MUSIC
TECHNOLOGY

SCRITTI POLITTI

Live is Dead

CONTEMPORARY CHRISTIAN TECH

Amy Grant & Michael W. Smith

DIGITAL AUDIO 101

Bare Bones Theory

Computer Notes Cover
Colin Cantwell © 1988 Crystal Chip, Inc.

EDITORIAL
Just Playing

BELIEVE IT OR not, I thoroughly enjoy working with electronic musical instruments and computers. I like to tweak synth patches, work on programming some of my own, compose with sequencers and engage in all the other tasks which were added to a musician's expected skills with the massive acceptance of synths and MIDI equipment. Sometimes I get so caught up in the programming of various pieces of equipment, though, that I don't leave myself the time or energy to do what these tools are supposed to actually make easier and more enjoyable: play!

Part of the problem stems from the fact that I don't have particularly good technique on keyboard (I grew up playing trombone but I haven't seen too many worthwhile trombone-to-MIDI converters - at least, not yet). As a result, the way to produce the music I hear in my head is to do some serious sequence programming, editing, etc. Of course, the capabilities of the equipment are making it easier and easier to achieve acceptable results in this way (for which many people are *very* thankful), but you still miss something when you create music in step time. Besides the fact that you often get a mechanical feel, you lose the sense of spontaneity that occurs with live playing.

You also lose that sense of joy which comes from simply playing a musical instrument. No matter how well (or poorly) you play, there's something magical about creating your own sound and your own music. If you work with electronic instruments exclusively, and thus need to concern yourself with programming, it's sometimes easy to overlook that idea. I think every now and then you just need to say, "The hell with programming, play!" and then do just that. If you're not sure of what sounds to use, go ahead and use the factory presets - and don't be ashamed of it - there's no law that says that you have to have all of your own customized, personally programmed patches to make good music.

I realize that to many of you that sounds incredibly obvious, but occasionally it seems like actually playing an instrument is almost the farthest thing from my mind when I'm working with it. It's a nasty trap which I think others fall into as well.

Part of the problem, it seems, stems from the nature of the gear itself. I do have passable guitar chops and access to a guitar-to-MIDI converter, so when I get the urge to jam on my toys I just pick up my Casio MG and go. Before I can get anywhere, however, I have to make sure my switcher is set to the right patch, that all the synths are on the appropriate reception channels and in the proper mode, and that they're playing the patches I want to hear. Not a big deal really, but it does make me appreciate my acoustic guitar a little bit more every time.

Of course, once you do get things set up, synths and samplers provide a type of instant gratification which you just can't get with other instruments. Hit one note on that mondo pad and you're in seventh heaven. (I'm not going to even attempt to address the question of whether or not that capability for instant gratification is a good thing or a bad one - I'll save that for a day when I'm feeling particularly masochistic.) And having that thirty-second note run come off cleaner than humanly possible from your sequencer can be extraordinarily cool. Many people are starting to shy away from overprogramming, though. It seems like now that they know they can do it and have heard it done, they've gotten sick of the notion and just want to play. It's an interesting backlash; though I suppose it's not that terribly surprising. (History does have a way of working in a pendulum-type fashion.)

Not everybody is moving away from extensive programming and back to live playing, though, as this month's interview with Scritti Politti so pointedly relates. It seems everyone needs to find the system which works best for them. Just don't forget that playing music is what it's all about. ■ *Bob O'Donnell*

World Radio History

IN THIS ISSUE

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Programmed for everything but obsolescence.

When the Kurzweil 250® was introduced in 1984, it set the industry standard for sampling excellence and programming sophistication. But what may have been a crowning achievement for others was just a starting point for us. The K250®, you see, is being updated continuously to keep pace with future technological advances.

For example, we've recently developed powerful, portable RAM cartridges that vastly extend the K250's memory. We've just released our ninth sound library and have more on the way. Also, the option of having twelve separate outputs for unprecedented versatility in using those renown K250 voices is available.

Things have changed a lot since 1984, but the K250 and 250RMX® (rack mount expander) are still on the leading edge of music technology. So when you buy a K250, you know you're investing in an instrument that will continue to earn the admiration of music makers everywhere.

Visit your nearest Kurzweil dealer today. By the time you get there, he'll have more exciting updates to tell you about.

KURZWEIL
Music Systems

NAMM FLASH!

Just prior to our leaving for NAMM, two manufacturers who aren't attending the show announced some intriguing new products. First, Korg introduced the Z3 MIDI guitar system, which consists of a pickup-like interface that can be mounted on any guitar and a separate converter unit. The converter also includes a built-in six-voice synth (four operator, eight waveform à la the TX81Z) with 128 programs. The Z3 can work in a basic and advanced mode (for more serious tweaking) and operates both in MIDI Poly and Mono Modes. It is also compatible with existing guitar synth systems.

New from Roland are the E10 and E20, the company's first home synthesizers. Making use of the same L/A synthesis technology as found in the D10 and D20, the new E series of instruments come in a completely redesigned case and include built-in speakers. In addition, both instruments include a number of intelligent auto-accompaniment and auto-play features, such as the ability to create basslines or melodies based on the chords you play. The improvised lines can also be sent out the instruments' MIDI ports. The E10 is expected to list for around $1500 and the E20 for about $2000.

MORE FROM Korg USA, 89 Frost Street, Westbury, NY 11590. Tel: (800) 654-3188.
RolandCorp US, 7200 Dominion Circle, Los Angeles, CA 90040. Tel: (213) 685-5141.

EVERY WHICH WAY

Hot off the press: Music Systems Inc of Salt Lake City, a new manufacturing company, has announced the introduction of the MIDI Routing/Merging unit and the MIDI Processing unit, designed to get the right signals to the right places.

The MRU16 (MIDI routing/merging) is a 16×16 MIDI management unit, which can be expanded to a total matrix of 64×64 by cascading four of the devices together. Each output can have an eight-character name and can utilize up to four inputs each. 128 patch settings can be stored by name.

and each patch is accessible by MIDI program number, MIDI program number mapping, front panel switch or MIDI Time Code.

The MPU16 (MIDI processor) is a 16 in/16 out device offering data filtering, message filtering, keyboard layering, chord generation, data mapping, data scaling, data offset, reverse data around break point, limit data range, MIDI delay, channel output assignment, channel offset, and (whew!) output port assignment for each input and output. Memory stores 128 patches by name, selectable by MIDI program change. MIDI program change map, front panel switch or MTC sequencing.

Although retail prices were not available at press time, the inside scoop is that the MRU will go for less than $700; the MPU16 for less than $800.

Coming soon from MSI are a 16-track MIDI storage and setup device and a 16-track full function sequencer. All products are expandable via a LAN interface to 64 ports and 64 tracks. Love to see the system that can use all of this!
MORE FROM Music Systems Inc, 47 W. 200 South, Suite 305, Salt Lake City, UT 84101. Tel: (801) 355-2048
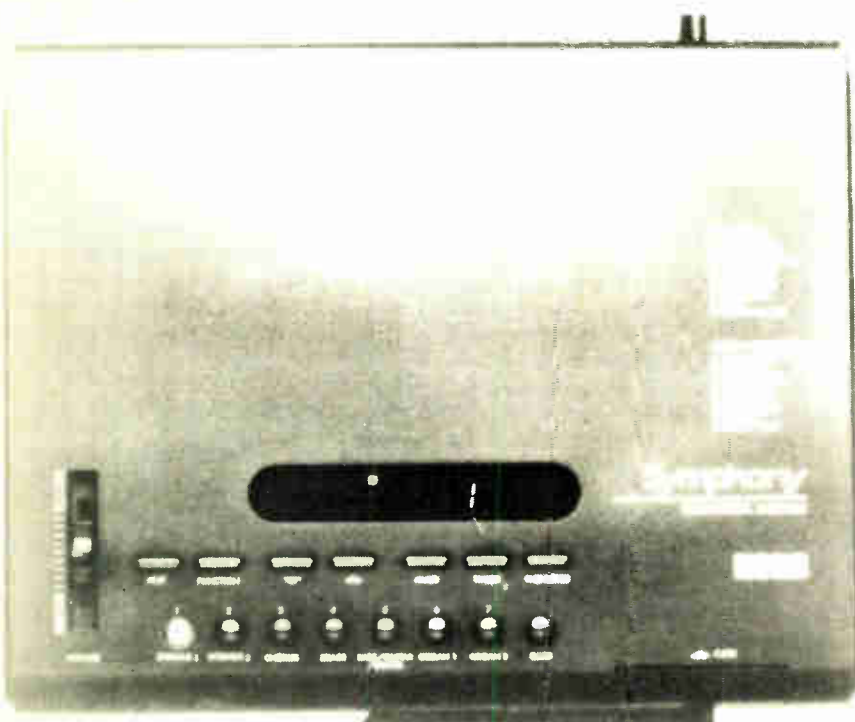
Symphony, one of Korg's newest modules, promises multitimbral access to sampled ensemble and orchestral sounds.

MULTIPLYING MODULES

If you like the looks of the M1 and S1, you'll love the similar sleek design of Korg's two new modules, the P3 Piano Sound Module and Symphony.

Symphony features seven selectable on-board sampled sounds, including strings, brass, choral, organ, bass and guitar, and promises high-capacity ROM cards to fill in with acoustic instruments and synthesized sounds. Up to eight different sounds may be played simultaneously, and layer and split functions provide flexibility in piling up the sounds.

The P3 Piano Module produces sounds of two different acoustic pianos as well as of several electric versions. Features include 16-voice polyphony, a slot for additional ROM cards, and several operating modes – normal, split and multitimbral.

Though Korg doesn't offer suggested retail prices any more, these units are both being touted as extremely affordable. A call to your local music store will probably get you the whole story.

MORE FROM Korg USA, 89 Frost Street, Westbury, NY 11590. Tel: (800) 654-3188.

MIRROR, MIRROR...

And here's another guitar controller to keep your eyes peeled for... the Mirror 6 from Zeta Music Systems.

This new guitar MIDI controller utilizes fret-scanning, purportedly allowing the unit to trigger within a half millisecond. It can be set to trigger a MIDI note on command as soon as a string makes contact with a fret, or it can be set to expect a string to be plucked before a MIDI note on command is used. Pitch extraction circuitry is then used to analyze the pitch to accurately respond to bends, glissandos and tremolo tricks. Sounds good...

Each string can send Note On, Velocity, Key Number, Bend, Continuous Controller and Program Number information, and each can be assigned to its own MIDI channel or all six can be assigned to one channel. Four knobs control Master Volume, Mix, Guitar Tone, and a MIDI assignable soft-knob, and switches include Synth/Guitar/Both, three-way Mag Pickup and a Rhythm/Neck/Lead selector. There are optional velocity-sensitive Expression Pads with polyphonic aftertouch, a Breath Controller Interface, the AmpTrak Synthesizer Dynamics Processor, a hard shell case and the FS64 Footswitch.

The suggested retail price is $1495 for the Mirror 6; $1695 for the Zeta standard

The Mirror 6 combines fret-scanning with pitch extraction, providing guitarists control over their MIDI systems.

guitar; and $1995 for the Zeta Deluxe (which sports a Kahler Tremolo bridge).

MORE FROM Zeta Music Systems Inc, 2823 Ninth Street, Berkeley, CA 94710. Tel: (415) 849-9648.
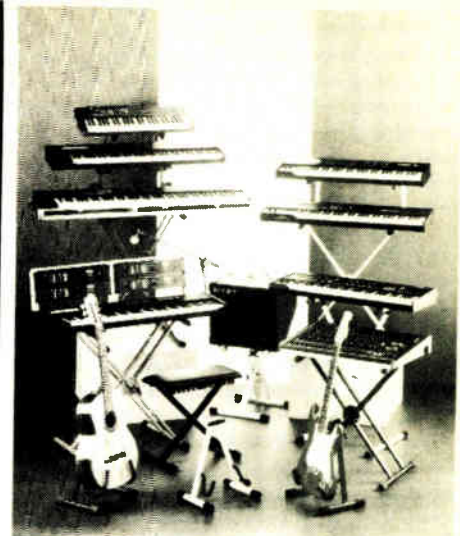
Looking for support? Check out the Quik-Lok stands.

STANDING ROOM ONLY

Six new models of stands have been added to the Quik-Lok series offering a pretty good selection of styles and prices.

If you like what you see in the photo, you might just check this out – as well as the new introductions of a guitar stand and one for smaller PA systems and amplifiers.
MORE FROM Music Industries, 100 Fourth Avenue, Garden City Park, NY 11040. Tel: (516) 352-4110.

TICKLING THE IVORIES

If you came up in music by playing wooden or ivory keys rather than plastic laminates, your interest may really be sparked by KTI's latest products: the GZ Performance Keyboard Series.

KTI claims that the GZ1000 contains "true piano hammer action," provided by adjustable mechanical and software user adjustments for the feel of the keyboard. Percussive key sensors and MIDI operator control system allow the player to design multiple keyboard response parameters, which are simultaneously transmitted over MIDI. An edit footswitch allows the keyboard to control data entry, selecting edit functions and cursor movement. A 3½" disk drive will dump and save thousands of GZ MIDI setups (globals), and system exclusive synth patches help store and manipulate extensive sound libraries.

The software-controlled rear panel MIDI ports feature four Uarts with eight

individual MIDI Out/Thrus, two MIDI Ins (MIDI merge), three programmable footswitches and four programmable MIDI foot pedals.
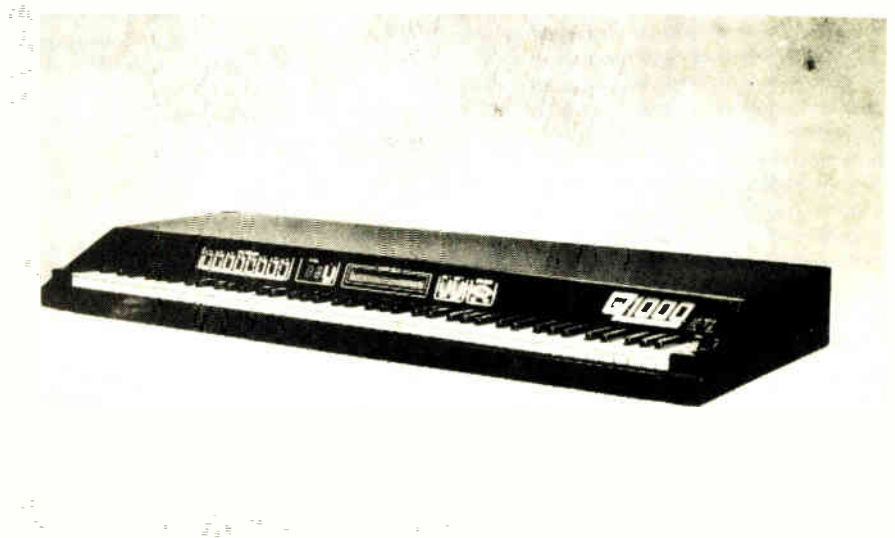
Is it real or is it plastic? The GZ Performance Keyboards promise true piano hammer action.

Price was unavailable at press time.
MORE FROM Keyboard Technologies Inc, 16137 Sherman Way, Suite 169, Van Nuys, CA 91406. Tel: (818) 891-6999.

ATTACK OF THE KILLER OCTOPUS

Scholz Research and Development has announced the development of the Rockman MIDI Octopus, designed to place Rockman Rockmodules and other effects

devices under programmable MIDI control.

The Octopus is capable of controlling up to eight footswitchable functions and storing 100 effect-combination programs – all in a half-rack module.

Programs may be accessed via MIDI foot

controller, keyboard, sequencer or front panel controls. The price is still to be announced.

MORE FROM Scholz Research and Development Inc, 1560 Trapelo Road, Waltham, MA 02154. Tel: (617) 890-5211.

D! DRIVE

Grey Matter's latest, D!, is an internal 3.5" floppy disk drive for the Yamaha DX7IID. The kit contains all the materials needed to install its disk drive into the cabinet of the DX7IID, including a new plastic side piece. Once the drive is installed, the DX7IID becomes fully compatible with the DX7IIFD, allowing the user to store Voice and Performance data, Fractional scaling, micro tunings and MIDI data from other instruments.

Whether you can install it yourself is another question . . .

The suggested retail price is $399.

MORE FROM Grey Matter Response, Inc, 15916 Haven Avenue, Tinley Park, IL 60477. Tel: (312) 349-1889.

VISUAL BASICS

Want to learn more about MIDI without reading a few thousand pages? Silver Eagle Inc has announced the availability of a new instructional video, *Making the Most of MIDI*, promising practical information on setting up and using a MIDI system.

Among the topics covered are Basics, Devices, Using MIDI and the Language of MIDI, the latter dealing with all the terms for MIDI Data and setups. Included with each video is a MIDI technical reference chart, troubleshooting tipsheet and MIDI implementation chart. The running time is 60 minutes, and guest appearances include visits from Stanley Clarke, Christopher Cross, Larry Williams and Michael Bernard. The "class" is taught by Marc Mann, a performer and programmer in the music industry.

The suggested list price is $49.95.

MORE FROM Silver Eagle, 6747 Valjean Avenue, Van Nuys, CA 91406. Tel: (818) 786-8696.

OPEN AND SHUT CASE

FSK, the FootSwitch Kit from Mescal Music, is capable of converting normally-open footswitches to normally-closed footswitches, and vice-versa. The FSK fits inside of the Yamaha FC4 sustain pedal, but works with most other pedals as well.

This product may be especially interesting to Ensoniq buffs, allowing the Mirage, ESQI, SQ80 and EPS to use the FC4 as a sustain pedal. The FSK comes assembled and tested and works with a 9V battery.

The FSK with instructions sells for $12.75.

MORE FROM Mescal Music, PO Box 5372, Hercules, CA 94547. Tel: (415) 724-0804.

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READERS' LETTERS

Dear Music Technology,

In regards to Chris Meyer's editorial, June '88:

OK, let's admit this: "let's all . . . admit it – we don't want copy protection because we want the software for free" was aimed purely at generating maximum controversy. Perhaps the magazine is not happy with the size of its "Readers' Letters" section? The point is that if we all swallowed an "honest pill," the answers wouldn't be the same. If you truly feel this way, well, you must hang out with an interesting bunch of fellows.

With this, there's an implied "the only thing wrong with copy protection is that we can't get the software for free." Copy protected software is a pain, even with the hard disk installation capability. The programs are slow to load (since they need to be unraveled from the encrypted code on disk), and are a headache for backing up and optimizing your hard disk. I write software for a living, and so I'm naturally biased a bit heavier than some towards actually (gasp!) paying for the software I use. I don't like copy protection, but I am not completely insensitive to the plight of the developer. Still, there's something quite magical about paying $400 for a piece of software and receiving those sage words: "Please Insert Master Disk."

