me that the end result could have been achieved a little more elegantly. In any case, it's not hard to do, just a bit unwieldy if you're going to go through the trouble of learning to use the sequencer in the first place.

But this package is not really about sequencing music, it's about scoring. The sequencer is admittedly not the main emphasis of the program and it's included so that it rounds out the product. Being that SuperScore is so dependent upon MIDI and sequencers in the first place, it truly would have been an oversight not to include some way to create a score within the package itself.

PRINTING & MANUAL

The printout of the music itself is quite good, and the printer drivers for both 9-pin and 24-pin printers are functioning. There is also a page size control, so printout can be scaled to print to any size paper, not just 8½"×11". You also have a wide degree of control over what you do print out. For example, if you have a score for six instruments, you can print out the conductor's score, or each part individually, or combinations of up to three staves. Transposition of individual parts can also be done from the printout stage, one of the main advantages to using a computer-based scoring product.

The manual is barely adequate. Although it's better than some, it still has that small type you have to squint at to read; there's no index, meaning you have to refer to the Table of Contents to look something up; and the text is primarily a description of each of the menu choices, with little attention given to the application of the product overall. On the instructional side, in an addendum there are two columns devoted to hints on using SuperScore, and the first chapter gives a quick start to getting going with the program, but I still found the manual lacking in depth. Too often I had to figure something out myself as I was either unable to find it easily in the manual or it wasn't there at all.

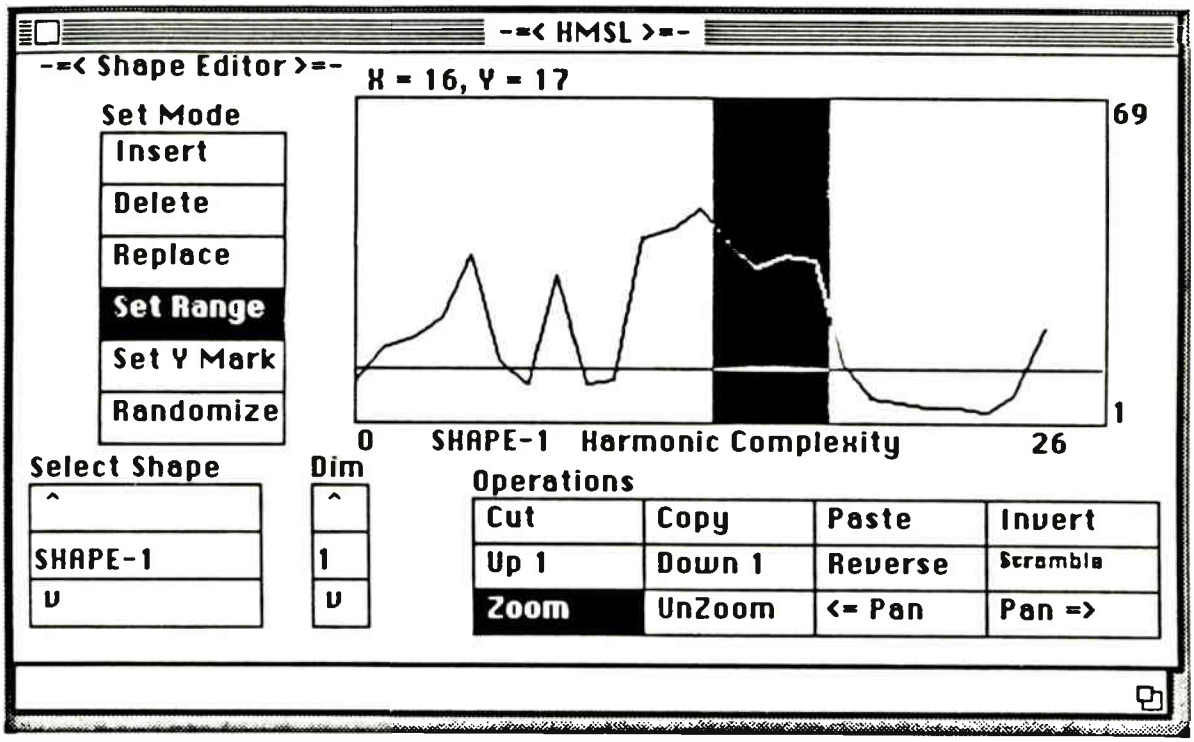
CONCLUSIONS

Overall, SuperScore is a good, solid scoring program, albeit currently missing a few features. The sequencing module, although not a whiz bang feature, certainly functions properly, containing most features you need. The updates underway on the program should rectify the weak areas and make this a complete package for the MIDI musician. Providing the conversion routines for other sequencing packages are implemented quickly, you'll be able to use this product with your favorite ST sequencer.

The problems with SuperScore are inherent in most scoring programs on the market today, which brings us back to the original question: does it make work easier? You'd have to compare the amount of time it takes to correct the computer's conversion of MIDI files to notation information, the time it takes to edit your score into a usable piece of music, complete with markings, titles, etc, the time it takes to program the sequencer with your music, and all the other factors that would go into getting a viable product out of the package . . . to the simple action of taking pencil to paper (or ink to onion skin if you prefer). You may not be able to publish your handwritten manuscripts, but you also may find it a faster way of accomplishing your goal.

And that's what it boils down to, determining if the use of a scoring program will assist you in achieving a musical product. If it does, SuperScore is certainly a product you'll want to take a close look at. ■

HMSL Software Language



The Shape Editing Screen

For the adventurous, a derivative of Forth that runs on the Amiga or Mac for creating compositional environments. *Review by Carter Scholz.*

ONCE THE SOLE domain of university studios with expensive or one-of-a-kind equipment, "computer music" has broadened with the advent of MIDI to include lower-cost, mass-produced gear – both hardware and software. Designed for gigging pop musicians, this new gear nevertheless has found its way back into academic studios, where it's ironically generating a new wave of software more suited to the interests and needs of academic and experimental composers.

The recent spate of notation programs, along with "intelligent" instruments and algorithmic composers, and the addition of microtuning to some new synthesizers – ideas that have been around the academic music scene for years – show to what extent these previously disparate worlds are finding common ground.

One of the most interesting fruits of this hybridization is HMSL, the Hierarchical Music Specification Language, developed at the Mills College Center for Contemporary Music. Mills College has a venerable tradition of experimental music – many of the most significant composers of recent decades studied or taught there, including Steve Reich, Terry Riley, Lou Harrison, Morton Subotnick, Pauline Oliveros, Robert Ashley, David Berhman, Ron Kuivila, Maggi Payne, and Anthony Braxton. HMSL is the work of Mills College regulars David Rosenboom and Larry Polansky in conjunction with the creator of the programming language "Forth," Phil Burk.

HMSL OVERVIEW

HMSL is not an application – it's a programming language itself specifically tailored for musical use. The form of the language was first sketched by Polansky and Rosenboom in 1978. By 1981, before MIDI and before the Macintosh, HMSL was running on a custom-built 68000 microcomputer controlling a Buchla digital synthesizer. Today it runs on the Amiga and the Macintosh, using the Amiga's local sound and MIDI instruments. But the basic ideas of the language haven't changed much. "The original design of HMSL doesn't specify any hardware," says Polansky. "The formal ideas are more important than the implementation. When MIDI came along, we just added it; it wasn't a big deal."

Technically, HMSL is an object-oriented extension of the Forth programming language. It requires the appropriate Forth compiler (JForth for the Amiga, Mach2 Forth for the Macintosh) and some familiarity with the Forth language. Says Polansky, "The vision that guided HMSL was that composers could be their own programmers. We saw that the next step in computer music was sophisticated languages, not bigger applications."