You imply that computer manufacturers condone copy protection, and that sampler manufacturers are negligent for not doing the same. Computer operating systems are not "built to allow copy protection." I don't even know of one that makes it easy for a vendor to copy protect his software. Rather, the protected program is perverted in some manner that makes it difficult for the operating system to deal with it normally, without help from the program itself. What makes it difficult to protect patches and samples is that they are just data files, not programs, and copy protection would require the operating system software to be modified.

Anyway, I'm beginning to suspect (based on its popularity as a topic for editorials in music magazines) that the copy protection subject is mainly useful as a marketing ploy for the magazine. Heck, it got a lot of print in Keyboard, didn't it? It is reminiscent of computer magazine editorials of a few years ago. This kind of editorial serves only to annoy those who pay for their software by throwing everyone in one big vat labeled "untrustworthy." All this is rather pointless. The argument won't be resolved by editorials and letters to the editor. It will be resolved in the market place itself. I don't have to be concerned about whether a word processing, drawing, painting, spread sheet, programming language, and any general utility or business software package for my Macintosh is copy protected, because

virtually none of them are. The battle has already been decided there. Copy protection is only tolerated in relatively new and uncrowded markets. Who knows, perhaps, as you suggest, the budding music software market needs this protection. As far as a sampler manufacturer building copy protection into his product, I don't know of anyone who is willing to voluntarily commit suicide for the cause of third party patch developers, do you? Let's take a reality pill, shall we?

Nigel Redmon
Torrance, CA

I'm always amazed at how rabid anti-copy protection people are. Their arguments are pretty interesting, too – like complaining about the extra minute (tops) it takes to boot a program they're probably going to be running for at least an hour or so.

Did I write the editorial just to create controversy? No; I wrote it to shed some light on an ignored subject. And so far, every reply we've received has attacked the problem from a different angle (which has been fun). You'll be seeing more replies in these pages in the near future.

I'm not sure people who abide by copy protection feel they are getting lumped in with those who abuse it, or are just as angry at me (and manufacturers) as at the abusers. Copy protection has been getting dropped as prices come down, with manufacturers hoping to make it less worth the effort of copying a program and dealing without the manual (I thought Southworth's MIDI Paint was an interesting first step in this direction in the sequencer market segment). For an analysis on why others have been dropping protection, see an article on the subject elsewhere in this issue. However, sounds are already so cheap, dropping the price further may not be an option (and there's no manual or customer support to factor in).

And as far as nobody voluntarily committing suicide by copy protecting their sounds, read on . . . – CM

Dear Music Technology,

As a software and sound disk developer I must agree with Chris Meyer's June '88 editorial about copy protection. Unauthorized copying at music stores as well as by purchasers of my products threatens to put me out of business completely. The only way to beat these jerks who copy software is to simply not offer my product in the first place. So who wins?

I was intrigued with Chris' idea to build copy

protection into synths and samplers. I did just that over a year ago. I rewrote Ensoniq's Mirage operating system to include a foolproof protection routine which would allow normal loading and saving of unprotected sounds. Any attempt to save the sound to another disk would result in the copy being completely scrambled.

I offered to license this protection routine to all the major Mirage sound disk developers at that time. Their initial reaction was very positive, but apparently after consulting their marketing people, nearly all of them decided against the idea since it might be seen as a negative factor by potential customers, and they also would have to provide a back-up service for those who did buy their disks.

So my company and another developer in New Jersey were the only companies with enough guts to actually market this protection scheme, and we're both still using it today with very little negative feedback from our customers. Oddly enough, all the other companies I approached a year ago have now either gone out of business or are not selling Mirage disks anymore.

This same sort of protection can easily be incorporated into any synth or sampler's operating system with a relatively small amount of development time. A carefully designed system would allow the owner to easily configure his library of sounds as he wishes, so long as he doesn't try to swap his new sounds with his friends!

Steven Fox
Leaping Lizards
Seattle, WA

Brick throwing to be continued . . .

Dear Music Technology,

First, I've been reading/subscribing since issue #2 or 3, and for the most part good work! As with anything, there are things that I find interesting and things that I find not so interesting.

One thing that I would like to see discussed is how to use System Exclusive and System Exclusive tables, but in a way that musicians, not computer programmers, can understand. Books like Ferro Technologies' "The MIDI System Exclusive Book" are good and somewhat helpful, but they are pretty difficult to decipher in terms of all the different bits and bytes (and different categories of these) if you're only your basic bonehead musician-type. So it's a wee-bit esoteric, but how about it? How about the mysteries of System Exclusive deciphered?

If this is a little bit too out there, any idea of who I can talk to about this?

Stewart Liebig
Los Angeles, CA

Nothing is too out there for us, Stewart! Seriously, though, your idea about demystifying SysEx charts is a good one; look for response to your request in the near future.

Dear Music Technology,

I would never argue with an opinion as to the likes and dislikes of an instrument's sound. Music is an aesthetic matter and as such is subject to such a personal level of interpretation that it is left up to the individual to judge for themselves. The fact that Lorenz Rychner did not like the sounds of the Keytek CTS-5000 much in his Pianos to Go article in the June issue is not the problem here. What is a problem are the gross errors and prejudicial statements that make me honestly wonder if any time was actually spent looking at the front panel, not to mention the possibly novel idea of actually playing the instrument.

To make the statement, "The keys are said to be weighted . . ." has a very strong implication that the speaker does not believe the claim. The weighted action that Keytek uses in the CTS-5000 is the same action used by several other manufacturers, including one reviewed earlier in the article. I would be happy to loan Mr. Rychner a screwdriver and an instrument so he can verify this on his own. If he does not like the feel of the keyboard, that's fine, as it is a matter of personal perference, but it was most unfair to complain about it only when it was on the Keytek and not when it is used on other instruments.

I would agree with the statement that the CTS-5000 has a basic MIDI implementation. However, if Mr. Rychner had taken the time to read the labels on the front panel controls, he would not have made the wildly erroneous statements concerning limitations on the Keytek digital piano. The button labeled "MIDI int/ext" (for internal/external) is his missing Local Control Off function. The button labeled "OMNI on/off" switches between OMNI mode I and POLY.

There are in fact 13 controls specifically for MIDI including assignable program change numbers for each of the 16 user programs, the ability to send on different channels on each side of the split, independent channel numbers for send and receive, pitch-bend receive on/off . . . basic MIDI but thorough.

I'll stop now, as this is beginning to sound like an ad and all I wanted to do was correct the misstatements in the article. I am sure that Mr. Rychner is a nice person who was just having a bad day when it came time for our review. Thanks for spelling our name correctly (as the old theater saying goes, "as long as they spell your name right it's not a bad review . . .) and mentioning that Keytek is available at your local Gibson dealer.

Richard Bugg
Gibson U.S.A
Oklahoma City, OK

Sorry to hear about the factual mistakes regarding the MIDI implementation; we pride ourselves on getting those things right. As for the feel of the keyboard, Mr. Rychner did not compare it to any other instruments, nor make any harsh judgements of it; he simply mentioned that he didn't like it. – BO'D

Dear Music Technology,

I am writing this in response to your review of our product, Concepts:One. I would like to make a correction to your article. Real time muting is possible from the Phrase and Step Editor. The usage of this feature appears in the relevant chapters in the manual for those screens. Also, those functions appear in the associated Help Menus in version 1.73.

The merge bug you mentioned in the article was fixed approximately two months ago, with release 1.73.

With regard to Matt's comments on the overall user friendliness of the program, we have taken steps to address this question. We are shipping tutorials with all of our products, and we have a new release of Concepts:One, version 2.0, which grew out of our continuing dialog with our customers. This new version addresses the need for a simpler, more intuitive user interface.

Our users comments are important, and it helps build a better program for all of us.

Chuck Eaton
MIDIconcepts Inc.
Canoga Park, CA

L E T T E R S

Introducing Kawai's Maximum music.

You'll be amazed at the natural sound, expressive power and easy-to-use functions of Kawai's remarkable K1 keyboard and K1m tone module.

The Kawai K1 is an exciting new class of digital synthesizer giving you an inspiring range of dynamic sound at an unbelievable low price.

Amazing sound that inspires the imagination.

Kawai's new Sampled Digital Waveform technology results in a rich multi-timbral sound that you've got to hear to believe!

The K1 features 32 sound sources that generate 256 different digital waveforms. Kawai's advanced digital circuitry creates a clean, well-defined sound which resonates with surprising depth and character.

From the crisp snap of an electric bass or snare to smooth sustaining strings and voices, the K1's advanced waveform samples let you master the musical possibilities.

Imagine, up to four waves combined into a single sound. The strike of a piano hammer mixed with a violin sustain. Or the breath of a flute matched with a vocal choir. Each source has independent enveloping and modulation controls to give you total control.

Become an expert at sophisticated color tone mixing.

Unlock the creative genius in you with the K1's programming versatility.

With a single patch, you can program up to 8 sounds into a combination that includes splits, layers, multiple MIDI channels and independent keyboard control.

The K1m low-profile design is ideal for desktop use.

World Radio History

A Vocal Chord
PART TWO

Illustration Sophie Laurence-Jones

With the raw vocal material safely on tape, a stunning set of vocal samples is just one operation away: transferring the voice to the sampler. Text by Tom McLaughlin.

TO JUMP RIGHT in: last month, I recommended recording every semitone within your vocalist's range while singing both ascending and descending semitone scales. With good reason too. After giving your recordings a rest and listening to them with a fresh set of ears you'll find that some notes sound infinitely better than others. Maybe some are more in tune or have a more consistent coloration. Maybe the higher register of the descending scale sounds less strained than the ascending scale. Whatever the reason, some notes will shine brighter than the rest.

List the pitches you recorded and, while listening to a rough mix of your recorded voices, note the takes that sound best to your ears. These are the ones you'll want to consider for use in your multisample.

To avoid sampling every one of these "superior" takes (even samplers at the upper end of the market have a limited amount of memory space), you'll need to whittle the intervals between the takes you're going to use down to maybe every second, major/minor third or perfect fourth. Keep an ear out for consistency in tone color from one selected take to another, paying special attention to the

transition area between your vocalist's chest and falsetto registers.

Mixing for Sampling

THIS IS VERY similar to audio mixing. If you're going to be sampling in mono, and I'll assume that you are, be sure to monitor your recordings in mono, so you'll get a good idea of what's going to end up in your sampler. If your sampler has a monitor output, by all means monitor your recordings using that. If you're going through a mixing board, set the equalization flat, and turn off all the effects.

If you multitracked your vocalist's takes, route your multitrack tape recorder's outputs through individual channels of a mixer, and start playing your vocal recordings from the beginning of the tape. Listen to the first vocal track and while the tape is running, bring up the level of the second track until a good balance between the two is obtained. For a homogenous ensemble mix, repeat this with the other tracks so that no one track predominates.

Eq & FX

QUITE A LOT can be done with equalization. It all comes down to personal taste, but with EQ

you can manipulate vocals to make them sound close in or far away, full or airy, subtle and sexy or tinny and thin. It's conceivable that several sets of vocal samples can be obtained from one set of recordings, just by using different EQ settings.

When pushing the top end of your recordings with EQ, or cranking up the "drive" on your psychoacoustic enhancer (aural exciter) to bring out clarity or breathiness, it's a good idea to put a low-pass filter between the mixer and sampler, and set it to let only frequencies just below half the sampling rate through (perhaps use a graphic and slam down all the sliders for frequencies above this magical mark). This way, you can do whatever you want to the top end, while preventing any possibility of aliasing.

If you hadn't used any reverb while recording your vocals and feel the need for some "space" around them, now's the time to add it. A good vocal plate or hall setting, with a little top and bottom end rolled off, will make your samples sound more lush and professional. If you're multitracking and put a different room around each vocal pass while recording to add an individual character to each track, routing them all through the same room now will help to homogenize the lot.

Sampling

NOW TO THE serious sampling stuff. Make sure you have enough formatted disks on hand. You'll need at least three for the work ahead of you; one for storing your unedited source samples, a working disk, and one to save your edited and manipulated samples to. Once you've arrived at the final versions of your samples, the working disks can be erased and used for another project. Make sure to name each of your samples according to pitch, octave and whether it's from the ascending or descending scale. This is not only for keeping track of them while editing and looping, but will make life a lot easier when mapping time comes.

Your job of sample editing, looping and manipulation will be made infinitely easier and less time-consuming with some sort of visual editing. Forgetting all the fancy facilities, just being able to see the waveform you're working on will cut finding acceptable loop beginning and end points down from possible hours to probable minutes.

Be careful with input level when sampling vocal ensembles. Monitoring peaks on tape recorder VU meters and sampler input meters can be rather deceptive. As with other ensembles, there will be wide variations in level along the length of a note due to the different vocalists hovering around the central pitch (visually displayed, vocal ensembles often look like some sort of roller-coaster ride), so it's easy to run into digital clipping on the peaks. Unlike many percussive sounds, where a slight amount

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can be used to your advantage, digital clipping sounds really ugly on sustained samples, so err on the side of too little input level to be safe.

Sampling Rates

THE SAMPLING RATE(S) will be governed by the available rates on your sampling unit, the amount of memory space available on your

your sample, fade out, then reverse again to hear the result (if you have a choice of curves, use linear or reverse exponential – when turned around, it will become a natural exponential attack).

There are several methods of editing and looping vocals. The method you use will depend upon how much time you want to spend, the

although you may find that alternating loops work more successfully than forward loops with this technique.

8. Discard sample material before and after loop, then save.

– **METHOD D** is the same as above, except that you leave enough un-looped sample material at the beginning to crossfade the attack portion of the original sample through. The advantage of this is that it gives you a more natural entry to your manipulated sample. You must make sure, however, to fade out the attack portion of the original sample before the loop of the looped sample begins.

– **METHOD E.** If I have the time and enthusiasm this is my favorite method of ensemble sample looping and manipulation. It uses up the least amount of memory space, gives unrivalled loop results and gives your samples a silky smoothness. The one snag is that it requires a loop crossfade that takes its crossfade material only from before the loop. If you're clever enough, you could probably work it out for other forms of loop "blending" by keeping judicious track of loop and sample positions. This method assumes that your sampler or editing software doesn't need any sound after the loop to fade back in.

1. Make sure original sample is saved as a backup.
2. Locate smoothest portion of sample with the most consistent pitch and coloration.
3. Discard sample material before and after this portion.
4. Save this new material for safety.
5. Loop up to 50% of the later part of this material.
6. Discard all material after loop.
7. You want to be left with exactly twice the amount of material in your loop so discard any material in the front portion of your sample that adds up to more than two times the loop length.
8. Calculate and execute loop crossfade, save.
9. Reverse, save and combine with above material.
10. Calculate and execute loop crossfade.
11. Pass GO, collect $200.

As in D, you can crossfade the attack portion of the sample through the first half of this material for a more natural entry or discard the first half, leaving only the loop, to save even more storage space. This process, theoretically, can be repeated on the remaining material, dicarding the first half each successive time, until you are left with a loop only a few wave cycles long epitomizing your vowel sound. I've repeated this three or four times on samples with varying degrees of success.

Mapping

ONCE YOU'VE EDITED and looped your samples, the time has come to map them out on your keyboard as a multisample. If you haven't already, assemble all your final versions on the same floppy disk, preferably in order from low to high.

The best place to start mapping is to assign samples to pitches on the keyboard relating to the original sampled pitch. From there it is merely a matter of experimentation to see how

---

"Unlike many percussive sounds, where a slight amount can be used to your advantage, digital clipping sounds really ugly on sustained samples."

---

sampler, how many edited samples you expect to fit into that amount of memory space and the amount of top end and fidelity you require.

It's widely accepted that a playback frequency response of 3-4kHz is the bare minimum needed to sample intelligible speech. Since you're working on samples that will hopefully be more musical than 'I Speak Your Weight' scales, you'll want to work with sample rates considerably higher than this.

From my experience, you'll need a minimum playback bandwidth of 8-10kHz. If you can afford the memory space, by all means sample at full bandwidth, but for the sake of economy, a good compromise seems to be in the 12-14kHz range or 15-16kHz if you're sampling exceedingly breathy vocal samples. Remember, your sampling rate must be at least twice your required bandwidth (plus another 10% or so at the top, to allow for the anti-aliasing filter).

Take time to experiment with a few different sampling rates to see what you can get away with, and let your ears be the judge. Low-pitched samples can often be sampled at lower rates with little or no apparent loss in fidelity.

Trimming and Looping

LET'S ASSUME YOU'RE working on a set of oohs, aahs, humming and so on, so you'll want to sample as long a piece of vocal material as is practical. It's not uncommon when I'm working on a set of vocal or string samples to fill up three or four entire floppy disks with raw material. Once fully edited, these samples might take up anywhere from 75% down to as little as 15% of the original memory space.

You'll do a little of this trimming before you start looping (particularly off the beginning). Unless you like your samples *au naturel*, you can make your vocalist(s) sound a lot more polished if you fade in after any harsh or "out-of-tune" portions have passed.

Aahs are notorious for having rather rough attacks and virtually all of your takes will start out flat or sharp of the desired pitch. Not having the precision of a key or fret to rely on, we approximate the pitch we're aiming at with our vocal tract before we even open our mouths, and there's always a few milliseconds of fine-tuning that goes on with both pitch and tone color once somebody actually starts singing.

Although you can use a VCA to fade in harsh attacks, if you play a sample back at progressively lower pitches, the VCA will allow progressively more of the rough portion to pass. With the right software you can digitally fade the sample in and ensure that it has a smooth attack. If you don't have "fade-in" software, but *do* have software for "fading out," simply reverse

editing and manipulation facilities available to you and the amount of memory in your sampler. To minimize confusion, adopt some sort of numbering/filing system for your "samples in works" . . . maybe ClaI = CI ascending, 1st version.

– **METHOD A** is the most straightforward, being not a lot more than sampling common sense, and requires the least amount of time and editing software. Unfortunately it eats up memory space like there's no tomorrow, relying heavily on long loop lengths for its success.
1. Make sure original sample is saved as a backup.
2. Find suitable sample start point.
3. Locate and set passable loop start and end points aiming for as long a loop as possible.
4. If all is well, discard unused sample material and save it. If not, recall backup and repeat 2 and 3.

– **METHOD B** is pretty much the same as Method A with the addition of loop crossfading, merging, blending or whatever your sampler's software calls it. This is what I fall back on if I'm in the middle of a session and a client wants "instant" results. After 3 and before 4 in method A, making sure I have enough sample material after and/or before the loop to meet the requirements of the software I'm working with, I'll calculate and carry out a "loop crossfade" (while keeping my fingers crossed).