The central idea of HMSL is that of "morphologies" or "morphs." A morph is a set of ordered points which refer to musical data. The simplest sort of morph is a "shape" – raw data points that might represent the notes of a melody or the points of a waveform. But morphs can also be complex, multi-dimensional hierarchies, which may include other morphs, or methods of transforming or playing morphs (for example, Polansky has created a piece called 'Cocks Crow' that treats the nine programmable parameters of a Roland DEP5 signal processor as a nine-dimensional shape).

The purpose of keeping things so abstract is to avoid imposing a fixed meaning on a shape – to give the composer full control over its meaning. Contrast this to a MIDI sequencer, in which every bit of data has a fixed and irrevocable meaning: you can add crescendi and transpose melodies, but velocity data always

remains velocity data, and a key number always remains a key number. In HMSL, a shape isn't given a concrete musical realization until the moment it is performed. These shape-givers are morphs known as "players," which are designed by the composer.

DEALING WITH HMSL

The high degree of generality built into HMSL throws a lot of responsibility into the lap of the user. Learning HMSL is a task far more formidable than learning a stand-alone application. It requires: 1) learning Forth, 2) learning ODE, an object-oriented set of extensions to Forth, and 3) learning HMSL, which was itself written in ODE. Acknowledging the difficulty, Polansky says, "We probably have some users who have had trouble learning it because of its complexity, and I hope the new manual we've just printed will help that. With the new extended tutorials you can probably become a decent HMSL programmer within a few weeks."

Although it can be used to produce fixed, deterministic pieces, HMSL is designed for real-time, interactive performance. Its "stimulus-response" mechanism can parse, or break down and identify incoming MIDI data, so any programmed action or set of actions can be triggered in response to specific MIDI events. It can also trigger actions in response to internal processes. In addition, you can interact with HMSL via two user screens: the shape editor, which lets you edit any dimension of any shape as it's playing; and the action screen, from which you can trigger previously-defined responses to specific stimuli and adjust their probabilities and priorities.

Action tables are where these probabilities and priorities are held. For example, you might tell an action table "whenever you see a middle C with a velocity over 88, there's a sixty-percent probability that you should play sequence A." There can be up to 64 actions in the table. Unlike a simple sequencer, however, the actions triggered need not be deterministic – they can be varied and transformed in response to other stimuli (including stimuli generated by ongoing HMSL actions), creating a vast, complex, interdependent web of processes.

COMPOSITIONS

What sort of people are using HMSL? There are about one hundred registered copies in use. Phil Burk estimates that half are in university studios; ". . . then there's a fairly wide mix of people who've been seriously involved in experimental music and see this as a tool." Polansky adds: "There are also a lot of Amiga people getting into the sample manipulation aspect of it. There are some people who've only used commercial products; we get letters that say, 'I'm just so frustrated – but HMSL looks like it'll allow me to do the things that I want to do.' "

And what sort of music are they making? Pauline Oliveros wrote a piece called 'Dear.John' for John Cage's 75th birthday using HMSL. The pitches C, A, G, and E were arranged in intricate polyrhythmic patterns with the computer deciding aspects of the polyrhythms. 'Dear.John' was written and realized in a couple of days at the Mills CCM, using a Kurzweil 250. HMSL was used to write and record the piece, but the piece is "performed" from magnetic tape, not HMSL.

David Dunn has designed an HMSL piece called 'Mock' which takes his battery-powered Amiga 500 computer into the field, using a Sony Watchman as a monitor. An A/D converter allows him to sample birdsongs in real time, and play back up to four songs at once, transforming them in real time, while the birds respond to the Amiga's sounds.

Phil Burk's 'Swirl' is a theme-and-development study which takes a melodic shape and rotates it in two-dimensional pitch-time space. The sound of the piece as it evolves is greatly complemented by the sight of the melodic shapes being rotated on the video screen.

David Rosenboom's 'Champ Vital' (Life Field) uses HMSL to set up what he calls "concept spaces," in which parametric transformations and mutations of structural entities are calculated. Though the complexities of the work are such that it demands HMSL for its calculations, it is scored for violin, piano, and

Larry Polansky, one of HMSL's authors, using the language in concert.

percussion, and is performed from traditional notation on traditional instruments. Rosenboom's 'On Being Invisible II' takes a different approach – as HMSL performs the piece, it analyzes a listener's brain waves, and uses his or her responses to affect the musical flow and structure.

Polansky's 'Simple Actions' is an improvisational work for solo performer and computer; he currently performs it using the Amiga local sound and a Yamaha FB01. A multitude of simple but "intelligent" algorithms interact to produce complex results beyond the performer's expectations (and, sometimes, control). His 'Seventeen Simple Melodies of the Same Length' uses an IVL Pitchrider to listen to improvised melodies played on any instrument, then uses algorithms to compute the "distance" between the melodic shapes. It's the first "portable" computer music piece Polansky has written, and he's enthusiastic about it: "I can just send performers a disk. All a performer needs is a Mac or an Amiga, a Pitchrider, and any MIDI synthesizer. And they don't have to know anything about computers. I find that exciting because David and I have talked for many years about documenting algorithmic pieces, so that what you write down isn't the notes that result from the process, but the process itself. How do you notate that so that it maintains some continuity with our traditional feelings about notation? And the answer turns out to be simpler than we thought: just send the code on a disk."

AND BEYOND...

Just as shifting HMSL over from a Buchla 400 to MIDI was no big deal, adding controllers for other hardware is not a problem. New York sculptor Sarah Armstrong is using HMSL to control air blowers in her enormous installations. Polansky believes it can go further than that: "I think when video people discover it, it's going to really take off, especially with these new products on the Amiga like Mandala (*see the review in MT February '87 – Ed.*). Artists are going to realize how powerful a video and graphics generator it is. We're also hoping that people will write lots of system exclusive toolboxes for various MIDI instruments – we already have them for the FB01, D50, and Roland DEP5. That will make some applications easier."

That's not to say that HMSL will ever become the language of choice for commercial applications. "The HMSL license states clearly that people cannot write an application in HMSL and then sell that application." Rolansky explains further: "This is not meant to be prohibitive. In fact, it's community-oriented. We want everything written in HMSL to be in the public domain. We want it to serve as the basis for shared musical experimentation, and if people start marketing their ideas from it, that's not in the spirit of the language's development. HMSL itself is not public domain because we don't think it would survive – it's too hard to support."

Although one goal of HMSL is to avoid "stylistic bias" (and the diversity of work done with it supports that claim) it does have a pronounced philosophical bias. In a way, HMSL can be seen as an embodiment of Polansky & Rosenboom's music-theoretic ideas, and as such may be more influential as a way of thinking about music than as a way of making music. That's okay with Polansky.

"A lot of people are just curious, and buy it, and dabble in it, and then don't use it as their primary music software. But I think it affects what they do later. I think its impact is disproportionate to the number of actual users. I mean, how many people have actually heard (Steve Reich's) 'Come Out'? But how many people have been influenced by its existence? I think that's one of the most important reasons for doing it – as an idea." ■

PRICE HMSL, $150; JForth or Mach2 Forth, $100
MORE FROM Frog Peak Music, Box 9911, Oakland, CA 94613. Tel: (415) 430-2191.

LIVE & DIRECT

Photography *Nicholas Rowland*

Michael Martin

After fourteen years as an "underground" UK reggae band, Aswad has their first US success. But where does high technology fit in with the hard-edged music of Rastafarian culture?

Interview by Nicholas Rowland.

GOOD FRIDAY, ON stage at the Astoria, London. The lead vocalist steps forward to the microphone. "I don't know if you've seen our latest album. But when you do, you'll notice there's a little sticker on there which says 'Includes the hit single . . .' I just wanna say thank you to everyone who helped put that sticker there." In response, whistles, shouts and air horns: the packed auditorium goes wild. The record in question is at No. 1 on the British pop charts.