– **METHOD C** uses only the choicest portion of your vocal sample, and is an excellent way to cut down on valuable memory space. Depending on the amount of pitch variation along the length of your sample, it can also make your ensemble sound twice as large as the original sample. Ideal for sustained vowel sounds, it ignores the attack portion of a vocal sample and relies on VCA manipulation to fade it in. For added realism, automatic pitchbend (or "warp" as Akai calls it) on the way into a note can be employed.
1. Make sure the original sample is saved as a backup.
2. Find portion of sample with the most consistent pitch and coloration.
3. Discard sample material before and after this portion.
4. Save this new material.
5. Reverse this new material and save it.
6. Combine the forwards and backwards versions of the same sample at a 50/50 mix via software, then save.
7. Locate loop start and end points equidistant from the center of this mixed sample. With sufficient loop-point hunting, you shouldn't need to employ any form of loop crossfading,

► far from the original pitch a sample can be taken to meet its adjoining sample.

Samples generally travel downwards better than upwards. If I've sampled every minor third I'll take a sample down two semitones and up one semitone from its original pitch as a starting point.

As mentioned last month, this is where having ascending and descending semitone scales really comes into play. To arrive at the smoothest transition between vocal registers in your multisample, you'll probably have to do a fair bit of "mixing and matching" between your ascending and descending scales, maybe even a bit of back-tracking to your source recordings to fill up any holes in your multisample. (It's a good idea to jot down or make a mental note of your mixing levels, EQ and effects settings for this reason.)

Positional crossfading is a life-saver with awkward transitions between different samples, but really should be used as a last resort. If a sample sticks out of your multisample like a sore thumb, after a positional crossfade to the samples on either side of it the sample will still stick out, but with a smoother transition between adjacent samples. It would be wise to sit on that sample for a while and see if you can find a substitute from your source recordings or do without it altogether, extending the adjacent samples further up and down to fill the gap and use a crossfade between them if necessary.

Remember that each positional cross-fade uses up two "voices" on your sampler to do its business. Employing extensive positional crossfading to make your keyboard map will smooth things out, but also cuts your keyboard polyphony in half – eight-note polyphony instantly becomes four. Considering that keyboard maps take up relatively little memory space, experiment to your ears content.

VCA Envelope Shaping

VCAS ARE YOUR samples' window to the world. You don't hear nothin' if these aren't turned on.

Unless you want a percussive entry or have your vocals fade away while a key is held down, you really don't need to worry about the decay control of a standard ADSR envelope (attack, decay, sustain, release). A good starting point for oohs and aahs is to set the attack to about a half-second, sustain up full and the release to suit your taste. The VCA release will be very much like an ambience control on samples recorded with reverb.

Once a suitable envelope has been created, for added realism try making the attack and release times slightly longer (progressively so as you go down the keyboard, if you have the patience). You see, not only does it take longer physically and acoustically for lower pitches to be produced and to reach our ears, they also die away at a slightly slower rate to higher pitches.

Vocal Formants

IN THE JULY issue of MT, I touched upon the subject of vocal formants. To recap: vocal formants are accentuated audio frequency bands that tell our brains what vowel sound we're listening to, what size person is singing and what register is being sung in. No matter which pitch

is sung, these accentuated bands stay the same for a given vowel sound. No ifs, ands or buts.

There are two main formants for each vowel sound. For example, the formants for an average adult male:

Oo (as in spook) – 400 and 800Hz
Aah (father) – 825 and 1200Hz
Ee (feet) – 375 and 2400Hz
Womens formants are generally 17% higher, children 25% higher.

To hear vocal formants at work and to demonstrate that they stay fixed for a given vowel, try this little experiment: shape your mouth and vocal tract as if you were to utter a vowel (say, Aah) and whisper this vowel rather than producing a pitch. The effect should be kind of like white noise with the distinct characteristics of the vowel placed upon it. You'll find that there is very little leeway in the placement of your tongue and the shaping of your vocal tract before the unsung vowel starts resembling another vowel.

There comes a point when singing vowels in the extreme upper ranges, especially with females, that these vowels start losing their identity. This is due to the fundamental of the note being higher than the lower formants. Sampling the vowel whispered, with no sign of pitch present, and mixing it in with these can help retain its identity. Whispered vowels can also be mixed with all the members of a multi-sample to add a breathier quality to them.

Silly Stuff

CERTAIN SONGS HAVE succeeded in sending shivers up my spine using vocal sounds that would be impossible without sampling technology. While some of us go to great pains to ensure that samples sound as natural as possible, in many instances vocal samples played well out of their range sound fantastic; for example, when playing soaring lead lines in songs that ten years ago would have been delegated to the electric guitar. This is healthy not only for its importance in extending the palette of tone colors heard in today's music, it also shows the Doubting Thomases of the music world that sampling can be a lot more creative than merely putting session musicians in the unemployment line. "Rules" are meant only as guidelines, so that we don't waste a lot of time making the same mistakes as our predecessors did to achieve acceptable results. Once the reasons for their usage are understood you should choose to use them or not, as your creativity and better judgment dictate.

For the whackos among us, here are some silly things to experiment with for the strange coloration they impart to a voice:
- Sing down a cardboard or plastic tube; it renders speech more sibilant and creates a static flanging effect.
- Sing with your lungs full of helium; it raises the pitch of your voice, and to my knowledge, used in moderation is perfectly harmless (although you may find yourself re-enacting scenes from *The Wizard of Oz*).
- Sample only the reverbed or effected signal.
- Hum into a balloon, kazoo or piece of cellophane.
- Mix humming and whistling the same pitch at

equal levels. This creates a most unsettling, unearthly effect, perfect for film scores.

And while you're at it, don't rule out things like sending the voice through a distortion, octave-divider or other pedal effect, cardboard megaphone, electric bullhorn, telephone, or Leslie cabinet if you're looking for new vocal colorations to experiment with. Try a contact mic attached to drums, cymbals, guitar, piano (a brick holding down the sustain pedal), cardboard box, or a cookie tin, and sing in close proximity to these for added resonance. Once you get started with these ideas, I'm sure you won't stop and I'm equally sure that you'll come up with others.

Imitative Sampling

AS YOU'RE PROBABLY aware, many tracks have used vocal samples to imitate percussion and bass sounds. It shouldn't tax your imagination too heavily to realize that with sampling, a complete band or orchestra can be assembled with little more raw materials than vocal samples – either clean or heavily effected. With few exceptions, almost any vocal sound can be made to center around a frequency or frequency band using the tuning and/or looping provisions on your sampler, and totally re-shaped with a voltage or digitally-controlled amplifier and low-pass filter to resemble just about any existing or imaginary instrument or synthesizer effect. Such is the magic of sampling.

When using sampled voices to imitate other instruments, your raw sound material can be tailored at source by mimicking the instrument/effect you want to hear. Treating your vocal approximation with old VCF and VCA envelopes will go a long way in furthering the illusion. (Well worth brushing up on your analog synthesis theory if you're not thoroughly familiar with all the different qualities that can be imposed upon a signal, by altering the manner in which we hear its loudness and brightness.)

General Attack Characteristics of Instrument Families

AMPLITUDE:

Attack	Instrument Group
Immediate	*Percussion, plucked/struck strings.*
Moderate	*Brass, wind, bowed string, vocals.*
Slow	*Wind, bowed strings, vocals.*

FREQUENCY:

In tune	*Keyboard, wind, percussion.*
Flat	*Vocals, bowed strings, brass.*
Sharp	*Percussion, plucked/struck string, vocals, brass.*

Of the universe of sounds available to us, the human voice strikes a special resonance in all of us impossible with any other instrument or combination of instruments. With creative sampling still in its infancy, we've a wealth of sonic colors open to us, requiring little more than a microphone, a sampler and your own voice. ∎

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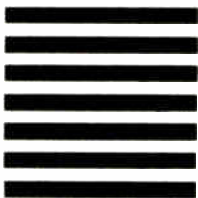
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E-mu Systems
Emulator III version 1.21
Digital Sound Production Workstation

The long awaited follow-up to the company's popular EII ups the ante to 16-bit stereo sampling. Is it a Fairlight without a CRT? *Review by Chris Meyer.*

RIGHT UP FRONT – I'm into testing the EIII, and I have a few problems and complaints. I call two people at E-mu, and both ask, "What version of software do you have? Aw, you should have the *new* stuff." One Federal Express package later, and I find almost all of my problems have already been yelled about by someone else (and therefore fixed), plus a dozen features have been added. Point – E-mu is going to keep this machine growing; this review is going to have to be a snapshot of it circa late June, 1988. However, that shouldn't change any purchase decision you may have pending on this review.

Why? Because reviewing the EIII isn't like reviewing yet another under $2000 synth or yet another 12-bit sampler. For one, it currently sits in a price category all

by itself (twice or more that of the 16-bit machines announced by Sequential, Akai, and Dynacord – let alone the 12 bitters; half or less that of the AudioFrame, Fairlight, and Synclavier. This may change once Simmons releases the keyboard and sequencer software for the SDX, however). Second, it's so comprehensive (we'll run down the specs in a bit) that spec wars are meaningless. It'll really come down to whether or not you need all (or a significant bit) of what's here, and how painlessly it fits your way of working. So, with that, here we go . . .

The Hardware

THE EIII IS a 16-voice "stereo" 16-bit linear sampling instrument with sample rates of 33.3 and 44.1kHz, 4Megabytes of RAM standard (expandable to twice that), and

an internal 40Megabyte hard disk. Of course, there's a 3.5" floppy drive, but you'll be ignoring it most of the time (there's that internal hard disk, and a standard memory-full takes six floppies to back up).

Each voice has it's own (analog – yay!) filter and amplifier, along with panning. There are 16 individual outputs along with stereo mix outs (the individuals do indeed take their related voice(s) out of the stereo mix, and bravo for that – the Emax and others don't). Other holes in the back panel include MIDI In, Out, and Thru, SMPTE/clocking in and out. stereo sample inputs, an RS232 and SCSI disk ports (the latter allowing external hard disks to be added – E-mu gives a list of nine that already work, and are making their own 300Meg rack-mountable one), a metronome output, and two footswitch and one footpedal inputs (supplied with the unit). Also making an appearance are a pair of trigger inputs that accept a wide range of signals (from pads to audio), with a nice set

of software trims to prevent double triggering and the like. But why only two? That makes this thing pretty expensive as a drum slave; you'd want more for either an electronic drum kit or replacing all your drum sounds on tape. Oh, well . . .

The keyboard version (there's a rack unit available too) also has a five-octave plastic keyboard with velocity, aftertouch, and mildly clunky feel along with a pair of assignable wheels. There's a fan built into both to keep all of this cool; between it and the hard disk, the EIII isn't exactly silent just sitting there (not bad; just not silent). I'm told E-mu is switching hard disks brands to one that also happens to be quieter.

User interface consists of a 4×20 character backlit LCD, a ten-key pad, four cursor controls (that occasionally double

up for other functions, such as moving through zero crossings of samples), inc/dec buttons that double up for yes/no responses, a data entry slider (curiously placed away from the keypad, as opposed to next to it, where the master volume and sample input level faders are), a number of dedicated buttons for the sequencer transport control and things like common disk functions, and a button per editing module. Spartan, but it works. The buttons have a good, solid feel, but the Enter button on my machine was prone to double-clicking. LEDs prompt where appropriate.

Communing with the Machine

E-MU'S SAMPLERS TEND to be based around the concept of a number of

product yet, and placing the operating system on the hard disk has all but eliminated the delays in switching modules. Unfortunately, I still hate using it.

Why? Because the modules won't let me stay in any one place. You hit a module button, it asks you for a submodule. You scroll thru the choices and hit enter (or hit the number directly), and there you are. As soon as you're done with a subfunction, you get thrown out of the module – no chance to stick around and fiddle with it if you didn't like what you did, or want to go do it again to somebody else. The worst example is sampling – if your sample clipped, it will sit there and tell you so, but won't let you resample until you leave the sampling module just to go back in.

Bear with me for one more paragraph of gripes, and then I'll tell you what I like. The EIII is not consistent from module to

Sequencer "The reasons (for building in a sequencer) aren't much different than for building in sample editing features – it's nice to be able to do something simple quickly at the machine."

modules with a number of submodules (and occasionally, subsubmodules) accessed inside each module to get the job done. Graphically, the EIII presents the module approach better than any E-mu

module (or submodule to submodule) as to how everything works. In some, the keypad can be used to enter numbers directly; in others, they instantly select alternate submodules. The machine likes to ▶

20

switch between Enter and Yes for how you acknowledge its latest step (that's intended as a safety feature in some cases, but it just slows me down). You can't back up or gracefully abort a function if you found yourself in the wrong place, or the Enter button bounced again – you have to either say "No" at the very last step (harrowing), or go to another module and come back. There's a wonderful Undo function for every time you actually alter a sample, but it's in its own submodule – you can't just stay within a submodule and try again. Bleah, bleah, bleah.

What is cool is the EIII has a bunch of defaults (most of them even sensible), which means you don't have to perform every step if you're not being too strange or picky (ie. it will automap samples upon sampling, if you wish). Undoing is cool – no more hesitation or remembering to make a backup before pushing that feared Enter button in the Mutilate module (it saves a copy of the original on the hard disk). A lot of functions are trimmable (such as how sensitive the zero crossing detector is). Dealing with stereo takes no additional thought on the part of the user. Numbers tend to be displayed in both time and number of individual samples (envelopes in time; LFOs in terms of frequency). Also, the EIII tends to be really good about telling you what you're working on (ie. which sample), even if it forces you to go to another module sometimes to change it.

Whereas most sampler manufacturers base things such as analog treatments around individual samples, E-mu bases them around "zones." Once you have spread a collection of samples across the keyboard (in one of two layers), you now change envelope shapes, filter cutoffs, etc, by ranges of keys with no attention to what portions of what samples you're taking in. Once you get used to this, it's nice to be able to forget exactly where samples are placed (and the display tells you their names, anyway). Copying a zone of a keyboard into another preset (or even another bank) also automatically drags all the samples that make it up along with it – another of the many places the EIII manages to keep senseless bookkeeping away from the user.

A Stroll Through the Modules

THE EIII IS a deep machine. There's no way to detail using every function of the machine here – even the manual doesn't do a very good job of it. Here's some of the more interesting points I wandered across in the week I went mano a mano with it.

After playing the piano preset (which is very nice, by the way – lots of hammer and felt), I went straight to sampling. The user gets to set up a template of parameters for new samples: mono or stereo, sample rate, number of keys to automap it over,

"True" Stereo Sampling

THERE HAVE BEEN two "stereo" samplers – Yamaha's TX16W and the EIII – reviewed in these pages so far. We have also run an article on "stereo" sampling with normally monophonic samplers by exploiting their stack mode (MT May '87). Alchemy can convert stereo samples from a Dyaxis into the equivalent on a mono instrument. However, none of these machines (with the exception of the Dyaxis) are truly stereo in the audiophile sense.

Ears are very sensitive to delay and phase shift – that's how they localize sounds. To keep a stereo image intact, you have to keep the time relationship between the left and

500 Hz Sine Waves

1 Msec delay
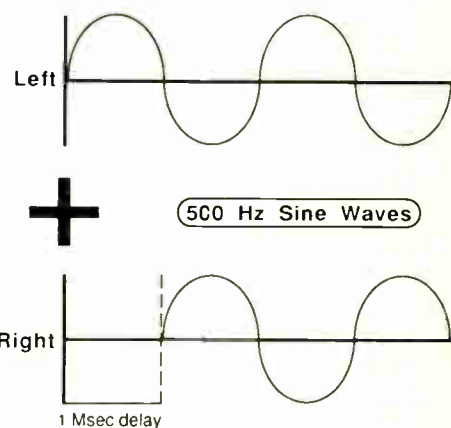
right sides very rigid. To keep this relationship, the two sides must be sampled in sync (or edited later to make their start points match perfectly), and must be started again at precisely the same time. Shift playback of one side in time against the other, and the stereo image will shift (or be destroyed altogether; add the two channels back together in mono, and some higher frequencies cancel). For example, if one side is delayed a scant one

whether or not to normalize the gain and automatically trim the ends. From there, you can sample away without a bother. The EIII is one of those samplers that actually grabs a little bit of sound before the sampling threshold is crossed; if I set the sensitivity on the zero crossing detector really high, it got the perfect balance of not leaving any silence on the front of samples while not chopping off any of the attack. Autotrimming the end was less successful; even at full sensitivity, some sound got lost (I was told the hardware in my machine may have been slightly out, thereby fooling the detector).

Looping was fairly cool. You have all

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millisecond, a sine wave at 500Hz (and every harmonic thereof) will be totally cancelled. This is because it has a wavelength of 2msec, and a delay of Imsec puts it 180° out of phase with itself, yielding perfect cancellation (see Figure I).

Most samplers have a delay from 500μsec to a few milliseconds in starting two voices "simultaneously," and this delay may even vary from note to note. Refer to the June '87 issue where Matt Isaacson and I tested a number of samplers, and look at the variance in the time between note ons and sounds out. To preserve a true stereo image, this delay would have to be one sample or less to make sure the highest frequencies didn't cancel (we're talking 25μsec to keep 20kHz clean).

To test your sampler for stereo cancellation, perform the following test: feed a mono signal with lots of high frequencies (like a snare drum) into the left and right inputs, or

sample it in mono and make a copy of it. Now play it back in "stereo," and mix the two outputs back to mono. Hear the flanged effect? That's the cancellation. If split back into stereo and listened to through headphones, the sound will appear to be off center. Play multiple keys and see if it changes character. True, this may even turn out to be a special effect you like, but it's yet another example of what the ads say not equalling "true" reality. ∎

sorts of tools to bring to bear - single stepping, zero crossing detectors, autocorrelation, and crossfade looping. Autocorrelation is a great idea in theory, but I haven't found an implementation in any machine that works as well as zero crossing detection. In the EIII's favor, autocorrelation managed to slightly improve upon its own zero crossing points. You adjust the loop start and loop length (as opposed to start and end), which personally annoyed me, but just as many people work the other way. In all digital editing functions, the pitch-bend wheel can be used like a video shuttle control to move back and forth through the sound to locate

a point (way cool). Loop points can be changed while the key is held down, which makes life easier. Crossfading may be linear or equal power, with adjustable length. The forwards equal power cross-fade is as smooth as any I've heard; the backwards/forwards left clicks worse than the original (back to the Undo function . . .). Stereo samples crossfaded just as well as mono samples; they merely took longer (it's harder to get a straight loop out of them, because either side could click).

Truncation is as you would expect. There's a nice collection of cut/copy/paste features, which saves sample fragments to an internal "clipboard" (à la the Macintosh). Gain change can be performed over selected portions of the sound, and can be faded in or out with four different fade curves - kudos for the number of choices; unfortunately, it didn't sound quite smooth to my ears. There's also a Taper function for just smoothing the starts and ends of sounds.

Once I got into the specialized digital processing routines, I almost never got out - kind of like a fly that happily drowns in a martini. The subfunction I thought most superfluous - digital sample rate conversion - ended up being the most useful. One, you really can buy back memory by lowering the effective sample rate after the fact to pass just the bandwidth you need (warning: loop *after* you're done). I also found myself artificially converting sounds to a *higher* sample rate, which made them sound less grungy when transposed downward. Using the sample rate calculator built in, the sample rate can be optimized to make perfect loop boundaries fall on actual sample boundaries. A variation on sample rate conversion allows you to change the pitch of sounds, to make it easier to mix them with others that were originally sampled at a different pitch (unfortunately, this function doesn't also keep the length of the sound the same, so there's no free ride in making one sample stretch all across the keyboard without chipmunk effects). All of these functions take time (typically minutes) to perform, but make me realize I was really missing them before they existed.

Off in funland, there's submodules to superimpose DDL and autopanning effects onto samples that both work very cleanly. I had a gas taking a mono sample, copying it to make it stereo, tapering the two sides to silence at different rates, autopanning the two sides against each other (for a twirling pattern), and then running them through the DDL. Complaints? The DDL had really restricted values for mix/feedback amounts (just numbers that were easy on the processor, instead of the "any value you want" philosophy of the rest of the machine), and I wondered where the Emax's DSP functions (convolution, additive synthesis) were. Yeah, I'm greedy. ▶

► Analog processing includes three envelopes: one VCA, one VCF, and one wildcard that can go to pitch, panning, LFO rate, or LFO depth to pitch, volume, filter, or panning (another good one, E-mu). All envelopes are of the ADHSR (stock decade-old synth envelope plus a useful-for-samplers hold stage) – I've gotten kinda used to multistage envelopes, but these are very adequate. The envelopes also have a one-shot mode (they skip decay and sustain) for percussive sounds and playing. The filter envelope can be inverted, and as you would expect, the filter has adjustable Q, cutoff, and keyboard tracking.

There's only one LFO per voice (boo, hiss), but it has four shapes, a wide frequency range (0.08 to over 18Hz), a delay, separate depths for pitch, volume, filter, and panning, and "variation": it can vary in pitch per keystroke (a little humanization). I couldn't find a way to sync the LFOs to keystrokes or each other. The LFO can be delayed, but it doesn't fade in – it just suddenly turns on. Velocity can go to ten different places, all with their own amounts (*nice* for tweaking effects in), including such esoterica as pitch, pan, filter Q, and sample start point.

Like on the EII and Emax, samples can be given a range of voices (and therefore individual outputs) to use. Since the EIII performs some of its processing in analog, it can't premix multiple voices to one output jack like all-digital units (Roland S550, Yamaha TX16W, etc) can.

As mentioned a bit earlier, two layers of samples can be defined. These can be stacked (with delay and detune), cross-faded by position, velocity, or real-time controller (ie. pressure), or switched between by velocity (with adjustable threshold) or external switch. I was slightly disappointed that samples on the same layer couldn't be positionally crossfaded – you have to checkerboard them and set up a number of small crossfades alternately between the primary and secondary layers. Up to eight presets can be stacked for a truly mondo sound. In general, the preset module (and the MIDI that drives it) seems to be really well thought out.

Remembering Notes

THE EIII HAS A nice rudimentary algorithmic composition device built-in that's particularly well suited for new age music (read: it has a comprehensive arpeggiator). That's not E-mu's sell line, that's mine. Many slag off putting an arpeggiator in anything these days; it's obvious what E-mu did is take the fun arpeggiator built into the Emax and merely slapped it into this beast too. No complaints.

Next to building in an arpeggiator, one of the most common questions these days

is why build a sequencer into an instrument anymore. Everybody has their favorite outboard hardware or software sequencer, and they tend to have more features anyway . . . The reasons aren't much different than for building in sample editing features, despite the widespread availability of external sample editing packages – it's nice to be able to do something simple quickly at the machine. I like the concept of a scratchpad sequencer that I can later download the contents of to a more "serious" compositional sequencer; others prefer having a sequencer built in for live performance rather than hauling a computer around. And some just prefer having everything in one box. Neither the arpeggiator nor the sequencer added much

cost (except for the SMPTE); it's up to a user's individual preference as to whether the software development time could have been better spent.

By the way, the EIII's sequencer ain't no slouch. It's at least as competent as any hardware sequencer you can find, with SMPTE, MIDI Time Code (the idea of linking this up with a Yamaha DMP7 and some automation software boggles the mind), decent resolution (96ppqn), volume levels per track, cut, copy, and paste, step-time entry and editing, some pretty advanced autocorrection, and the ability to view and edit events in terms of SMPTE timecode or bars, beats, and clocks. It does have some shortcomings – only 16 tracks, no time slipping or rotation

The EIII for Electronic Foley

One of E-mu's beta testers discusses the pros and cons of using the EIII for sound effects work. *Text by Bill Koepnick.*

IF YOU DO sounds effects for a living, approaching a musical instrument with black and white keys to do electronic Foley may seem totally alien. However, once you think of those keys as 61 switches and triggers to a multitude of custom sounds, it starts to make a lot of sense.

Having been a near-fanatical proponent of the EII for sound effects work for over two years, I was ready to move up to the 23 seconds of stereo 16-bit sampling at 44.1kHz offered by the EIII. And after much pleading with those in power at E-mu Systems and a check sufficient to purchase a nice mid-size automobile, I finally received my beta EIII –just in time for Christmas, 1987.

The EIII is a logical upgrade path for studios already using the EII by virtue of the similarity in terminology, layout, and overall feel of the two machines. Aside from sound quality, one of the biggest advantages the EIII has over the EII is the ability to load single samples (say, your favorite door slam) from anywhere on any disk up into the preset you're currently working on (perhaps a bedroom riot ensemble) – an improvement over having to haul out your library disks to build a performance one. However, if you have a library of EII sounds, be forewarned that you will have to use Sound Designer or Alchemy (when they become available for the EIII) to provide the file format conversions to transfer your library.

If you have never used a sampler in your post-production work, I feel the EIII is a logical place to start. In my opinion, its audio fidelity and ease of use are truly superior to devices of equal or lesser cost (as well as many systems

costing several times as much). There are a lot of automated sample editing features (covered in more detail in the body review) which take much of the drudgery and guesswork out of sampling and editing your sounds. The analog and digital processing sections of the EIII offer sound editors an impressive array of options for manipulating samples.

OK, as a Beta tester, you might think I wouldn't dare say anything bad about this machine less I jeopardize my insider status. But, yes, there are some things I must mention that are less than perfect . . .

For example, I had hoped that E-mu would have provided for an audio loop through the sampling chain (as in the EII) to give the user confidence that his or her sounds are being sampled correctly, but they had to omit that feature due to CPU restrictions during the sampling process. Too bad. Also, the sequencer is a marvel of engineering – allowing some functions that even the expensive computer-based music sequencers lack – but its implementation of timecode-based effects editing is cumbersome. It works well, but the user is required to operate in a fashion that seems awkward.

The most difficult problem is mass storage. High sampling rates at 16-bit density have the inconvenient drawback of requiring lots of memory. Once you sample 47 seconds of sound, how do you store it? The standard 40Meg internal hard disk is capable of containing only five full 8Meg banks of data, and most people are going to require more storage. The SCSI port allows attaching up to six more hard disks for increased storage, but I

functions, and to drive an external device, it forces you to create an empty preset and assign that the outside MIDI channel (one of the few cases where the EIII says "To heck with the user; we're gonna make things easy on me here"). Overall, if you have advanced computeritis and want to use a dedicated sequencer, you really won't be hurting here.

Uh – the Sound?

THIS IS THE first sampler I've used in which what came back out was almost exactly like what went in. I was scared. That's never happened to me before.

What can be said? Clean 16-bit is cool. There's tons of dynamic range, and the signal's quiet. Sounds hold up under

would prefer to have fewer hard disks and a streaming tape storage unit . . . or removable media hard disks . . . or – best yet – a WORM (Write Once/Read Many) drive.

I've been waiting for a WORM drive for its archival qualities, inexpensive disks (storage at about $1.00 per 8Meg bank), and huge – 800Meg – capacity. E-mu has been working with Optical Media International (who brought us the CD-ROM for the EII) to develop the WORM system for the EIII, and they hope to announce availability soon. I can hardly wait – I don't want to invest in more hard disks, and saving to floppies (and reloading from them) is incredibly tedious.

The biggest drawback to the EIII in a post-production environment is, I'm afraid, inherent in any high quality sampler: it takes a while to load – even from the hard disk – due to the huge amount of data being transferred. If you are used to the EII loading in three

transposition at least as well as any other sampler I've used – low end grunge stays away for quite a while when transposing downwards, and I couldn't detect any sidebands under any transposition. There was a low level buzz that would come up whenever I was in a digital editing module (it seems to come from the pitch/scrub wheel looping on one sample while waiting to move), but playback was pretty clean. And I like having real VCAs and VCFs around – I'm sure they distort more than their digital brethren (or having none at all), but it makes the instrument more musical to my ears.

Downside – the EIII just barely fails my "true" stereo reproduction test (see sidebar for more details).

seconds from its hard disk, you will need to adjust to the EIII taking about sixteen times as long when loading a full 8Meg bank (of course, you are loading more than five times as much sound (in mono), so you would need to load less often). Waiting times can be reduced, however, if you work with smaller banks of data. The EIII now allows you to save and reload a bank of any size as opposed to the EII's method of saving a full 512K even if most of it was unused.

So, what's the bottom line? The EIII is an evolving instrument; the company has already made some hardware modifications to overcome early engineering flaws. Presently, it is a fantastic tool for post-production sound work, and I feel confident (based on experiences so far) that new features will be implemented as E-mu continues to respond to suggestions from users. ∎

Bill Koepnick is the principol in Zound FX, bosed in Von Nuys, CA. A former drummer turned recording engineer turned effects designer/editor ond Foley ortist, he has severol feature films and well over 100 hours of television (mostly cortoons) to his credit.

Conclusions

A COMPARISON I keep making in my head is between the EIII and a Fairlight Series IIX. This comes up because used Series II's are starting to drop down under that of a new EIII (from a list price over twice that), and from remembering how many bands, studios, and megaproducers bought one to make themselves truly big league (that's no jab – it really happened that way). The Fairlight was loved because it was so powerful, and so integrated – multitimbral, sampling, sample editing, a couple of sequencers (including the much-loved Page R, known for how quick it was to use), pretty friendly user interface, SMPTE, and MIDI. Cool rhythm tracks through a whole tune could be written solo on one of them. I find some mental block in thinking of an EIII as such a star-maker, but in reality, it beats the old Series II in almost every category: it sounds far, *far* better (and is stereo to boot), it has more advanced sample editing, more voices than the base unit, and is also multitimbral. The sequencers are a wash, depending on working preferences. The EIII only loses out in terms of user interface – a four-line LCD and some cursor buttons is no match for a CRT and light pen.

Sure, I could niggle about a few small features I would like to see in the EIII, but the fact of the matter is that this machine is about as full-featured and sonically clean as you're going to get for less than a house. And, as mentioned at the very top, the EIII is isolated from its competition in terms of price and features (it doesn't outfeature any more expensive system; no cheaper system outfeatures it). Therefore, you have to make a business decision as to whether or not the EIII is a fit for how you are expecting to be making money for the next year or so.

Me? I personally found the EIII to be a lot more refined and thorough than I expected; I also found it to be a bit more cumbersome to use. And it sounds *great*. Methinks the EIII is perhaps best suited for the artist or studio who wants just one instrument (and therefore no interfacing hassles to deal with), who wants to and can devote the time to sampling, perfecting the details of those samples, who would like (at the least) a sketchpad available to compose them into a song, and who requires CD quality. For that (and nothing more, such as sophisticated synthesis), it's the best buy available. ∎

Roland D110

LA Synthesizer

Roland promises to populate the world with as many D's as Yamaha has DX's. Here's a quickie on their latest variation.
Review by Thomas J. Clement.

A LOT CAN happen in four years. Sitting in my rack is an "old" Roland MKS30 Planet S. The two rack-space Planet sports 64 internal and 64 cartridge RAM memories and friendly analog parameters. Its sounds can be huge, but the Planet is unfortunately limited to six voices and only one timbre at a time. The 1984 list price was $999.

Directly over the Planet is Roland's new D110, a single rack-space module with 192 sounds on-board and another 192 sounds available via Memory Cards. The D110 utilizes Roland's well-known LA synthesis; the sounds may not be as fat as some of the Planet's soup-thick timbres, but they aren't digital-thin, either. They have a warmth, a majesty, and a certain coolness all their own. Up to 32 notes can be played at the same time, and eight different timbres can be sequenced at once via MIDI. Not limited to keyboard lines, there's also 63 strong percussion sounds built in. In addition, there's on-board processing like reverb and delay, and I can add outboard effects to individual sounds via the D110's six separate outs. In other words, in one rack height is enough power to roll out music that four years ago would have taken several Planets, a drum machine, and digital effects to create. The 1988 list price? $999.

Sums and Differences

THE D110 HAS four Timbre groups: A, B, I, and R. Both A and B are preset with 64 Timbres each. The programmable I group has another 64 Timbres. The final group, R, stores the 63 percussion Timbres. "Timbre," by the way, is LA-ese for a "Tone" that's had performance parameters like pitch-bend tacked onto it (the D110

responds to mod and pitch wheels, but not to aftertouch).

The D110 is similar to other members of the LA family, especially the MT32, the D110's slightly older, less sophisticated, and $300 cheaper sibling. (One major improvement over the MT32 is that the D110 has a battery backed-up memory.) About 90% of the presets are from the MT32, but they're much improved on the D110 – cleaner, louder, and often tweaked for the better.

The MT32 and D110 aren't exactly the same inside (a few Timbres sounded better on the MT32 – 'Fantasia' and the trombones, for example), but they're fairly close. The major difference is that the D110 has new and better PCM samples (and more of them). I used Beaverton Digital's Editor/Librarian to transfer MT32 Timbres to the D110. Some made the crossing intact, while others changed, often with pleasing results. The D110 doesn't understand a full bank dump from an MT32 (some of the Timbres don't port over).

The D110's percussion section is another step up from the MT32. The sounds are meatier and, with four bass drums, six snares, four tom sets, plenty of cymbals, and lots of latin percussion, they're perfect for most sequencing needs. Any Rhythm Part can be programmed to include percussion and LA tones. If you have a computer, a sequencer, and a D110, owning a drum machine could quickly become redundant.

The D110 scores big with its multitimbral powers. It can store 64 setups called "Patches." Each Patch features up to eight Timbre "Parts." You can place these Parts anywhere in the 15 stereo pan positions,

assign key range and MIDI channel for each Part, or turn MIDI off for each Part. There's also a Partial Reserve feature which can give a Part "Partial priority" to help avoid note robbing in sequences. Finally, the Patches can be named (up to 10 characters). Overall, this is the easiest to use multitimbral system I've seen.

An LA synth wouldn't be true to the D50 line if it didn't have on-board stereo effects. The D110's processing is a big improvement over the MT32's, but isn't nearly as sophisticated as the D50's (eg. no chorusing or EQ). There's small and medium room, medium hall and large hall, plate, delay 1-3 and reverb Off. Effects can be sculpted by altering the level and time. Unfortunately (but not surprisingly), the reverb covers every Part in a Patch, not the individual Parts.

Roland must be commended for building multiple outs into the D110, but their use is limited. Assigning Parts to various outs isn't done in the Patch section; it's done when a Timbre is programmed. Therefore, whenever you use a Timbre in a Patch, it always has the same output assignment unless you reprogram the Timbre. Perhaps this is why the D110 won't let you use any of its internal effects for a Timbre sent to a direct output. And kiss outputs 5 and 6 goodbye if your Patch is using any internal effects; processing has to be shut off completely before all six outputs can be used.

Programming

PROGRAMMING THE D110 isn't impossible, but it isn't easy either (a discussion of LA theory is given in Music Technology's June and July '87 issues, when the D50 was originally reviewed). The D110 lacks a few niceties found on the D50: one LFO instead of three, and fewer effects.

The D110 uses a system of editing Pages which are first launched by pressing the Edit button (actually, there isn't a whole

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lot that can be done without pressing the Edit button). Remembering which Pages to step through and how to step through them takes some getting used to – which brings us to the D110's user interface.

The D110 has one of the most baffling interfaces I've seen (and there's some stiff competition). First, there's the front panel. The D110 features two rows of eight black buttons. They're so close together that it's easy to quickly press the wrong button a few times, and get thrust deep into some unwanted editing or full dump function (there's room on the panel to spread things out, and obviously I wish Roland had). It took a while to figure out which series of buttons to push to sample all of the Tones. Eventually, I came up with the magic combination: Timbre, Edit, Group, Bank, and Value (of course, it seems so simple now).

If I was going to spend more time playing than guessing, I'd have to defy the user's credo of "Damn the documentation – full speed ahead!" and crack the manual.

But, aha! – the plot thickens. The reading ain't easy. In the first three pages I meet the following terms: Partial, Common, Structure, Part, Tone, Patch, and Timbre (and a cast of others). Roland's lexicon is unnecessarily inflated and the definitions (when given) are foggy. The illustrations aren't much help, either. Under the description of the 8 modes, I'm shown a diagram – of 13 modes (there really are only 8 – you go figure). And there's about 100 pages of this stuff, too. In its favor, the manual finally begins to make sense at page 36 with the discussion of editing.

Naturally, Roland is peddling yet another in a long line of hardware programmers – the PG10. Perhaps they should put warnings on their synths: "Sounds can be edited from the panel, but we've made it so annoying that you'll hopefully want to foot $399 for one of our external programmers." Now, in fairness, knobs and switches for easy programming do cost money, and the philosophy in leaving them off the base unit is to not charge owners who don't bother to program (all too many). However, those of us who do feel slightly stung.

Problems With Being Early

BUT THE D110 has more problems than just a poor manual. My D110's display window occasionally shows gibberish, a

combination of English, strange designs, and what appears to be Japanese. This, coupled with the fact that some Patches that I've saved don't stay saved, but instead mutate when I move to another Patch (a glitch I've been able to repeat!), probably means that Roland didn't work all the bugs out before the D110 went to market.