A familiar scene: youthful band plays live version of current hit to appreciative audience. But hang on a minute – this is a band called Aswad, a band who has been together 14 years. Who in that time has never had a record in the Top 40, let alone at pole position. The reggae band from Ladbroke Grove with a hardcore grassroots following, the respect of many of their pop contemporaries for their electric stage presence, but who to most of us within the mainstream remain unknown.

'Don't Turn Around' is the piece of vinyl which has reversed their fortunes, proving the crossover which has always been the band's aspiration. And while its sweet lovers rock tones may not represent Aswad at their hardest, the fact that they have that reputation behind them gives hope for the future of reggae as a commercial genre. No wonder the present mood is one of euphoria, future expectation mixed with a healthy dose of disbelief.

Despite 14 years spent together in the music business, success has hit the band like a sledgehammer. You wouldn't know it seeing them on stage, but they are totally exhausted. What was intended to be just another tour has turned into a 24-hour press conference. Having ignored them for the last decade, the popular press has descended to claim Aswad for their own.

This much is obvious before the gig. Singer Brinsley Forde, bass player Tony Gad and drummer/singer (Angus) Drummie Zeb are running round like the proverbial chickens with their heads chopped, instinctively throwing quotes into any Walkman thrust under their nose and smiles at anyone armed with SLR and flash. I reckon under this sort of pressure they'd sign their grandmother away.

Drummie Zeb manages to sum it up in fewer words. As he rushes between interviews and photo sessions he's heard to mutter, "I'm dizzy, boss."

It was a very different story when Aswad released their first single, 'Back to Africa,' in '76. At that time, the five-strong outfit – Forde and Drummie, plus Donald Benjamin (vocals and lead guitar), Courtney Hemmings (keyboards) and Ras George Levi (bass guitar) – had just signed to Island, the first UK reggae band to be taken up by a major label. 'Back To Africa' went straight to the top of the reggae charts, paving the way for the success (in reggae terms) of their eponymous debut album. This was quite an achievement considering that at the time it was thought that good reggae music was the sole preserve of Jamaican studios, even though on closer inspection many Jamaican records were little more than reggae covers of soul classics. Aswad's original sound and diverse influences proved to be the much needed boost of confidence the home-grown reggae scene needed, particularly as their message tempered the general cry of Rastafarianism with the everyday experiences of blacks in Britain: racial discrimination and continual police harrassment.

Martin: *"As more of a player than a programmer, I don't like the playability of the D50 compared to the DX7 – it feels like the early synthesizers. But the sounds just knock me out."*

Aswad and Island parted company shortly afterwards, the band leaving Britain to tour on the West Coast of Africa and record in Jamaica. *It's Not Our Wish* appeared in '78 on the Grove Music label, with *Hulet* the following year. Meanwhile Tony Gad had replaced Hemmings on keyboards, one of the band's more significant line-up changes. Gad joined in time to record a song called 'Warrior Charge,' featured in *Babylon*, a film starring Brinsley Forde.

In 1980 Aswad moved to CBS, recording *New Chapter* and *Not Satisfied*. This too proved a short-lived alliance, though *Not Satisfied* charted briefly. Then Aswad was back with Island for *Live and Direct*, a recording made at the Notting Hill Carnival. This successfully captured the band's impressive stage performance and proved beyond all doubt that Aswad was now the most important influence in reggae. But as far as the long-hoped-for crossover to the wider pop world was concerned, well it simply wasn't happening.

THE NEXT FEW years saw some creditable near misses. 'Chasing For The Breeze,' '54-46 That's My Number' and 'Bubbling' (a Toots and the Maytals classic) gave them minor-league pop chart success in '84/'85, while 'Roots Rocking' and 'Gimme the Dub' struck hard at club level. But since Aswad was back with Simba, their own indie label at the time, the resources just weren't available for any greater success. No wonder then that survivors Gad, Forde and Zeb took their new management's advice and once more knocked on Island's door.

The current album, *Distant Thunder* has been over a year in the making, and is Aswad's slickest and most polished production to date, clearly aimed at daytime radio, not the "graveyard slot" on KISS FM.

Ironic really, when you consider that just as Aswad has taken onboard mainstream pop's glossy production values, the recent chart success of hip hop and house has shown that the charts are able to assimilate a more rough and ready approach to recording. There's further irony in the contrast between the sentiments of tracks like 'The Message' (the opening cover of Cymande's rare groove classic) and 'Set Them Free' (a musical discourse on Soweto, Sharpeville and sanctions) and the squeaky clean production.

Surely reggae's rebel message and pop's plastic medium are, to a certain extent, mutually exclusive. Zeb, for one, thinks not.

"What we want to do is not to turn reggae music into pop, but to make reggae music popular. There's a difference there. Why should the sound of reggae be dirty? Why should it not be as polished as any other form of music? Yes, there is a tradition of reggae being a certain way, but if we try that, people say, 'This is violent music, it's not well produced so we're not going to play it.' But now they say, 'This is great, this is *ethnic*, let's put it on daytime radio.' It's not so much that we're looking for a commercial sound as much as an international sound. I see nothing wrong with that. Reggae has an international message."

But while Zeb sees the new Aswad studio style resulting from a more positive attitude to future direction, bassman Gad explains that, in practice, it has much to do with solving problems of the past.

Gad: *"Pop music is doing today what reggae was doing in the early '70s taking parts out of the mix, or bringing in echo effects is all taken from Dub."*

"I think up to now we've been very uncomfortable in the studio and so we never really got down onto tape those vibes that we gave out live. But also, we had a lot of problems getting the sort of sound we always had at the back of our minds. I remember the times when we used to sit in the studio and spend the first few days just trying to get a good bass drum sound. That's tedious stuff, man. And when you ain't got too much money and you're paying for the studio, you end up doing things too quickly and always thinking at the end, you should have done that different, or better, by which time it's too late.

"What's happened, though, is that while we've become much more familiar with the studio, the technology has also become much more accessible. Good sounds are so ▶

Drummie Zeb

Photography *E*

Brinsley Forde

► much easier to come by. If you want a good drum sound all you have to do is go out and find a good sample. I see no cheating in that, if the end result is that you don't spend days in the studio trying to mic up a real drum."

WHILE ASWAD HAS consistently embraced new technology throughout their career, the name of programmer Pete Gleadall is mentioned as the "brother" who has really brought the band's equipment and attitudes up to date over the last two years. Both Gad and Zeb talk about MIDI and sampling as though they were inventions of the last 18 months, which leads me to suspect that it was really the latest deal with Island which allowed them to get into these particular things in a big way. On more than one occasion Forde has been heard to say that there's a big difference in how you use technology when you actually own it, rather than just renting it when you go into the studio.

This Gad confirms: "We've all got the MIDI gear at home now. So basically you can get to learn all about it before you go into the studio. At one time I used to have two tape decks. I bounced things from tape to tape and ended up with 16 tracks of hiss. Now I've got an MC500 sequencer, a TR707 and a TX81Z so I can sit in my house and get ideas down really quickly. Drummie has a Studio 440, Brinsley has a whole rack of gear too."

Apart from giving Aswad greater control of the overall band sound, both Zeb and Gad say that increasing control over the means of production also changed the way they approached recording *Distant Thunder* even before they went into the studio.

"This time we were able to plan ahead much more," elaborates Gad. "Instead of going in and just jamming together, we could all sit round and work on things together. Like we might all work on a drum pattern before all looking at the bassline."