When I called Roland about some of these problems, I encountered a temporary dead-end. It's not that the tech people don't want to help, it's just that the person I talked to hadn't received a D110 by the time I had been able to buy mine off the shelf. Not very reassuring.

Later, Roland informed me that many of my problems had to do with slaving the D110 from an Ensoniq EPS. Apparently, the D110 has a very small MIDI buffer, which is easily overflown by the gobs of polyphonic aftertouch information the EPS sends out. Roland is going to reset my machine and feels that will cure most (if not all) of my problems; meanwhile, I'm going to set up

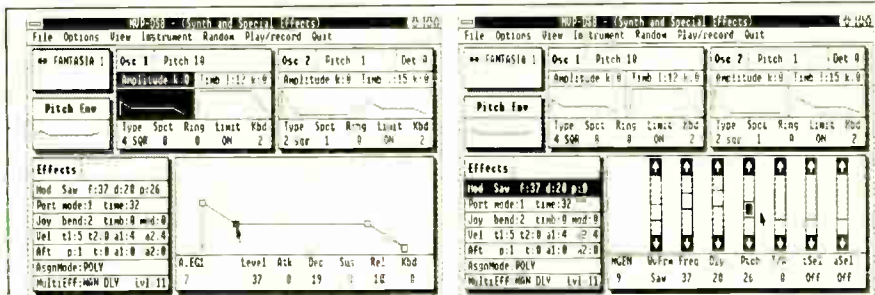the EPS to just transmit normal channel aftertouch.

Der Verdict

THIS BEAST SOUNDS fabulous (if you don't believe me, listen to the eight built-in ROM demos). It's a flexible all-arounder and a powerful tool.

Time is music, though, and learning the D110 wastes a lot of time. But whenever I got frustrated with the interface, I just stopped, played those beautiful "Timbres," and tried again. The D110 is aspirin for the ears. It's too bad it can also be an occasional headache. ∎

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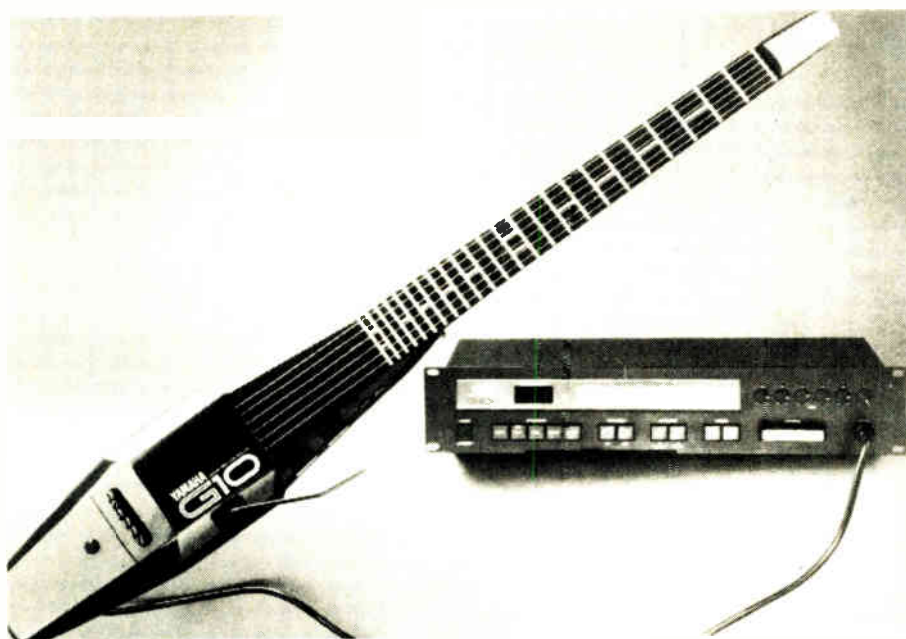
The secret is sonar in the company's long-awaited MIDI guitar system. *Preview by Bob O'Donnell.*

THE HISTORY OF products which attempt to allow guitarists to play synthesizers has not been a pretty one. From the mid-'70s days of the infamous and ill-fated Arp Avatar up to the present day of the SynthAxe and the Stepp guitars, quite a few manufacturers have bet it all and lost on the hopes that their product would be the one that would connect the millions of guitar players with the joys of synthesis.

A few months ago, however, a great deal of hope and anticipation was created by the announcements from several companies (notably Passac, Beetle and Zeta) who claimed to have finally overcome the technical hurdles that prevented most guitar synth systems from really working in a way that makes sense to guitarists. Finally, it seemed, guitarists were going to have several workable alternatives for making the leap to MIDI. Strangely enough, now some of these same companies are starting to hedge their bets, saying that maybe the MIDI guitar market is never really going to reach the enormous proportions that many people originally envisioned.

Meanwhile, industry giant Yamaha continued to quietly work on their version of a MIDI guitar controller – an instrument that some felt would be *the* answer for guitarists looking to take part in the MIDI revolution.

With the imminent release of the funky-shaped G10 Guitar MIDI Controller and G10C Guitar MIDI Converter, it's clear that the price alone will prevent the new Yamaha system from becoming a standard addition to most guitarists' equipment collections. On the other hand, it does look to be a significant addition to the professional market.

The approach Yamaha has taken with the G10 and G10C is that of a dedicated MIDI controller. It uses a single set of .016 gauge (G-string size) undamped guitar strings and has no conventional pickups or output jacks (and therefore produces no sound of its own).

The technology used by the G10 system to convert guitar performance into MIDI data (which bears a striking resemblance to the system used in the Beetle Quantar, previewed in these pages last month) is the main thing that helps separate it from the rest of the field. To quote the Yamaha literature, "An ultrasonic sound is transmitted along the strings and the fingered fret is determined by analyzing the reflected wave – sort of a musical 'sonar' system." The whole process supposedly occurs in well under a millisecond so that no delays of the type found in systems which use pitch-to-MIDI conversion occur (see the review of the Casio PG380 elsewhere in this issue for more on the topic). The velocity of the note is determined by a separate electromagnetic pickup, referred to by Yamaha as a string velocity sensor, and a third pickup, using "optical shutter" technology, is used to detect string bends.

The three pickups work in tandem to produce a system which is supposed to be able to accurately track things like muting, bends, hammer-ons and pulloffs. I haven't actually worked with the G10 – this is a preview, after all – but reports from those who tried it at the Frankfurt trade show (where it was introduced) suggest that it does an excellent job of tracking.

The G10 guitar connects to the two space

rackmount "brains" of the system, the G10C, via a special 7-meter long cord. The signals from the guitar are then processed and sent out the G10C's MIDI Out port (it also has a Thru port and an In for Sys Ex dumps). Though it may provide more flexibility, this two-piece system seems more cumbersome than the all-in-one approach taken by the Casio MIDI guitars and the Beetle system. They both allow you to simply connect a MIDI cable straight from the guitar into whatever synth you're using. Other connections on the G10C include jacks for a single footswitch and a single foot pedal, which can be assigned to control a variety of parameters per G10 program, and two jacks for incrementing and decrementing through the 64 available G10 programs (an additional 64 can be stored on an external RAM4 cartridge).

You can also select synth programs from the G10 itself, which has two program select buttons located along the bottom edge of the guitar and an LED along the top edge to show you what program you've selected. Next to the program inc/dec buttons are two programmable control wheels – almost like miniature pitch and mod wheels – which can be assigned to things like volume, modulation, portamento time, panning, etc. You can also assign the whammy bar to performance controls like pitch-bend. In addition, the G10 features an overall sensitivity control (the G10C features individual gain knobs per string) and a breath controller input. All in all, a pretty extensive degree of performance control.

Programs on the G10C include volume, transposition, velocity curve (four preset and four programmable) and MIDI channel per string, mute level, program changes, controller assignments and other parameters.

Though the G10C does not produce any sounds of its own, within its memory Yamaha has stored a special group of patches and performance parameters for the TX81Z and TX802 synths – designed and programmed to be used specifically with the G10 system. You can dump these into the respective synths via the G10C's MIDI Out. It's a trickly little incentive to buy a Yamaha synth as your sound source. Smart, too, in that anything other than a keyboard controller needs something other than keyboard-oriented patches to best show it off – and Yamaha has always tried to provide plug-in-and-go services to their customers.

The odd shape of the guitar and the system's price may make the G10 a tough sell for Yamaha, but if it works well as an expressive and effective controller, these considerations may not matter. Only time will tell. ■

READERS' TAPES

The question to avoid from the Master of the Mystical: What's New? *Reviews by Yung Dragen.*

THE TOPIC OF this month's pseudo-philosophical ramblings is the word "neo." I asked a few months ago if anything could be neo. No one answered (and when I threw the I Ching, the sticks wouldn't even shake out of the can). I guess I'm just trying to find some grounding to tie the term "neoclassical" to without resorting to reincarnation; what Bach did really has no more to do with this month's collection of neoclassical tapes (there *are* some other styles here, but there's nothing resembling a yin/yang sort of balance) than does Scritti Politti. Yet it's a term that gets applied to "serious" music that's modern in nature – kinda like the same pigeonholing jazz gets, but without the beat.

The best example is **Glenn Meade's** three-movement demo of his "new music for synthesizer orchestra." Glenn "has been writing contemporary orchestral, chamber, solo, and musical-drama works since 1975," and his three-page list of compositions does indeed look like that of a semi-important new local (to the Chicago area) serious composer. Instead of writing at the piano and training others to perform his works, he employs his MIDI "orchestra" consisting of

Yamaha TX816 and FB01 synths, an Oberheim Matrix 6, a Mirage, and the ubiquitous CZ101. All are driven with a Roland master keyboard and recorded into a Mac.

The three pieces ('Momenta,' 'In Motion – In Repose' and 'Rondo') all feature fairly bland but uninsulting, mostly percussive timbres that electronic neoclassical composers seem to prefer (the Dan Schaaf Ensemble comes to mind, who, by the way, have recently submitted a much improved second tape). These pieces are tonal – while still having challenging sections – and show confidence and maturity. When I listen closely, I'm impressed. They are, however, a bit lacking in human emotion to keep my interest from wandering. Nonetheless, Glenn, you're good now, and more years of study will further you down your path.

Stepping left into a slightly different continuum finds **Bruce Maccabee** and *The Joy of Ivories, Part II*. The music here consists of personal compositions split between neoclassical and piano boogie woogie. Bruce uses an ESQ1, Mirage, and Roland CR1000 drum machine to compose on; and he details in his letter how various pieces came about from collecting jams, or laying down scratch

neon tetra

parts on the sequencer to layer over (and replacing the scratch lines later). 'Mountain Storm' is a dark one-pass composition that starts off with a wonderfully ominous low string part that then wanders aimlessly around some very good themes and ideas. This is followed by 'Smooth Sailing,' which is a laidback, wistful instrumental that also features some nice chords, but doesn't grab.

The rest can be quite unfairly dismissed as boogie riffing at the piano that sounds fun, but also, unfortunately is not riveting. Bruce – skills and talents are wasted if one doesn't latch onto a light (inner or outer) to follow (or at least stare at constantly while moving in that general direction).

I keep invoking the name of Jaxon Crow as mantra for those who want to do interesting, soothing electronics without being seduced by the current opiate of the masses, New Age music. Jaxon has formed a cassette label (Neon Tetra) with some friends, and sent a handful of their first releases along with a letter.

Crow's new cassette, *Amazonia*, is a pleasant if ever so slightly weak follow-up to his own *Nextworld*. *Amazonia* is less dense, and shows a greater variety of styles (to its benefit and detriment). The first and last pieces are the best (the first including a real nice breathy flute inflected in a totally non-flute manner; the latter is a meditative water piece composed with Music Mouse); in between are pleasant *Nextworld* spinoffs along with an annoying, harshly clanging "dance of the spirits" style piece and one that sounds like "short theme for MIDI stack." However, calling this bad would be like calling any one of Bach's Brandenburgs an obvious slump.

Jaxon pairs with David Price for *Alphawaves* and deals up a denser, cloudier


Glenn Meade

version of *Nextworld* that I really like (hazes of sequences, sounds and unusual drum machines). Even hazier is David Price and Tim Boone as *Square Wave - Scubatronics*. We're thoroughly underwater at this point; one side was amazingly performed live (with Will Clay on sax waaaay in the background). These two are great for having your neighborhood meditational class over.

Jaxon's psychedelic rock band is *BYOTV - Flirting With Disaster*. Teamed with James E. Flory III and J.D. Haight, we get a twisted, quirky acid rock version of his meditative efforts complete with fascinating lyrics ("Come and see what I have done/I think I've killed him with my gun/With a little more light I could see/It's just a criminal mystery"). One of my all-time favorite tapes. I pretty much want everything on his label.

Another sideways shift to another neo-classical alternate universe, and I find **Ray Brunelle** and his first tape, *The Fall of the House of Ipanema*. Ray is another percussionist who has used MIDI and a slew of equipment to create semi-serious music. This again falls into the class of Simmons MTM music that Paul M. Van Patten and Bob Gatzen have previously submitted (fragmented times, percussive chords and notes), but Ray applies more flow and melody. The lead cut, 'Equal Amounts of the Same Thing,' is perhaps one of the most successful pieces of this genre. The third cut, 'The Powered Himalayas,' starts to sound like travelog music; the fifth and title cut (which closes side one) sounds like a cross between the timbres of Patrick Gleeson's staid electronic version of *The Planets* and an alternate universe version of Alan Parson's 'The Fall of the House of Usher' (heavy themes, rain, and the like, but too few ideas for seven minutes).

'The Talking People' is a humorous run-and-hide piece with treated gibberish voices, but the jokes get old quick. This is followed by a twelve-minute opus ('Eleven Bucks')
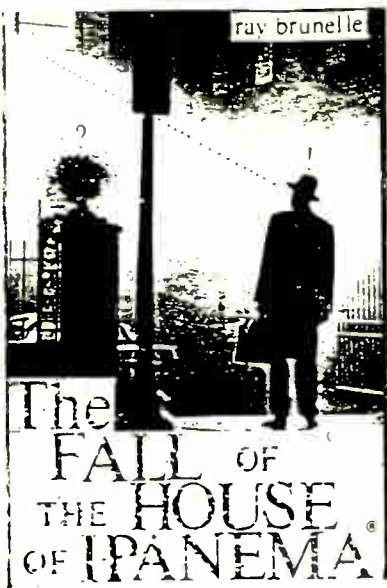
that is too light again for my tastes - which then leaves me to wonder if *I've* had too many TM sessions and can't take humor in music (I can, but it's gotta be Zappa, Creme and Godley, or Devo) - kinda like the Creme

and Godley lyric, "Does getting into Zappa/ mean getting out of zen?" The last cut, 'Idle Ju Ju,' is the first time I've been relieved to hear a drum machine come in, because Ray

Byron Fordham

creates some really nice patterns, sparsely weaved around a good collection of chords and timbres. Ray is closer around the ring to musical enlightenment than most of the percussion-heavy artists I've heard so far.

Alas, I end this month's séance with another songwriter that is falling short of the mark. **Byron Fordham** is writing and producing adult-oriented tunes at home with a Roland HS60 and Dr. Rhythm drum machine. He submits a harmless, lightweight, fast-paced instrumental called 'My Cup of T' that gets bogged down by a lack of variety and attempts to comp a string pad faster than the patch can articulate it. The second cut, a spacey love song called 'Startrip,' is based on a dream Byron had of a woman that came to him, and they flew out among the stars. Again, it comes off a bit too light and syrupy, and attention is not paid to selected sounds - the echo on the voice garbles the line, instead of enhancing it. It seems that too often songwriters aren't looking at a big enough universe when they compose - they please themselves with a nice idea, but don't hold their match up to the candle of the rest of the world.

New worlds (and for that matter, musical genres) are being created faster than names for them. And you know, I think I prefer this bit of chaos in my world more than perfect Western order. Until next month . . . ∎

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Send your demo-tape, a biography, equipment details, and full color charcoal portrait (if you have one) to: **Readers' Tapes, Music Technology, 22024 Lassen St., Suite 118, Chatsworth, CA 91311.**

World Radio History

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Rock, pop, rap, heavy metal – is this part of your concept of Contemporary Christian music? To initiate a series of interviews with artists working in particular genres, we take a glimpse into the technology and philosophy behind two of today's top Christian artists – Amy Grant and Michael W. Smith.

Interview by Wheat Williams III.

DOING AN ARTICLE on Contemporary Christian music seemed pretty radical to me at first, especially for a magazine like *Music Technology*. For some reason, I tend to pigeonhole the breadth of a magazine into a well-defined area – like *TV Guide* should only interview people in television, *Creem* should only interview heavy metal acts, and *MT* should only interview people who are first and foremost famous for using electronics. You know what I mean – they should play it safe. No reason to go out on a limb.

"But realize," says Don Roberts, a technological consultant to the music industry in Nashville, "that the Contemporary Christian music we record in this town, and the tours we put on, are up to the same high standards of technical excellence you'll find in LA, New York . . . anywhere. And the artistic creativity is on a par with anybody's."

OK, so maybe pigeonholing the kind of interview *Music Technology* should run is just as absurd as, say, trying to keep an artist locked into a particular classification of music. Perhaps these labels only serve to alienate listeners who would otherwise be able to enjoy all kinds of music. Maybe this labeling process hurts everyone.