"That's not to say that we put it all into a sequencer, took that into the studio and just pressed the button," counters Zeb quickly. "There was still a lot of jamming, but at an earlier stage, we were using the sequencers as digital recorders. Like I played rhythms into the Roland MC500 in real time using the metronome as a guide. Then we'd edit it, or just use it to trigger samples."

Did they find that having their own home setups meant that their songwriting became less of a collaboration than in the past because each of them now had the power to come up with a finished track?

Gad shakes his head: "I know what you're saying. There is a danger of that, of one man sitting at home and making tracks by himself, which is why you get a lot of records produced so quickly, but without any real quality. But I think where Aswad is concerned, it would be impossible for just one of us to come up with anything whole by himself. I mean, I'm really the player in the group, Drummie engineers and Brinsley writes, so really it's only when we all get together that it comes out as the finished whole."

A different and more objective perspective on the triumvirate in action comes from the man who's currently tickling the ivories with them on stage; Michael "Cool Walk" Martin joined the expanded stage line-up just over a year ago, though he too was involved in the recording of *Distant Thunder*. (It's his keyboard playing that you can hear on 'Don't Turn Around.')

"Basically," he says with a smile, "I get called in whenever they get stuck. Like when there's a technical keyboard part or they just want someone with new ideas to take them in a different direction."

Martin is currently at the Guildhall School of Music studying piano and jazz composition, though with Aswad's heavy touring schedule ahead, it's unlikely he'll be doing much studying for the next few months. Before joining Aswad, he toured with Maxi Priest, a reggae artist who has also made a successful crossover to the mainstream pop charts with an equally well-crafted sound.

"I must say that before I started working with Aswad, I couldn't really understand this thing of 'Britain's greatest reggae band.' Now I do. I don't know if it goes with the years they've been together, but there's something that makes the music gel much more than the other British bands I've played with. It's more like guys in Jamaica. I think it's because a lot of their thing is really very spontaneous. Other people

wanted it played exactly like their album, which was cool, because after all it was their gig. But Aswad is much more relaxed about things.

"Now I realize that a lot of it has got to do with their rhythmic concept, which has the foundations and influences of *all* 'black' music. So you find a flavor of calypso and latin or touches of African along with a hardcore reggae bassline. Like they have breaks that no-one else would have, or spontaneous patterns between the bass and drums.

"They're really open-minded to all kinds of different musical styles. They want you to come in and try something out. That's where my jazz background comes in. And the new lead guitarist has got that as well – he's really jazz oriented. You know, he takes two chords and turns them into four – and they listen to what you suggest too. In that respect they don't make you feel like a session player. When I came in to the studio there was a basic foundation, obviously, but I was allowed to spread harmonically over the top and stretch the chords a bit and make them more interesting. Perhaps listen to the brass and then stretch the arrangements down to meet it."

'Warrior Charge,' say, which has that kind of feel. Man, I don't know what a bossa nova beat is, but it must be there inside me."

Gad: "It's not so different anyway. What pop music is doing today is taken from what reggae was doing in the early '70s. Like the idea of mixing a track, of taking parts out of the mix, or bringing in echo effects – all that is taken from Dub. Reggae shouldn't be really thought of as a separate thing."

ON STAGE IS really the best place to see Aswad in action: the technology, the improvisation, the sheer good time that is had by all. It's one hell of a band, and everyone seems to have a stack of gear behind them. Forde sports electric guitar with a MIDI pickup controlling a TX7, the sound of both being processed by an SPX90 and Roland GP8. Zeb sits behind a set of Simmons pads and an Octapad which trigger a combination of two SDS9s and the S900. (The Studio 440 has been dropped after many breakdowns – it's now known as The Hassle Machine.)

Meanwhile, Gad alternates between electric bass and the Roland GR77 MIDI bass controller.

Gad: *"At one time I used to have two tape decks, and I ended up with 16 tracks of hiss – now I've got an MC500 sequencer, a TR707 and a TX81Z."*

Martin sees his experience with Aswad as quite a contrast to the kind of amateur and semi-pro reggae acts he played with as a teenager.

"They didn't experiment at all. It was like there was a strict definition of reggae which you had to stick to. I used to think, man, there has to be more to music than this. That's partly why I decided to go to college and study jazz.

"Reggae has a lot of experimenting still to do and Aswad seems prepared to do it. It's like 'Don't Turn Around' – a lot of people don't automatically think of it as a reggae track, I think because the keyboard arrangements actually take your mind away from it, they disguise it in a strange way. It shows that reggae doesn't have to be heavy, or just drum and bass, that it can be just about good songs and good arrangements."

Main man Zeb is happy to confirm this spirit of keeping options open, putting it down to the kind of musical company that Aswad has kept over the last 14 years.

"Soul, hip hop, calypso – we listen to all these things, we know the people involved, so naturally we end up being influenced. It just seems to come out. But we don't analyze things. Often I'm doing things on the drums and I don't know what I'm playing until I listen to it on tape. Then someone might point out I've been playing a bossa nova beat or something, like on

"Before the GR77, I really used to get bogged down on the bass. I originally joined the band as a keyboard player, so in the studio I used to do all the basslines on a keyboard. But when we played live, the keyboard player had to do them instead because we usually wanted those specific keyboard sounds again. So when the MIDI bass came out, it was like 'Hell yeah, it's arrived.' Suddenly it was like I was free. And it meant it allowed me to get an identity on the bass, which is really important to me. Up to then I was beginning to feel that in terms of sound possibilities the bass guitar had just about become obsolete."

Also plugging into the power of MIDI live is lead guitarist Stanley "Soon Come" Andrew. Meanwhile, over his right shoulder can be seen Martin with a D50 and two DX7s, one of which acts as a controller keyboard for an S900.

"They just gave me a D50 to play. Up to then I'd worked mainly on the DX7. I must say, as more of a player than a programmer, I don't like the playability of the D50 compared to the DX7 – it feels to me like the early synthesizers. But the sounds just knock me out. It's the reverb I suppose which makes the sounds so much more sparkly and clear. The highs really ping – I have to get one after the tour."

Meanwhile to Martin's left there's another keyboard player on RD1000 piano and another D50, who also doubles up on drums (not simultaneously, you understand) whenever Zeb comes center stage to sing lead vocal. Only the four-piece brass section is exclusively acoustic.

But while this may sound like MIDI paradise to hi-tech addicts, Martin explains that, when just about everybody in the band has access to the same sorts of sounds, it takes some getting used to and not an inconsiderable amount of organization.

"I must admit that when I found out what Aswad used on stage, I had a few problems with it. I think it was partly an ego thing I suppose, like 'Look, I'm the keyboard player and these sounds are my patch so leave it alone.' But then I realized that if we worked it out, we could actually get some really powerful arrangements. If the guitar plays a wash of strings or whatever, it gives the keyboard player the freedom to do something else. Your hands aren't tied to just padding out the sound.

"In the end, with Brinsley in particular, it worked out naturally. We just gave each other space at the right time. He didn't interfere with my keyboard parts, didn't try to play over me. And if he did, well, it's his band anyway! But I'm glad now because through having to be careful about who was doing what, I really opened my ears to what he was playing."

As for the future, there's a lyric on *Distant Thunder* which sums up Aswad's aspirations: "I'd like to find a melody that the whole wide world could sing." That's quite an ambition, especially if you've got to get the world to listen to your melody first. Happily, 'Don't Turn Around' seems to have opened things up, not just for the album but for the massive world tour too, a tour which has already taken them through Europe, Japan, New York and Los Angeles. But if Aswad becomes international big business, could there be a danger of losing contact with their roots?

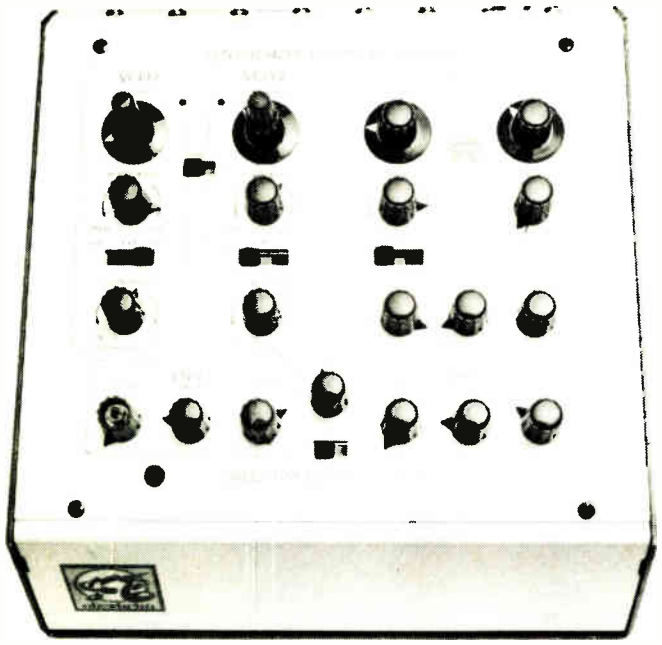
Zeb: "As I've said before, this album is still for all our hardcore fans, for all the people who have supported us, but it's also for the world, for the kids who see us on *Top Of The Pops*. I think reggae can take it."

The closing comment comes from Martin: "I'm glad it happened for them. After Marley and Tosh, reggae has had no ambassadors. The success of 'Don't Turn Around' is like a ray of hope. It shows to all the other reggae bands out there that something can happen."

And as if to illustrate the point, as I leave Aswad's Astoria gig in England, a leaflet is thrust into my hand advertising an evening of Message and Music, featuring Misty in Roots, Benjamin Zephania and Ras Messengers. ■

classic synths of yesteryear

BOX OF TRICKS



If you thought that synth expander modules were a post-MIDI phenomenon think again – the Oberheim Synthesizer Expander Module (SEM) was a box to be reckoned with over a decade ago. *Text by Steve Howell.*

SOME TIME AGO I saw an ad in an old *Melody Maker* offering an Oberheim expander module for sale for (what amounted to) under $200. I phoned instantly on the off chance it hadn't already been sold. "It's still here," came the reply, "but it's not what you're expecting. It's not the recent Xpander but an old analog box I've had for years." Well, the secondhand market may not favor the seller of musical equipment at the moment, but I hadn't really expected to pick up three grand's worth of synth module for less than a tenth of list. I had other hopes about what the mystery expander might be. My luck was in – the man was selling an old Oberheim Synthesizer Expander Module – or SEM. We met, we talked, I bought . . .

When I got my new toy home I found that it wasn't working too well. No problem – this was a machine that came from a time when circuit diagrams were part of the handbook. Although I'm no whizz-kid, 10 minutes with a soldering iron cured a simple grounding problem. Eagerly I set it up and played. I still haven't entirely recovered.

The innocuous little white box you see in the photograph is actually a complete synthesizer. It contains two VCOs, a multimode VCF, a VCA, two EGs and an LFO. Along the top panel are facilities for all the inputs and outputs you're ever likely to need – apart from MIDI, of course. Nineteen seventy-four is as closely as I can date this particular example of the SEM, at which time it was being sold as an expander for layering sounds with the Moog Minimoog and ARP Odyssey. Later it appeared as the basis of the Oberheim OBI programmable monophonic synth and the two-, four-, and eight-voice multitimbral synths.

So what makes the SEM kick? The short answer is I don't know – you try one and tell me. There are plenty of synths around that have similar specs but not *that* sound. Perhaps it's the oscillators: rich sawtooth and square/pulse waveforms that can overdrive the filter for smooth, controlled distortion. Or perhaps it's the characteristic Oberheim filter (designed by Jim Cooper, now of JL Cooper Electronics), or the punchy envelope generators. Most likely, it's a magical combination of all these things, but there's no denying that this diminutive parcel of analog electronics has a unique sound quality, paralleled only by the Minimoog and Odyssey in the same era.

On Closer Examination

THOSE TWO OSCILLATORS mentioned above have coarse and fine frequency controls, along with control inputs where you can choose among signals from the EGs, the LFO or an external signal. These control signals can be independently inverted for further modulation flexibility. Oscillator pulse width is variable in ratios from 10/90% through 50/50% to 90/10% and can be modulated by the same control sources as the pitch. Sadly, only one waveform is available from each oscillator at any time (although it is possible to modify a SEM to have both up at the same time) – one knob fades from 100% sawtooth to silence to 100% pulse. Still, this doesn't seem to detract from the power of the SEM. The oscillators also feature a switchable hard sync facility capable of producing a dirty sound that should satisfy even the most fanatical analog devotee.

The only simple way to describe the filter is "unique." It is totally voltage controlled with the same modulation facilities as the oscillators. It also offers no less than four filter modes: low pass, band pass, high pass and notch reject. Interestingly, the selection between

"Final pages of the manual detail options for user-modification, and the board itself has pins that can be used to solder sockets to, according to your own requirements."

low pass, notch and high pass is totally variable – you can set up a combination of low pass and notch reject or a combination of notch reject and high pass in completely variable amounts. These filtering options gave a wealth of new sounds (nasal woodwinds, pizzy harpsichords) and are rarely equalled today and seldom surpassed (Oberheim's own Xpander and the Yamaha TX16W are the only ones that do, and they don't have variable mixes across their filter types). Many old synth addicts are quick to dismiss 12dB/octave two-pole filters (as in the Oberheim) as useless up against the 24dB/octave four-pole variety found in the likes of the Minimoog, but these simply cook (some say it's because of all the noise and dirt in the internal power supply . . .). The filter also has an external signal input, for processing sounds outside the box (such as guitars or tapes).

The two envelope generators are similar to those found on the Minimoog in that the initial decay and final release times are controlled by one knob. In some ways this is a limitation, but it doesn't seem to have handicapped the Minimoog too greatly, does it? The transient times of the SEM are comprehensive – from about the fastest attack time you'll still find today (1msec) to a leisurely 15 second plus release time.

The LFO generates only a triangle wave for vibrato and sweep effects – if you want anything more whacky you'll have to "import" a more unusual control signal through the unit's external inputs. VCA control is also limited to just the envelope, with a "drone" switch also available to keep it open at all times. Again, anything more adventurous necessitates the use of an external control signal.

A Real Manual

THE MANUAL THAT accompanies the old Oberheim should be made required reading for all writers of today's equipment manuals. It gives a comprehensive run-down on the SEM's controls and gives clear and detailed information on interfacing the SEM with various other synths. The best is kept 'til the end, however, as the final pages are devoted to offering various options for user-modification. There's a diagram of the circuit board showing the inputs and outputs of the various modules, and the board itself has a number of pins that can be used to solder sockets (and so on) to, according to your own requirements. To complete your modification Oberheim thoughtfully provided a row of ⅛" jacks – though there is room to drill holes in the case to fit more. (Our technical editor has an old Oberheim Two-Voice that has two of these modules and a sequencer which he turned into swiss cheese with jacks, and made fully patchable).

Briefly, the SEM allows you to fit: individual waveform outputs, sync inputs and outputs for each oscillator, control inputs for the VCOs, separate control and audio inputs to the filter(s) and independent filter outputs for simultaneous access to all four filter types (although they will all share the same modulation), final VCA output and input, LFO output and trigger input, separate EG voltage outputs, and various triggering options and a variety of external audio signal inputs. I was impressed. But the story doesn't end there, as Oberheim provided nice little stickers for you to give your mods a professional look. Of course, you can take things a stage further than Oberheim intended by adding facilities like external control of pulse width and EG sustain level. The world, as they say, is your oyster.