I started thinking about some of my favorite pop rock bands who *haven't* been labeled as Contemporary Christian acts, yet express a good deal of Christian values in
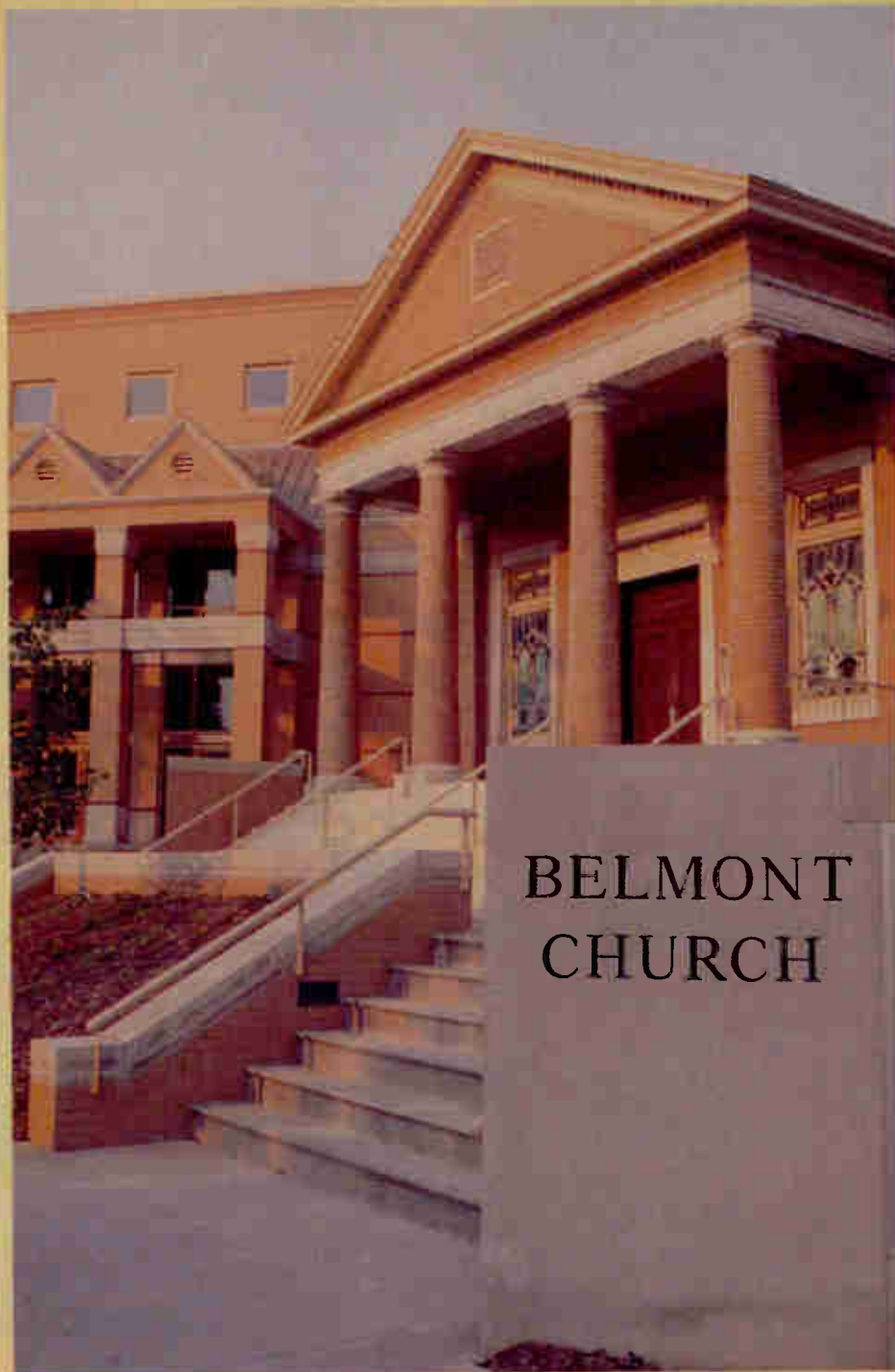
their music – U2, Mr. Mister, The Call, Bruce Cockburn, Earth, Wind & Fire, Deniece Williams, Aretha Franklin – and soon realized there are hundreds of them. So why should there be such a huge distinction between those folks and the artists who want to call themselves Contemporary Christian musicians? Why the prejudice against these acts? And why would anyone choose to use a label that might prevent them from becoming a commercial success?

A good number of musicians do decide to identify themselves with this genre, however. Superb instrumental artists include rock guitarist Phil Keaggy, producer/keyboardist Michael Omartian,

and classical guitarist Christopher Parkening. There are progressive rock artists like Petra and Michael W. Smith, and heavy metal bands like Stryper and Bloodgood. Amy Grant is an adult contemporary artist. Artists like Gary Chapman (Grant's husband) and Billy Sprague record country-flavored Christian music. There are even Christian rap acts, like Michael Peace. Now, please; somebody tell me why they've gone for the Christian label.

"There is a camaraderie among most of the people who do Contemporary Christian music because they can hardly pay the bills," says Amy Grant, the four-time Grammy award winner. "They want to ▶

BELMONT
CHURCH

▶ communicate their feelings, so they stick it out with this genre of music even though it has a negative reputation with many people. There are some great musicians that will never make a widespread mark on the musical community of their day because they have chosen to communicate Christ in their lyrics."

"I really think that there is a lot of high quality music to be found here," adds Brown Bannister, one of the genre's most respected producers. "In fact there is a sense in which people in Christian music

Amy Grant.

shoot for a higher degree of excellence. There's not as much money to be made or to invest in this music as you find in the mainstream industry. The market isn't as large. It's really amazing how far it has come with as many impairments as it has. It just comes down to the talent and integrity of the artists."

Of course, not everyone would agree with that assessment, as Grant relates. "There are some musicians that would never make it in any other area except for contemporary Christian music."

"There's good quality and bad quality in any musical genre, isn't there?" asks Bannister in response. "I'd recommend that people try to get to know the artist and not worry about the label so much. Sometimes you have to dig below the surface before you really find out what it's all about."

Michael W. Smith, Christian rock's best-known keyboardist and 1985 Grammy winner, echoes the point. "I think what we're doing is just as good or better than a lot of stuff that's out there. It seems like the only thing that makes Christian music Christian music is the lyrics. If anybody heard my records and didn't know anything about them, they'd think, this guy's making pop music. Labels confuse the issue. And I think I'm tired of trying to figure it out. I just need to do my thing and make the best

records possible."

Perhaps the most insightful information came from Don Butler, august Executive Director of the Gospel Music Association, which awards the industry's equivalent of the Grammys, the Dove Awards. "Contemporary Christian music is part of a larger form called 'Gospel, ' " he explains, "which merely means good news. What happened in the case of Contemporary Christian music is that in the early Sixties it was derisively called 'Jesus music,' associated with 'Jesus freaks.' It first hit the

scene in LA, and people like Gary McGuire and Larry Norman got it started. They were Christians, and yet they were bred into the rock idiom. They wanted to do their style of music and the church wouldn't let them. And that's when they went to the streets and coffeehouses. They wanted to play the rock style, but they wanted to let people know that they were Christians."

Christian music is, admittedly, sometimes hard for those not involved with it to understand. Grant observes, "I think some lyrics are written on the premise that we all have this certain common experience that's going to unlock the meaning of the lyric. And I think that can be seen from the outside as being exclusive."

Consider this: when you talk about your music to another electronic musician, you don't hesitate to load your conversation with comments about frequency response, envelope generators, overtones and MIDI status bytes. Those terms are the vernacular of your language. But if you have ever tried to explain your music to a non-technical friend, you know that you could not expect him to understand those terms, so you would describe your music in a language he speaks. Similarly, some Christian music addresses lyrics in a vernacular that is understandable and familiar only to people who spend a great deal of their lives going

to church and learning about the Bible. Not all Christian musicians take this approach, however.

"As Christians, the last thing that we should be is exclusive," asserts Grant. In her own writing, she seems to be particularly sensitive to and avoids the problems that occur from explaining the world with simple black and white dogmatic answers. Instead, she thrives in a world of greys and is able to communicate those feelings in and through her music.

"On this new album (Grant's 11th, *Lead Me On*, on A&M Records) you don't feel like 'Hey, I'm not a member of that club!' when you listen to it," explains Bannister. "It's more spiritual in a general way."

"When I started it, A&M wanted me to express who I really was, not just to record hit pop tunes," says Grant. "During tracking sessions I gave birth to my first child, Matthew. As far as writing this album, I kept thinking, 'What if something happened to me before Matt and I had a chance to talk about life? How would he know who I really was?' He'd be thinking, 'Gee, my mom was a Christian singer. What does that mean? I wonder what she really thought about God.' I found myself needing to say things on this album that I've never felt the need to say before. I felt a real need to just be painfully honest. Radically real. It's like saying, 'Do you want to see the reality of Jesus in my life? Here I am. And this is what it's like. And if you can come to understand God by looking at me, go ahead.' "

Two tracks off *Lead Me On* stand out as being particularly revealing: the Janis Ian-penned 'What About the Love' and Grant's own 'Faithless Heart,' which she admits to nearly pulling from the album on two different occasions. The latter deals with the occasional thoughts of infidelity which pass through a married woman's mind. Grant was worried about the perceptions it would create in others. "I was thinking, 'I don't want people to be needling my relationship with Gary (Chapman, her husband). I don't want anybody to try to put a name and a face on who I'm singing about.' There are a million names and faces, OK? But then I thought, 'Look, if you were not singing and you were sitting in your room at night listening to records, would you want someone to sing this song to you? Would you want somebody to articulate what was already there in your heart?' And I thought, 'Yes!' so I put it on."

GRANT'S LAST ALBUM, *Unguarded*, was a technological tour-de-force of dance rhythms, drum machines, keyboards, sequencers and Brown Bannister's skillful production. Bannister believes that her new album reflects a newfound maturity, musically and lyrically. The title track is a soaring arena-rocker, dealing with

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the pain and suffering of the American slaves of the Old South, and with the persecution and extermination of the Jews at Auschwitz under the Nazis. But it is a triumphal song.

The rest of this 62-minute album has an easier musical feel, a bit reminiscent of Bruce Hornsby's sound. "This time out, there are no sequencers, no drum machines, although a little sequencing will be needed on tour," explains Bannister.

The technological edge is still there, but it wears a different mantle. "*Lead Me On* was tracked live, with all the musicians in the same pair of rooms in the Bennett House in Franklin, Tennessee," continues Bannister. "It's an old Victorian mansion, with a studio behind the house, but I wanted to use a natural setting. So we took an 8" PVC pipe, knocked a window out of the house, drilled a hole in the wall of the control room out back and ran over a mile of Monster Cable through the pipe. The engineer in the control room communicated with everybody by a series of closed circuit TV monitors.

"Most of the musicians were set up in the living room. We put the drums in a bay window of the parlor, which has a loose floor, because we wanted a lot of vibration – totally opposite to what the recording technology dictates," laughs Bannister.

"We put the guitars in with the drummer in that room, and then we put the keyboards in the second room: Shane Keister, who relies on a Synclavier that I co-own with him; Robbie Buchannan; and Michael W. Smith. Alan Pasqua's PPG was used on virtually all the overdubs."

"My Synclavier has 100kHz stereo sampling and the new mouse-driven N series software," explains Shane Keister. "We actually recorded the entire piano part on 'If These Walls Could Speak,' played on a Bösendorfer grand, direct to disk on the Synclavier at another studio. For the other tracks, I used my Fairlight IIx, an Oberheim Matrix 12, an OB8, and a MIDI'd Oberheim Four-Voice analog (serial number 0024), along with a Yamaha DX5, an Ensoniq ESQ1, and a Memorymoog. We also had a Roland MKS20 and a Super Jupiter. I controlled everything from a Yamaha KX88.

"Robbie Buchannan used a KX88 as well, and his gear included an Akai sampler, a Sequential Vector synth, a PPG, a TX816, an OB8, a Super Jupiter, a Minimoog, and the first Roland D550 rackmount ever made, which arrived while we were tracking," says Keister.

"Benmont Tench, of Tom Petty and the Heartbreakers, was flown in to play a Hammond B-3 organ that's on several tracks," reveals Smith.

Bannister continues: "We put the bass player and the two guitarists in the same rooms with the keys and the drummer, but we ran cables and put the guitar amps in

separate bedrooms and the bass amp in a stone cellar with a unique frequency response. I wasn't quite expecting the results, but we ended up not using any processing like echo or reverb on the guitars. The drums were all recorded with natural room ambience – no AMS, no Lexicon, no nothing. We didn't even gate the microphones.

"The whole album was recorded on the Mitsubishi X850 digital 32-track. But the use of purely acoustic ambience, plus the fact that I mixed the album with Shelly Yakus in LA, who has a healthy distrust of digital recording, contributed to a very analog warmth."

MICHAEL W. SMITH will be touring with Amy Grant in Europe in late August when his fifth album, *i2 (EYE)*, is scheduled for release. His last album, 1986's *The Big Picture*, was a technological assault of progressive rock, recorded at The Power Station in New York and produced by John Potoker (of Mick Jagger fame). Complete with Synclaviers and Fairlights, and making the most innovative use of drum machines and percussion samples I've ever heard, *Picture* was augmented by session players like bassist Tony Levin (Peter Gabriel, King Crimson) and heavy metal guitarists Eddie Martinez and Dann Huff. Smith's gritty, distinctive tenor voice at times took on a surprisingly lyrical quality.

The songs on *Picture* are a relentlessly

L-R: Bill Whittington (engineer), Wayne Kirkpatrick, and Michael Smith.

paced montage of sequenced counterpoint, samples of found sounds and radio dialog, wailing guitars, and Shane Keister's trademark vocoder work.

"Yes, that's me," says Keister, "on an old, rare 16-channel Moog vocoder from 1978. Only 500 were made. I usually use it with an Electro-voice RE microphone to modulate an Oberheim synth with a two-pole filter. We doubled vocals and string lines. On one song we were able to do a

bridge where Michael's unprocessed voice faded naturally into the same lyric in a completely synthetic vocoder sound."

The lyrics of the songs Smith and Wayne Kirkpatrick wrote for *Picture* deal with the serious problems of maintaining integrity and self-esteem in modern society. Most of the songs' themes were actually suggested by letters from Smith's younger fans.

i2 (EYE) displays a more personal lyrical bent, and most of the recording was done in Smith's own home in Franklin, Tennessee. It began in the Bennett House just after tracking Amy Grant's album. Smith was involved in playing keyboards for her at the tracking stage. "Then we cut our own tracks in 32-track digital, and made a slave copy to 24-track analog. I continued working on the album in the studio in my house, which has an Otari 24-track and a Trident 80B console with Mastermix automation. And so all the vocals and overdubs went down analog. Here at Sixteenth Avenue Sound Studios in Nashville, we are syncing the original Mitsubishi digital tracks back up with my analog machine and mixing down to digital. So we've got both. We preserved the drums and the bass and a lot of the guitars digitally."

Smith has always been right at home with synths and studio technology. He was one of the first to use digital FM synthesis on an album, in the form of the Yamaha GS1 and GS2, before MIDI.

"The GS synths have a heavier sound. And I'm still not convinced that you can

duplicate that kind of sound on a DX7," he says. "So I still use them, even on this album. I'm a Yamaha endorser, but lately I've really come to love the Roland D50, and there will be a lot of it on this album, too." From his home studio, Smith has brought along two D50's, a Yamaha TX816, a DX7IIFD, a Kurzweil 1000SX String Expander, a Roland MKS20 piano module, and an Emulator II-HD jokingly restenciled to read 'Emulator III.' Smith's ancient ▶

L-R: Don and Doug Roberts.

Roland SVC vocoder sits in a corner, unused for the moment.

"Back home," says Don Roberts, "Smith also has a DX1, an old Prophet 5, a Roland Super Jupiter and an Oberheim Matrix 6R."

"I control everything from a Yamaha KX88 and a QX1 sequencer, even the GS2. I haven't found a MIDI kit for the GS1. If you find one, let me know!" implores Smith.

Back in the control room, I notice an E-mu SP12, which is being used to replace drum sounds triggered from hits played by drummer Paul Leim on the master tape, converted to MIDI and stored in a Macintosh SE running Passport's Master Tracks Pro. Hip indeed. Smith and company rely heavily on Don Roberts and his brother Doug, who own Roberts Music in Nashville, to provide the technical expertise that is necessary to translate their studio sound into live performance.

Roberts did all the pre-production sampling and sequencing for Amy Grant's Unguarded tour, and went on the road with Smith for the Big Picture tour. Grant and company were among the first to make use of sampled and sequenced backing vocals and vocoder effects, mostly from Grant's Emulator II. Don is now involved in sampling sounds from the new album.

"Brown Bannister and I are weighing the pros and cons of loading everything into either a pair of Emax HD racks or a single Emulator III," Roberts says. "I do all of my sample manipulation on a Macintosh, and all my sequencing in Master Tracks Pro," continues Don, "although we download all our sequences and much of the system exclusive data to a Yamaha QX1 for the road. We will be taking a Mac with us this time, but we depend on the QX because of its proven roadworthiness and its eight separate MIDI outs."

On the last tour with Smith, Don found himself managing a great deal of gear. "Smitty played a Yamaha KX88, and another stage keyboardist played Smitty's

DX1. We had three full TX816s and three Roland Super Jupiter modules. We had two of the original LinnDrums, and two E-mu Turbo SP12s, triggered by sequences and by the live drummer's Simmons SDS5 kit."

"At that time, managing concerts on that scale through MIDI was a relatively new idea," says Roberts, "and Smith's band went a bit overboard with the technology. They would constantly want to know if something was possible but a lot of times they wouldn't think if it was practical. With a little planning, you can keep things simple and still achieve results. For instance, the QX1 has eight MIDI outs. I always dedicate a certain output to a certain channel and use that for drum sounds, another for bass parts, others to melodic roles. We are looking into the possibility of live MIDI sound mixing automation. And the two keyboardists and the drummer on this tour will be triggering entirely separate racks of MIDI gear, going to separate groups of the 32 channels of audio I have to sub-mix backstage.

"Everybody enjoys working on our tours," he continues. "Roadies we hire who are not familiar with Christian music come away saying, 'There's something different here. A presence. There's something I like here.' And everybody gets treated like family. I felt that from the first day."

Lots of people are excited over the fact that Michael W. Smith and Amy Grant will be touring together this fall for the first time in years. "We will be touring Europe, and have three dates in England, in August," says Grant. "I've never actually taken a band over there. So that is the biggest stepping stone for us this time. Then we will do about sixty dates in arenas in North America. Australia and New Zealand are definitely part of the tour. This will be the third time that we've been there."

"I'm excited about it," says Smith. "We've worked together in the past. We did Amy's Age to Age tour together. We did the Straight Ahead tour, in '83-84. Those were very significant tours for both of us. It was magic. And now everybody's going, 'Gosh, Smitty and Amy are going to tour together again.' And we're feeling like this could be incredible!"

So who's afraid of contemporary Christian music? It can be as exciting, artistically satisfying and technologically excellent as any other form of music. It has to be judged on its own merits for its artistic creativity, just like any other form of expression.

"That's what you judge every mainstream album by," says Brown Bannister. "Every artist. I think you have to use the same standards."