Having all of these external connections makes them a breeze to MIDI, along with interconnecting them to other pieces of dinosaur equipment (like a Sequential Model 700 programmer, which includes two more comprehensive envelopes along with VCO and filter offsets and 64 memories – thus making your SEM programmable).

But, as I mentioned earlier, even without any of these modifications the SEM is not a synth to be quickly dismissed. Bass sounds that will kill at 1000 paces, lead sounds that would cut through a Motorhead gig and, using the filter options, a fantastic range of almost digital textures are all part of the expander's character. Fast filter attack transients also make it well-suited to the generation of percussive sounds and, if that weren't enough, I found it more than passable as an imitative synth.

The End

IF ALL THIS has whetted your appetite for the SEM, you could do worse for a demo than to listen to some of Jan Hammer's early albums – I'm reliably informed that his "lead guitar" patch is a combination of SEM and Minimoog. Roger Powell slaved a number of them from his polyphonic "Probe" keyboard in Utopia. Larry Fast is another SEM user and Weather Report's Joe Zawinul made great use of the SEM's close relative, the Oberheim four-voice. Many (Patrick O'Hearn, etc) still use the SEM today for bass sounds.

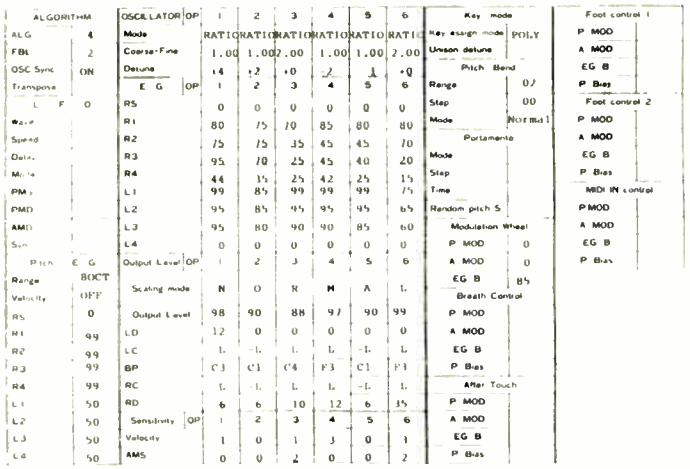
Although I managed to pick up my SEM for under a ton, you might not be as lucky as I was. Unfortunately, these wonderful boxes aren't as plentiful in isolation as they were incorporated into instruments. You might find it easier to pick up a Four-Voice, which is basically four of these beasts chained together in one box with a rudimentary programmer, or the Two-Voice which incorporates two SEMs and an analog sequencer (the latter also includes a white noise source and sample/hold facilities). In today's FM-obsessed world the SEM sounds just great to these ears. ■

PATCH WORK

Got a killer pad, some monstrous strings or any other sonic creation you'd care to share? If so, send it – on a patch chart from the owner's manual (along with a blank one for artwork) – and a demo-tape (very important) to: *Patchwork, Music Technology, 22024 Lassen St., Suite 118, Chatsworth, CA 91311.*

If you dazzle us with your brilliance, we'll reward you with a free one year's subscription to MT. Get twiddling and get scribbling.

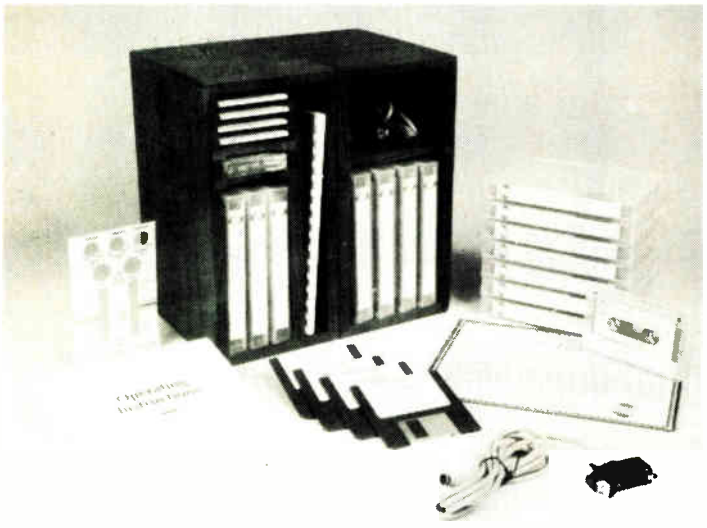
PROGRAMMING



YAMAHA DX7II
For Potts
Larry Mueth, Fenton, MO

Named for a friend who answered some questions for him, Larry's contribution is a warm electric piano-type timbre with a nice bottom end to it. The mod wheel adds a nice bit of brightness and bite to the sound. Enjoy. ■

PATCH WARE



Sound Genesis is offering a new library of sounds for the Fairlight Series III.

NEWS: Last month we had no news and one review; this month we've gone in the other direction.

Owners of any of the **Kurzweil** expanders (the 1000PX, SX, HX and GX) will probably be ecstatic over the introduction of a new line of **Sound Blocks**. The first Sound Block is for the 1000PX and will feature a full drum kit, electric bass, electric piano, flute, plus a number of synthesis waveforms. Sound Blocks for the SX, HX and GX are expected to follow soon, and Kurzweil has announced their intention to follow suit with an ongoing series of new Blocks. Both old and new sounds are simultaneously accessible – you won't lose anything with the addition of the chips. Installation can be done at authorized Kurzweil Service Centers. The suggested retail price for the first Sound Block for the 1000PX is $495.

We also found out that the Sound Blocks will soon be available for the K1000 as well, but for $100 more than the expander Sound Blocks (apparently K1000's need an extra "daughter" circuit board). According to Kurzweil representatives you'll be able to have two Sound Blocks in a K1000 and 1000PX and up to three in the HX, SX and GX modules. (Is that the sound of lips smacking?) For more, contact: *Kurzweil Music Systems, Inc, 411 Waverley Oaks Road, Waltham, MA 02154. Tel: (617) 893-5900.*

Sound Genesis Corporation has announced the availability of **Volume 2, Essential Percussion** for the **Fairlight CMI Series III**. Samples include 20 kick drums, 40 tom toms, 50 snare drums, 60 cymbals, tympani, gongs, chimes, latin drums, tabla, triangles, plus rhythm and bass guitars. The volume utilizes 161MBytes of total RAM, utilizing four data cartridges (streamers). The suggested price is $1600. For more details, or to place an order, contact: *Sound Genesis Corporation, 7808 Creekridge Center, Minneapolis, MN 55435. Tel: (612) 944-8528.*

Mac'ers who own samplers take note! **Optical Media International** has announced two new CD-ROM discs in the Sound Designer sound file format. The **CD-ROM Digital Sound Series** works with the Macintosh and a compatible CD-ROM drive. Volume 1 of the two-part series features sound effects and percussion, and Volume 2 has musical instruments, synthesizers and percussion sounds. Each disc contains approximately 1500 individual sound files. They retail for $595 each.