Christian musicians believe the perspective that they put into their music is a unique one, one that needs to be heard today. And the only way to judge that claim is to go out and listen to it yourself. ■

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
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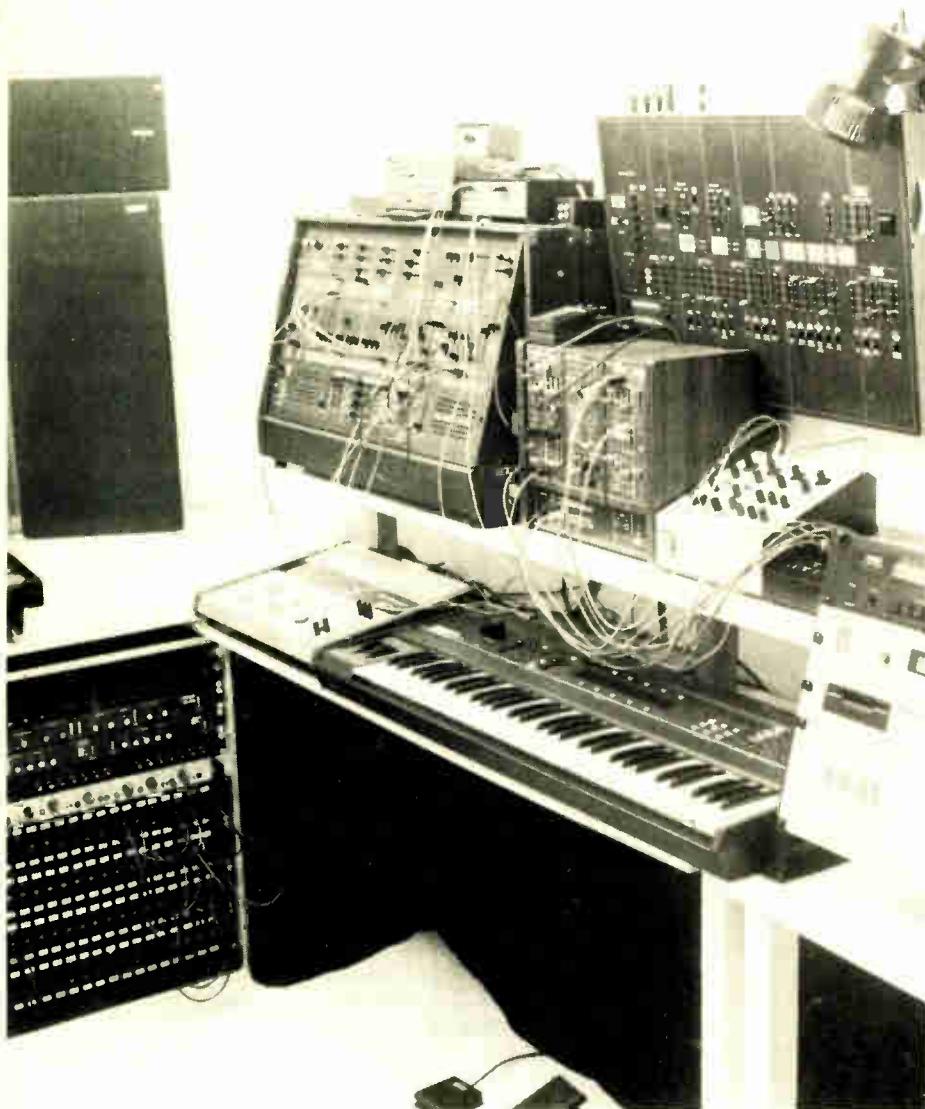
classic synths of yesteryear
2600
paths to happiness

Remember the days when a synthesizer was a collection of mysterious boxes linked together with yards of cable? Whether you do or not, take a nostalgic look with us at the ARP 2600.

Memories by Steve Howell.

with the alien mother ship at the end of *Close Encounters of the Third Kind*. ARP's first commercial success was the successor to the 2500, the 2600. Also a modular synth, the 2600 was essentially a slimmed-down 2500.

Pearlman decided that the rotary knobs on Moog equipment (and his own 2500) were not graphically representative of their setting, so he used slider controls throughout on the 2600. The idea (and one that was subsequently adopted by other companies) was that you could literally see a sound by looking at the control panel. Another move ARP made was to eliminate the necessity of using patchcords to connect modules together. Instead, patches were internally wired but could be overridden by inserting patch-cords into ⅛" minijacks on the front panel. ARP also saw fit to include an audio and/or voltage mixer at the front end of practically every module in the sound production chain. On Moog's modular equipment, you had to patch in separate mixers when you wanted to get fancy, which was not only laborious but also obscured the front panel with cords.

Modular Elation

LET'S TAKE A closer look at the hardware of the 2600. First off, there are three VCOs, each with a frequency range of 0.3Hz to over 10kHz, which could be used as control sources (ie. LFOs) as well as audio oscillators. These oscillators weren't as flexible as the 921 VCOs found on Moog's modular synths, but they were a lot more stable and consequently the 2600 didn't suffer the tuning problems of the early big Moogs.

Two of the VCOs offered square/pulse and sawtooth waves, while VCO2 also boasted sine and triangle waves. The 2600 was also (to my knowledge) the first synth to feature pulse width modulation. All the waveforms were available simultaneously for mixing from each oscillator, which is a feature I miss on today's synths. Each oscillator also had a three-channel voltage mixer for precise control of elaborate ▶

I N THE EARLY days of synthesizers, life was simple. There were only two main manufacturers offering a relatively limited range of instruments. Furthermore, these weren't being superseded every six months, so there was time to actually learn to use them. OK, we didn't have polyphony, touch sensitivity, programmability and all the other things we take for granted these days but somehow it didn't seem to matter.

The two manufacturers were, of course, Moog and ARP who, ironically, both no longer exist. Robert Moog needs no introduction, being the "father" of the synthesizer we know today, but Alan R. Pearlman is less well-known. Pearlman, the founder of ARP, tried to improve upon Moog's designs and provide a viable alternative to them.

The first ARP synth was the 2500 – an awesome modular affair seen "conversing"

▶ modulation on the input (say, keyboard plus vibrato plus FM).

The 2600 had a standard voltage-controlled lowpass filter with a 24dB/octave slope. It had a 5-channel audio input mixer, plus the usual resonance control. The ARP filter had a very different character than that of the Moog filter – in fact, Pearlman went out of his way to design a filter that would sound totally unlike the characteristic Moog filter, and in this he succeeded. To many's ears, the ARP filters aren't as "beefy" as Moog's design, but are a lot cleaner. Sadly, there was no highpass filtering on the 2600 – though it appeared a few years later on the less expensive Odyssey.

There was only one VCA on the 2600, which could be used to govern the level of control sources as well as the usual shaping of the volume envelope. It had two audio inputs and two control inputs, one of which offered exponential control over the amplifier's response, instead of the usual linear modulation option. The practical result of this was the availability of far more percussive envelope transients.

The 2600 had two envelope generators – one an Attack/Release (AR) type, the other a standard Attack/Decay/Sustain/Release (ADSR) type. All the transient slopes were variable between 2msec and 10 seconds, and were also exponential (as opposed to the more common linear). To my ears, this gave a certain undefinable "naturalness" to the 2600's sounds. Another case of the backwards march of technology is that exponential envelopes were de rigeur on all-analog synths, while computer controlled ones resorted to linear shapes because they were easier to implement in software. The EGs could be triggered from any external pulse as well as from the keyboard.

Other modules included a noise generator offering white, pink and low frequency noise types which could be used as a voltage control source if you so desired. There was also a serious programmable Sample-and-Hold module. This could sample the temporary level of any incoming waveform and hold it for a period of time in the form of a stepped control voltage output. The sampled control voltage could be derived from a low frequency VCO for arpeggios, or from the noise generator for random voltage changes. Another interesting application was to process the voltage from an envelope generator for stepped envelope transients (akin to the much later PPG's clicking effects). The S/H internal clock could be overridden and synchronized to an external pulse, such as a keyboard or sequencer (or an early drum machine) for synchronized sample and hold effects. In all, a very flexible module and a personal favorite.

A bi-directional clock allowed you to

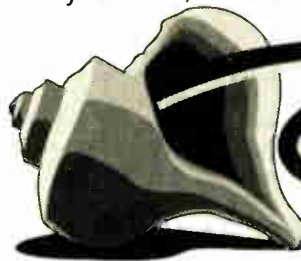automatically switch between two inputs (audio or control) at a rate determined by an external clock for chopping, or a form of quick patch changes. Also on the 2600 were two voltage processors, which were basically just voltage mixers/inverters as well as a source of +/−10 volts DC for manual control of modules (Zawinul used

to invert the keyboards on his two 2600s by sending them through this module). In the same section was a "lagtime processor," which added a variable amount of portamento to an incoming control voltage. This module could also be used as a very basic lowpass filter (since it in essence "smoothed out" whatever was being fed to it).

An external input preamp allowed processing of external audio signals, and an envelope follower generated an envelope voltage proportional to an incoming audio signal, which could be used to control modules (for example, using this with the VCF gave a dynamically controlled filter sweep) or to trigger devices such as the EGs or S/H circuit. There was also a ring modulator for the creation of clangorous sounds, but it could also double as a VCA. The 2600 also had a simple stereo output mixer with a pleasing (but noisy) stereo spring reverb. And if all that wasn't enough, it even had a built-in amplifier and speakers.

All the modules were designed and built to a standard more akin to test equipment than a musical instrument. This resulted in precise control over sounds as well as an exceptionally clean audio output; in fact, apart from the reverb, the 2600 is quieter than most of today's digital delights. That's progress for you.

Originally the 2600 came without a keyboard, although an optional "cheap 'n' nasty" affair was available upon request. Then came the "rev 2" 2600, complete with a four-octave duophonic (two-voiced) keyboard. Two separate pitches

could be derived and sent to different oscillators (though there really wasn't enough support modules to patch two entire voices) – which was a great improvement on the original. It had a built-in glide function and LFO so that you didn't waste a valuable modular VCO for vibrato effects. All very nice, but the rotary pitch-bend control was something of a dog, and was nowhere near as "musical" as the wheels found on Moog's designs. The ability to play two notes at once was a first for ARP and seemed quite a luxury in those early monophonic days. Interestingly, the keyboard was light enough to be worn around the neck as a remote keyboard.

Interfacing was easy and the 2600 (and all other ARP gear) was compatible with older Roland, Sequential and Oberheim equipment. If you intended hooking up to Moog, however, the story was quite different, as Moog was using "switch triggering" (S-trig) as opposed to ARP's voltage trigger (V-trig). The combination of a Roland MC4 Microcomposer (an early multitrack sequencer) and a 2600 gave you an absolutely evil compositional tool, and was well worth the trouble of sorting out

this pre-MIDI language barrier.

So there we are: an impressive line-up of modules that provided an alternative to the big modular Moog systems of the day.

The Bottom Line

SO WHAT DOES it sound like? To these ears, wonderful. The 2600 could deliver strong basslines, searing leadlines and an unlimited range of special effects. The fact that practically everything could be connected to everything else with a minimum of fuss gave the machine a
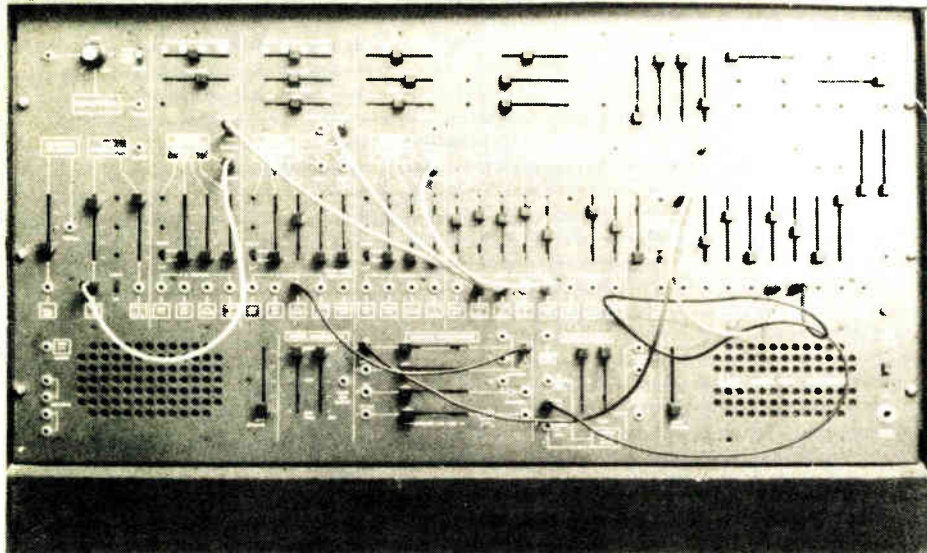
flexibility sadly missing from today's instruments. Because the VCOs could be used as control devices in the audio spectrum meant that some of us were dabbling in FM synthesis some 12 years before Yamaha gave us their DX synths.

Like most old analogs, the 2600 was littered with dedicated control knobs allowing you to alter any aspect of a sound instantly and hear the results in real time. In this respect the ARP is also graphically representative – thanks to those slider controls, you can see at a glance what's going on. And don't let anyone tell you that programming a modular synth is hard work – today's parameter-access synths are slow and tedious to program by comparison.

Sadly an ARP 2600 is hard to come by these days, but they are available if you keep your eyes peeled. When they were new, they were selling for a few thousand dollars, though the price dropped significantly just before ARP went under. Nowadays a second-hand model will cost you somewhere in the region of a thousand (less, if you're lucky). Don't worry about repairs – there are plenty of synth repairers that can fix them, and as components aren't too specialized (no VLSI technology here), you should have no problems.

If you're at all serious about synthesis, a modular system will go a long way towards helping you understand what's going on, and if you're contemplating spending your cash in this way you could do worse than provide a good home for one of these classic instruments (and sampling it to death). If your funds won't run to the 2600 but ARP's sound intrigues you, check out the Odyssey or the humble Axxe – they both share circuitry with the 2600, there's just less of it.

If you need any further recommendation, give Stevie Wonder, Josef Zawinul, Michael Boddicker, Tony Banks or Steve Porcaro a call and ask them what they think. If they're out ask me; I've had mine for eight years and, even now, I still write

"Because the 2600's VCOs could be used as control devices in the audio spectrum, some of us were dabbling in FM synthesis 12 years before the DX7."

pieces of music entirely on the 2600, resisting (temporarily) the attractions of an Ensoniq ESQl, Akai S900 and a Yamaha TX802 . . . I've MIDI'd up my 2600 with a Roland MPUI0I MIDI-CV converter, and so can take advantage of touch-sensitivity, better pitch-bend and a mod wheel that can be assigned to any of the 2600's modules. I'll admit the 2600 has its limitations – it's only monophonic (duophonic at best) and it certainly pre-dates programmability – but don't ask me to sell mine. I'd rather listen to a Flock of Seagulls. ∎

World Radio History

What today's mu
is a good do

For years now, Yamaha has been giving you the tools to explore new musical worlds.

But today we'd like to direct your attention to planet earth. Where you can capture sounds that are absolutely real, with the new TX16W Digital Wave Filtering Stereo Sampler.
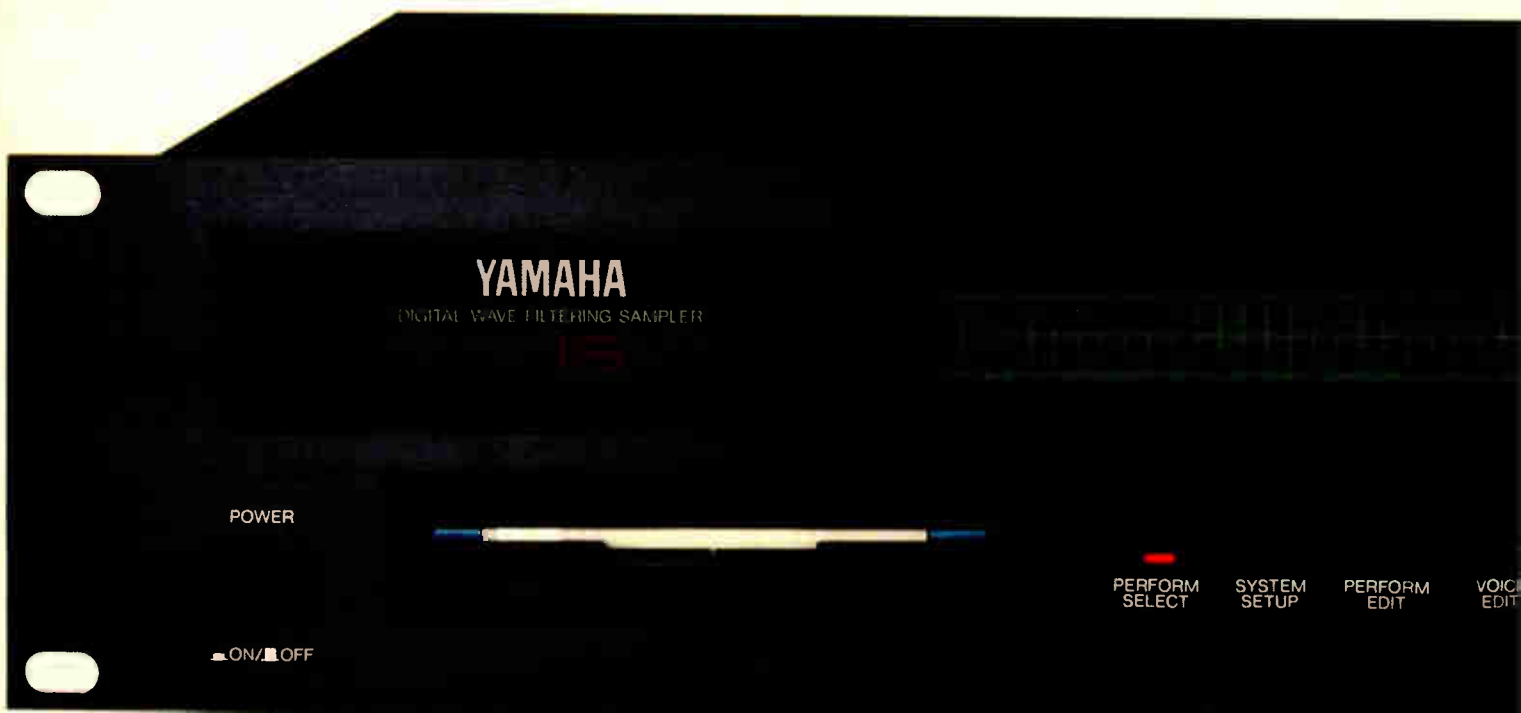
The TX16W allows you to sample anything you can hear, and then filter that sound while it's still in digital form. So you can create the cleanest possible tonal effects.

It also gives you the power to experiment with a wide range of sounds only possible with stereo sampling.

Perhaps even more important, the TX16W lets you build sounds in a very realistic way. For example, you can assign up to 32 samples to any single voice, for a natural effect across the entire keyboard.

In addition, you can layer up to 16 voices, using the Crossfade function to assign designated voices to different sections of the keyboard. Resulting in gradual tonal changes, as opposed to abrupt splits.

Likewise, a Touch-Crossfade feature makes it possible for you to combine designated voices as a function of key velocity.