Also from OMI are a series of custom **Virtual Instrument banks** for the **Emulator III**. The instrument and sound effects banks reside on floppy disks and include violin, celtic harp, zither, synth, tambura, sitar, and even psaltry sounds. The samples are available for $95 per bank from authorized OMI dealers. For more information, contact: *Optical Media International, 485 Alberto Way, Los Gatos, CA 95032. Tel: (408) 395-4332.*

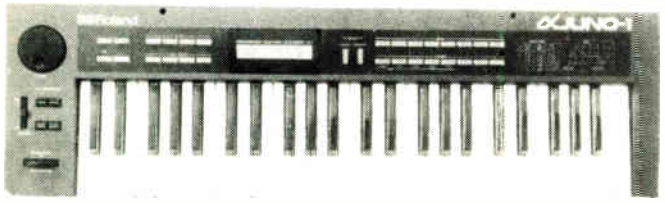
Finally, for all you diehard CZ fans, **Adrien Nash Creations**, has announced a **Commodore 64 disk with 55 banks of CZ patches**, as well as personalized versions of several public domain CZ librarian programs. The disk also has three menu/loaders to work with various types of C64 interfaces and at $75, seems like a great value. Contact *Adrien Nash Creations, 101 W. Fremont Square, Montebello, CA 90640*, for more information.

Until next month, keep tweaking. ■

ROLAND ALPHA JUNO I

The Juno Collection

J. Dixon, Cleveland, OH



Mr. Dixon sent us a whopping sixteen patches for his beloved Juno, of which we've selected what we feel to be the best eight. We'll leave it to J to describe the charms of his contributions (for which he's won a year's subscription):

- **Soft Tone (A):** Ideal for slow backing, use in stereo for more depth.
- **Synth Flute (B):** Will cut through the mix without being too harsh.
- **Seq One (C):** A must for Vince Clarke, Erasure impressions.
- **Electro Bass (D):** OSCar-type; warm, wide and fat.
- **UltraBass (E):** "Old-fashioned" bass as used by Ultravox.
- **Japanese Piano (F):** Delicate and warm, yet still brilliant.
- **Bell Chime (G):** Rich, bright, solo sound.
- **Emulator II (H):** Cold, almost metallic; typical of Emulator strings. ■

Parameter	(A)	(B)	(C)	(D)	(E)	(F)	(G)	(H)
DCO Range	8	8	8	16	16	4	4	8
DCO LFO Depth	1	0	17	15	0	9	9	10
DCO ENV Depth	0	0	0	0	0	0	0	0
DCO ENV Mode	D	⌐	⌐	⌐	D	⌐	⌐	⌐
DCO Aftertouch	2	0	4	2	4	4	0	2
DCO Bend	2	2	12	12	2	12	2	12
Waveforms:								
Pulse	3	1	3	3	3	1	3	3
Sawtooth	1	0	3	3	3	0	4	0
Sub Osc	4	0	0	0	0	0	4	0
Sub Level	0	0	0	1	0	3	3	0
Noise Level	0	0	0	0	0	0	0	0
PW/PWM Depth	90	23	97	102	105	53	0	62
PWM Rate	95	112	78	62	62	95	0	86
HPF Freq	2	3	1	1	1	0	0	1
VCF Freq	59	76	102	29	105	68	118	107
VCF Resonance	0	0	12	0	49	0	0	7
VCF Env Depth	28	21	127	57	5	97	127	0
VCF Env Mode	⌐	D⌐	⌐	⌐	⌐	D⌐	D⌐	Dyn
VCF LFO Depth	0	0	19	3	1	6	32	1
VCF Keyboard Follow	8	8	9	15	10	8	8	8
VCF Aftertouch	2	2	4	0	0	2	0	3
VCA Level	127	105	97	101	127	106	76	101
VCA Envelope	D⌐	⌐	⌐	⌐	D⌐	D⌐	D⌐	D⌐
VCA Aftertouch	4	0	10	0	7	4	0	3
Chorus	ON	ON	ON	ON	ON	ON	ON	ON
Chorus Rate	80	66	81	90	86	80	76	77
LFO Rate	88	0	89	83	85	88	82	89
LFO Delay	68	0	27	38	0	70	27	78
ENV Time 1	44	48	01	3	0	0	0	0
ENV Level 1	83	0	127	127	72	127	127	93
ENV Time 2	0	12	0	46	113	56	105	63
ENV Level 2	71	103	127	127	56	73	112	45
ENV Time 3	127	75	34	51	113	72	85	72
ENV Level 3	72	35	0	103	57	67	0	45
ENV Time 4	63	52	43	49	52	73	70	76
ENV Keyboard	4	0	1	0	9	6	0	3

READERS' TAPES

MUSIC

If you can't hum to it, is it valid? *Reviewed by Yung Dragen.*

Youth Decay at rest.

I WAS SITTING in one of those cute brass-and-fern clubs in Santa Monica with the former keyboardist from the Irish band Blue Movies, continuing a debate on what attracted us to various pieces of music (she despises a lot of what I listen to, and makes no bones of telling me about it). She had previously put forth the premise that if she couldn't hum it, she couldn't see the point. I regularly countered that there are whole temples full of music that is interesting, emotional, and purposeful (in either concept or execution) that you can't hum.

Lo and behold, a band comes on stage that *looked* interesting in concept, but executed in the most boring, unmemorable fashion possible. There was a drummer, a percussionist (with all sorts of metals and gongs), a five-string bassist, a keyboardist (with an Oberheim Xpander – I mean, such an open-ended instrument guarantees good music, right?), a guitarist with a full rack (more guarantees), and an EVI player with another Xpander (oh, joyous technoverload). The result? More boring light electric jazz.

One such tape I loved in concept but couldn't bring myself to like in reality was *You Do What You Gotta Do* by "electrical engineering student and multimedia troublemaker" **Youth Decay**. A few of the pieces are exercises in wildly feedbacking and punch-edited delays and reverbs, including the intense 'Chelation (A Heavy Metal Detox)' (none of which is nearly as controlled as David Myers back in July); two neoclassical (hah! that word again) snippets ('Prelude' and 'Postlude'); five techno-pop tunes straight out of the early '80s (kinda like Telex, but a little rougher) – and that's the first side. The second side ('Front Back Right Left') is taken up by 45 minutes of nighttime suburban ambience (cars, TV set heard through a window, background noise – quite relaxing, actually, if you're from the Midwest, like I am). I mean, there's yin and yang, but Youth is downright schizophrenic! The technopop has some good ideas (sped up tapes, vocal processing) but is damaged most of the time by resonate synth noodling on top. Youth, I like the concept – now deliver.

Even harder for me to deal with than the above is when a tape comes with a letter thanking me for this column, and then I don't like the music. Such is the case with **Michael Guy** and *All My Dreams –the MGM Sampler*. Michael uses a SixTrak (same as Youth, but Michael has occasionally done some nice programming), a Yamaha piano, TR505, and occasionally an acoustic guitar to create pop/vocal instrumentals and jazz/synth instrumentals. If you've walked into some bar with less than a half dozen people in it and a musician set up in one corner crooning away solo, you've heard the vocal numbers. Most are piano-based wanderings with the vocals absolutely submerged in echo and mixed below the instruments. No humming here. The instrumentals are soothing and actually a good bit more interesting, but suffer from more underwater production. To be fair, Michael seems to have a realistic approach: "I now feel this tape is history for me and am currently writing another album seeking newer and more stylistically consistent results."

Somewhere somebody said something about the meek inheriting the earth. **John Kimball** wins that award this month. His three song *Island Song* demo-tape came with the simplest promo literature and simplest equipment list (DX7, RXIS, SPX90, Aphex, four-track cassette), but it shows some of the most consistent talent. There's nothing real fancy here – DX bass and clav, fast drum programming, a Ritenour-like guitar solo by **Roger Feldman** on 'Nine Lives', and an equally speedy sax solo by **Rick Humphries** on 'Let Freewill Sing,' but all cook, and all are pleasant. Now, mind you, this is wallpaper music again – no humming the melody afterwards, and nothing to make you fall out of your yoga position – but it's tastefully done.

Another tape plops into the cassette deck, and *whoosh!* – a string chord bigger than that created by any debasing, mind-altering.

Michael Guy

consciousness-expanding drug, followed by arpeggiating bells and synthesizers, and you *know* we've got another new age tape this month. This time it's **James Hardman** and *Lapis Lazuli*, "transformative music for electronic instruments." Okay, I should take a break here for a minute and get off of new age's back, because the original *concept* is good (music for healing, meditation, or just

plain relaxation) and it's currently getting ripped off by record companies who sense a Big Thing in using this label to market all sorts of insipid light jazz and worthwhile neo-classical electronic music to yuppies. James is in the Paul Horn/Kitaro "real" new age mold, but with fewer changes (just chords; instrumentation rarely varies). Too beatific for me; if you like 30 minutes of floating strings (the tape's 40 minutes long, and the second side starts out interesting with minimal bells and African drums, but reverts), it may be just right for you.

Some people use self-restriction and discipline as a path to enlightenment. **Paul Stone** has done that on his *Music for a Harmonic Convergence*: it was all performed on a PPG Waveterm A computer (sequencer/8-bit sampling brain), Wave 2.3, and Expansion Voice unit, and mixed straight to cassette with no effects. To quote Peter himself, "What's so big about that you might ask? Well, after realizing no one really uses the PPG Waveterm computer because of its ability to scare anyone off at a distance, I decided to make it *my* machine. Making sequences on this machine is as easy as breaking eggs. Unfortunately, if you don't like eggs . . ." I *like* eggs, but the problem with this tape is that these seem to be Stone's first compositions with the unit, performed in chronological order. The entire first side is really too sparse and boring; the second side picks up and ends with a rousing rock instrumental ('Realization').

Stereo was faked by having the same sample come out of opposite channels at the

Damien Gossett/Robert Tabb's R,r(ar)

same time with a slight internal time difference. The PPGs appear to be quite noisy, and really need a single-ended noise reduction unit to quiet them down. The first side also suffers from bad loops on samples and the PPG "gulping" when notes were retriggered or stolen (for some reason, the second side was better – again, experience, I bet). The eight-bit drum samples hold up very well (a favorite was a *very* low Simmons tom). Some of the bass is awesome. The PPG has always had a unique sound; the strength of that sound made me not miss external processing. However, I think Paul underused some of the synth sounds in favor of trying out samples. As I've told others, I want to hear your *second* tape. For the rest of us, let's remember that treating your synth rig as a closed system like a guitar or saxophone is an equally valid path to enlightenment –it just means you have to work.

A second tape I asked for and received is *R,r(ar)* by **Damien Gossett** and **Robert Tabb**. Robert is a modern poet; Damien is an escapee from a rock band that went minimalist/dadaist. Their duets consist of processed vocals on modern urban themes ('Young Widow,' 'Sold Brother,' and my favorite, the ominous 'Korea') over dark and stark electronic rock foregrounds and backdrops. The entire second side is Damien's chance to stretch out instrumentally, and he does so with jerky rhythms, strange, moody ornamentations on top (you should hear 'Duck Hunting in Hell'), and an overall "bassy" feel. These guys just plain don't seem to have a good time – and mean it, too. If you'd like a dose of contemporary poetry and slowed-down experimental rock, check it out. ■

Contact addresses:
Youth Decay, 3212 Skycroft Drive, St Anthony Village, MN 55418.
Michael Guy c/o MGM Music, 6 Villa Court, DeLand, FL 32724.
John Kimball c/o JWK Studios, 9805 Brookford Road, Potomac, MD 20854.
James Hardman c/o Anahata, Route 1, Box 952, Eastsound, Orca Island, WA 98245 ($9.98 list).
Peter Stone, 1701 Clinton Street, #323, Los Angeles, CA 90026.
Damien Gossett/Robert Tabb c/o Verdant Ventures, Inc., 2736 Lyndale Avenue South, Minneapolis, MN 55408.

Send your demo or finished tape along with some biography/equipment details, a recent photo, and a price (if you dare sell it) to: *Readers' Tapes, Music Technology, 22024 Lassen St., Suite 118, Chatsworth, CA 91311.*

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MUSIC TECHNOLOGY's free classified section is designed to help private sellers find buyers and bands find other musicians. If you buy and sell musical equipment or sounds as part of a small business, please contact the Advertising Department to place a regular classified ad.

KEYBOARDS

CASIO CZ101, Korg EX800, excellent condition, package deal, $500. David, Tel: (704) 873-5127.

KORG POLY 61 with MIDI, excellent condition with cover, $400. Tel: (714) 680-3545.

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PPG 2.3 with Waveterm, $4500 obo; Roland Juno 106, $550 obo with case. Andrew, Tel: (718) 229-5057.

SYNERGY II plus MIDI 4-track sequencer, 5 cartridges software, pedal, $1250 obo. Scott, Tel: (312) 866-6732.

YAMAHA DX7, volume, modulation and sustain pedals, Bo Tomlyn ROM, IRAM 320 sounds, case, etc, Multi-Track keyboard, case, $1800. Tel: (216) 228-1152.

YAMAHA DX7 loaded, Expansion, 320 internal programs, modified MIDI, blue backlit display, $1100. Tel: (516) 221-6945, leave name and phone number.

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SAMPLING

AKAI S700 sampler with individual ¼" outputs, $850 obo. Tom, Tel: (901) 276-6967.

CASIO FZ1 16-bit memory, expansion over 30 (thirty seconds), case, etc, $1150 obo. Pat, Tel: (818) 508-8652.

CASIO FZ1 sampler with 2 Meg update, anvil case, 35 disks, mint condition, $1995 obo. Dan, Tel: (612) 542-9108, leave message.

E-MU SP12 sampling drum machine with turbo update, Commodore disk drive, roadcase for both, mint, $1000. Tel: (814) 868-3122.

E-MU SP12 with turbo version 2.6 with ATA case, plus Commodore 1751 disk drive with case, make offer. Tel: (412) 835-4703.

FREE MIRAGE DISK call M.U.G. Hotline for details. Tel: (914) 963-1768 or write G4 Productions, 622 Odel Ave., Yonkers, NY 10710.

KORG DSS1 sampling keyboard with 70+ disk library, asking $1550. Tel: (804) 442-5004.

KURZWEIL K250 sampler, MIDI controller, sound generator, fully loaded, includes QLS System/Macintosh, cables, stand, $8500. Eric Heiberg, Tel: (804) 358-3852.

ROLAND S50 OWNERS want to trade non-factory samples? Please contact Bruce Pelley, Box 444, South Lancaster, MA 01561.

SEQUENCERS

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LINN 9000, mint condition, sampling, memory expansion and latest software updates with flight case, $3250. Tel: (818) 705-0476.

ROLAND JSQ 60 for Juno 60 and Jupiter 8 in box, never used, $50. Mirek, Tel: (212) 799-7595.

ROLAND MCS00, 4 tracks, 16 channels, disk drive, 25,000 notes, editing studio use, perfect condition, $1000 obo. Rick, Tel: (513) 241-1600.

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SEQUENTIAL TOM drum machine, mint condition, all manuals, $450. Tel: (615) 385-0169.

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SHAREWARE MIDI SOFTWARE for Apple Mac, Atari ST, or Amiga, SASE for list, specify which computer, $4 disk. Jan & Russell, PO Box 1646, Orange Park, FL 32067-1646.

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ROLAND DEP5, $600; Crumar Orchestrator, $150; Peavey T40 electric bass and case, $300; Tapco 6-channel mixer, $120. Rick, Tel: (818) 841-6081.

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