YAMAHA
DIGITAL WAVE FILTERING SAMPLER

POWER

ON/OFF

PERFORM
SELECT

SYSTEM
SETUP

PERFORM
EDIT

VOIC
EDIT

sicians could use se of reality.

To give you an even more natural sound, the TX16W has 32 individual LFOs, one for each of the 32 timbres that can make up a single voice.

There's yet another LFO for each of the 32 digital filters. And for good measure, a global performance LFO.

Of course, in keeping with Yamaha tradition, the TX16W has a number of other features to make your musical life easier.

Like a built-in 3.5" disk drive that quickly saves or loads even complex setups. A 40 x 2 LCD you can read in most any kind of light. A numeric keypad for fast data entry.

As well as 1.5 megabytes of memory, which you can easily bump to six megabytes via plug-in modules.

And all this power in a package that takes up only two rack spaces.

If you'd like a sample of what the TX16W can do, stop in and see your nearest authorized Yamaha Digital Musical Instruments dealer.

We realize it may seem like too much sampler for the money, but that's the way it is.

Welcome to the real world.

⊛ YAMAHA

A PRoViSIoNAL arrangement

After a three-year hiatus Scritti Politti is back with a charting single and new album of immaculate white funk.
Interview by Tim Goodyer.

WHAT DO YOU think about live performance? "I enjoy it." What do you think is interesting or valuable about it? "I think you can create an atmosphere that you can't create on record. It's too easy to put a record on and be distracted; instead, you're shut in a concert hall for two hours and somebody says, 'I'm going to push my music down your throat whether you like it or not.' If you don't like it it's hell; if you like it . . . Either way it's an experience."

The interview is going smoothly – or it would be if I were asking the questions instead of answering them. Scritti Politti's Green Gartside and Dave Gamson have turned the tables on me and I find myself defending live music. They seem to have grown out of it, preferring the studio as an ideal environment in which to create their music. Although the music has been a long time coming. Scritti has remained ominously silent since the *Cupid and Psyche: 85* LP and its attendant singles, 'Wood Beez' and 'Absolute,' were released back in 1985. A new album, titled simply *Provision*, has brought about this unusual confrontation between interviewer and interviewee.

Getting back to the issue of live music, Gamson is in full flight.

"The end product is music; public performance, at least to me, is not that interesting. What we're dealing with is creating music and making records. I really couldn't care less if you're the greatest guitar player in the world. If there's another way for me to get that sound onto a record, I'll do it.

"What we're doing is not an improvised form in any way. What we're trying to do is to record a written song. Improvisation is not something that ever interested me."

And his sentiments are unequivocally shared by Green.

"Marveling at the dexterity of an individual on a platform is probably an unhealthy thing; it's certainly something that now seems very alien and old, even fascist in its own way. And it has no ardor for me.

"I'm not even happy with that distinction between what is written and what is improvised. I don't really believe in the mythology of this pure, untainted source of expressivity – it's garbage, it's metaphysical reactionary garbage.

"It was pointed out to me the other day that, as I'd started with punk I must be interested in the whole business of playing music live, but what actually interested me about it was independent records. People

had made records for a couple of hundred dollars. Our first record cost around $500 to make, that was what was appealing. Realizing a record was more appealing than putting myself on a stage in front of an audience."

Perhaps another tack . . . one of the cuts off the new album is 'Oh Patti (Don't Feel Sorry for Lover Boy),' which features the talents of jazz legend Miles Davis guesting on trumpet - after Davis had covered Scritti's 'Perfect Way' on his own album,

"What we're doing is not an improvised form in any way, we're trying to record a written song. Improvisation is not something that ever interested me."

Tutu. And the single currently shooting up the charts is 'Boom, There She Was,' which, along with 'Sugar and Spice,' features Zapp's Roger Troutman on voice box-treated synthesizer. In both cases the soloist's contribution has been improvised over Scritti's song structure and backing tracks. Not live music? Gamson to the defense.

"He's a very good musician responding to a situation. He's making it up, yes, but we've given him the rules."

"But then, we're making it up," interjects Green. "We're just conduits for our history and influences at a given moment. At the moment of writing, when you choose this progression or these intervals or this inversion over something else, it's as spontaneous as anything. He's thinking on his feet, but you're doing that when you write and when you sing. It's all just thinking as you go."

Gamson again: "Roger was playing, but the end product is our selection of what is going to work from what Roger's done. If we have ten tracks of him playing from beginning to end of the song, that isn't an interesting song. He was so nervous about doing the session that, at first, he found it difficult to play. It was only when he was able to loosen up and feel as though he was in the studio that he could be creative. Also, to some extent, what Roger's doing is responding to what we've got on tape."

ALTHOUGH GREEN LATER describes the interview as "adversarial," I appreciate his and Gamson's point of view. Indeed, popular music would be considerably poorer without it. Let's take a closer look at the nine songs that make up *Provision*. First of all, they've chosen to

release different singles in the UK than here - why?

"'Boom, There She Was' is the most like the last record in a lot of ways," comments Green. "I don't think we wanted to come out with something like that again. In England, it's better to throw it at a bit of a curve. Scritti's songs have always been a bit jittery and syncopated - well this one isn't."

"To me it's a definite continuation and an improvement on the last record," continues Gamson. "I think the vocals are much, much better and I think the arrangements generally are a little more open. We tried to take into consideration leaving a lot of space for the vocal, while still keeping the concept of how we build the arrangements. I feel much happier about this record than I did about the last one."

"This album has been a completely collaborative effort," Green comments. "Initially there was a little bit of a sense of David being brought in to a pre-existing Scritti as an accessory."

Gamson: "By the end of *Cupid and Psyche* we were working quite closely together and this record is a continuation of that. We went into it from the very beginning saying we're going to do this on equal terms. To me at least, if something happens on this record it's there for a reason; I tried to be very careful about ▶

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►where things were happening around the vocal."

As a result, *Provision* comes across as a refined continuation of the clean funk of *Cupid and Psyche*. Both are a far cry from the punk days that saw the conception of an almost unrecognizable Scritti Politti. Green is the only surviving member of the original line-up, Gamson only joining him for the making of *Cupid and Psyche*. Much has changed in both the album and single charts since the last Scritti long player, so where does Green see himself fitting into the popular scene of '88?

"There is a huge catalog of options offered to the consumers as to what they'd like to constitute themselves with. Aesthetic inclinations are choices people make either consciously or unconsciously to one degree or another. It would be impossible to generalize about music. I don't know where we are in it really. History dumps us pretty unequivocally in white pop territory; I'm afraid that it might dump us somewhere between Johnny Hates Jazz and whatever else is just around the corner. I've willingly lost all sense of where it's coming from or going to. I think that's a good thing.

"The brief that took the original Scritti into this Scritti was a reactive brief. But it's sustained enough interest for me – and, I suppose, for David – to sustain over two albums. I don't think of it in terms of being

happy with it, although I'm always dissatisfied.

"Whatever's conventionally thought of as more marginal music – whether it's the independent scene or whatever, they are the margins of conservatively designated space – there's nothing more inherently interesting or expressive or radical

happening there than anywhere else. That's why I embraced the idea of returning to a dominant aesthetic and I see no good reason to be anywhere else. Although I might – the map could be carved up again or I may want to be involved in carving it up again as much as it's possible to do so. I just want to emphasize the fact that I don't think there's anything inherently more challenging or truthful or more radical than where we are. You get involved in minutiae of pop, and the journalists and papers involved in packaging and presenting the consumer with a catalog of choices of musics that he can use to construct or deconstruct himself. That all tends to overlook the fact that pop music in itself is this gloriously enigmatic, pleasurable, meaningless/meaningful thing.

"But all you can ever do is talk around music, you can never actually refer to music – in the same way, music itself doesn't have a semantic level. I became hooked, when I was younger, on finding challenging musics. I would seek out records that initially frustrated me and were unsettling for me. The Beatles were an unsettling thing: each subsequent single was a sufficient departure from the last in terms of its language, its melody, its rhythms, to be a very big thing for a little boy – a little boy who took it very seriously. These things were thrown down as things of great power and beauty,

and troubled me. I searched them out and they led me to listening to rock 'n' roll: Matching Mole, Robert Wyatt, Henry Cow . . .

"I no longer have that cartography of the world – this is difficult music, this is easy music – I just don't think it's like that. But those musics are there to be found as challenging to listeners throughout the whole catalog of possibilities of music. There are musics that may even seem terribly anodyne to me and you, that hopefully are undoing little boys and girls throughout the country as they did me. I believe that to be the Great Hope. I'm retaining a willful naiveté about these things while theorizing at the same time and finding to my surprise and delight that I can keep both of those alive. Music is essentially so resiliently enigmatic."

TALKING ABOUT PROVISION in terms of the equipment that helped construct it proves considerably more straightforward. Being the more technical half of the partnership, Gamson takes the lead.

"All the drums were done on the Synclavier. We had a library of drum sounds that we transferred to the Synclavier and then we had a whole bunch more drum samples that we did in the studio. We already had all the drums sequenced and we did MIDI dumps into the Synclav and then we just went through and picked the drum sounds we wanted to use. That was all we used the Synclav for, really. Then it was just basically my keyboard setup on top of that: TX rack, Super Jupiter, Prophet 2000, DX7, Minimoog, Matrix 12, Prophet VS . . ."

Simple, really. Gamson came into Scritti through the band's decision to work in New York's Power Station studio (with help from producer Arif Mardin) on the Cupid and Psyche LP.

"What's happening now is that you're getting guys who are players and guys who are programmers," he explains. "The players don't usually have very big equipment setups and the programmers have ridiculously big setups but can't do very much with them. The players will hire a programmer to do all the programming while he just plays. I myself can't deal with somebody else doing my programming. If you know what a sound's supposed to sound like, it ends up being easier to do it yourself even if that means spending some time trying to do it. It depends on what kind of record you're making, but if you're going to make a sequenced record, a lot of what the sound is, is how it's being struck and how long it's holding for. When I sequence stuff I have to make sure every release is together and all the velocities are what I want so I can bring out certain notes within a chord. You could never do that playing – I don't care who you are, you could never have the kind of control you can have with a sequencer. It makes such a difference to the whole sound. I think it makes more sense to be a player who is also going to program because you have to control all the parameters."

And once again the idea of a "live" performance seems to go out of the window.

"I don't want to say that performance is bad or not valuable," Gamson continues, "but making a record is a totally different thing. When you're making a record you're using a whole bunch of things that don't exist in live playing. That's not to say that all live performance is uninteresting, but to me, live performance is uninteresting because I enjoy making records. ▶

"Arguably, music technology should lead you to have greater freedom in reshaping music – and I would testify to that being the case."

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BBE Goes on Jackson ''BAD'' Tour

The superb musicians in the Michael Jackson Band have a new tool to help them provide the musical framework for Michael Jackson's unique artistry. Music director Greg Phillinganes, Rory Kaplan, chief designer of keyboard systems and Chris Currell, guitarist and synclavier specialist, all have BBE 802's in their rack's for the tour. They love the marvelous clarity, definition and sheer musicality BBE provides.

▶ "The interesting thing about working in studios with electronic instruments is that what you're playing in the end is the studio. Hearing a violin in a room is a very different experience to hearing a violin over a speaker. When you go to a live pop performance you're hearing sound waves moved by large speakers which is very

David Gamson

different than going to see an orchestra. Each of these things is something you have to worry about; if you're going to perform using electronic instruments and speakers it's different to performing with acoustic instruments. If you're making records, they have to sound good over a speaker, which is different again to using a PA system. The Synclavier certainly doesn't seem to be designed to bridge the gap between live playing and the studio."

One of the more unusual aspects of the recording of *Provision* is Roger Troutman's contribution. His voice-treated synthesizer – usually assumed to be a vocoder – has long been one of Zapp's trademarks, but the details of the equipment he uses have · remained a closely-guarded secret.

"I think he likes to keep it a little mysterious," confirms Green. "Basically it's a bit of garden hose taped to a free-floating speaker and a Minimoog. It's amazing but it's obviously a whole different principle to a vocoder."

In fact, what Troutman has done is to build himself a DIY voice-box. These were most widely used during the '70s by guitarists such as Peter Frampton and Robin

Trower. More recently Bon Jovi used one on 'Living on a Prayer.'

"I saw Stevie Wonder doing all that with an ARP 2600 on TV back in the early '70s," comments Gamson. "I never knew what Roger used until he showed up."

Talking about the experience of working with Troutman elicits more natural

enthusiasm from both Green and Gamson than their considered views on the merits of live performance.

"He turned up in this tight-fitting, double-breasted red suit with red mock-snakeskin shoes, a little red tie, the hat and red-rimmed shades; it was classic," Green recalls. "And he had an identical outfit in blue which he wore the next day. He's definitely a throwback to 1974. He made us all put sunglasses on in the studio, and it was like a party. He's a wonderful man."

"We've both been fans and it's fun to get people that you've always admired on your record," says Gamson. "He was great, so quick –once through a song and he knows the whole thing. We got him plugged in, started the tape and just took it the minute he started. We got ten tracks of absolutely fantastic stuff that we had to wade through and decide what we were going to use because so much of it was just fabulous."

"Watching him work was like watching somebody possessed," Green adds. "He was suffering. He's so syncopatedly 'in there.' And with all that he has a great melodic sensibility.

"He's got a top five single, but he's still

something of an unsung hero. He's one of the few people keeping that whole P-funk flag alive in an age that's become a little arid as far as good grooves go."

Where did all the good grooves go? Could it be that our old friend technology has led the musician astray?

"You can say that about any instrument," replies Green. "The frets on a guitar lead you to a certain way of thinking about music. You're saying that *new* technology has led us, and I'm saying that *old* technology has led us. It's no more or less conducive to a conservative way of making music. Arguably music techology would lead you to have greater freedom in , undoing, unsettling, rewriting, reshaping music than before. And I would testify to that being the case in pretty general terms."

"You don't make music from an instrument," observes Gamson, "you make it from an idea. The instrument is only a way of communicating your idea. At any point in time there are going to be people doing uninteresting things – it could be with computers, it could be with anything. I don't see that making it easy for people is necessarily bad in itself."

In spite of their refusal to accept live performance as a useful part of Scritti's music, it transpires that concerts were once considered.

"We did rehearse to play live at one point," admits Gamson. "We said. 'Let's start from the song and work up.' What happened was that the people we got in focused in on the record – to them the song is the record.
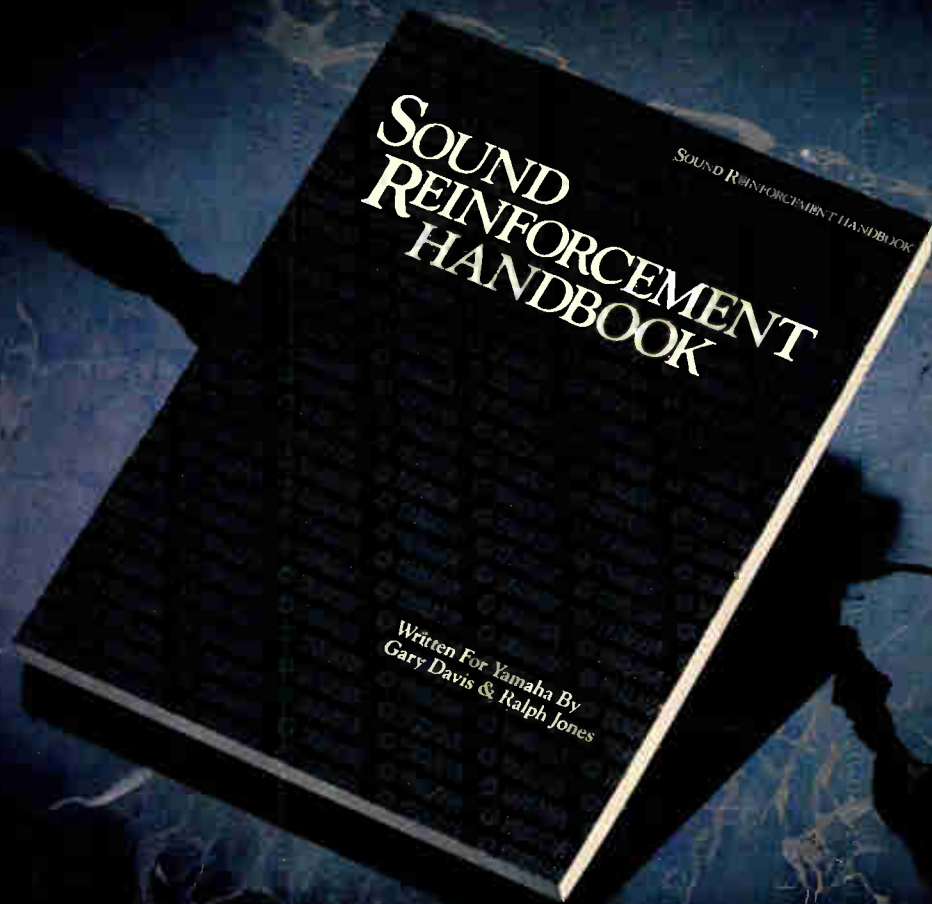
"But you'd have to take a radically different approach to doing it live. You certainly couldn't try to recreate the record, you'd have to totally rearrange it for live playing because we're talking about a very precise way of recording. Some of it is unplayable by a human being. And if it's sequenced live then what's the point? It certainly wouldn't be terribly interesting from our point of view."

But it must be interesting to Virgin · records who, having put money into the recording of *Provision*, must be eager to see a return on their investment – and a tour is one of the accepted ways of promoting an album.

"The music industry as a whole does expect you to go out and promote a record it has subsidized, and you're certainly up against some difficulty if you don't – and we don't, so we're up against some difficulty. It takes the form of making expensive records that then have to recoup that money through cover sales alone. I think Virgin has now got the message that we're not very keen, so they're not pushing us to do that."

Me, I'm still going to go to gigs, even if seeing Scritti Politti play live is one pleasure I'm likely to be denied. ∎

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Ever wish on

DIGITAL AUDIO 101 *theory*

Illustration *Christine Mallouf*

Digital technology is sneaking up on us all, so the time seems right to get the theory behind it straight. In Part One of a look at digital audio, we move quickly from basics to brawn.
Text by Peter Bergren.

THERE'S SOMETHING RATHER special happening in the world of music and audio technology which you are probably already aware of – the appearance of products which come equipped with a digital audio connector, in many cases allowing direct passage of digitized audio between one unit and another. Among these products are a digital mixer, an equalizer, a reverb system and various recording devices – including high-end units such as the AudioFrame and Fairlight III. These latter units can be properly called workstations, in that such functions as sampling, storage, sequencing, processing and other modifications all take place within one unit, with all signals remaining in the digital realm. Digital audio connectors are going to allow the signal to stay in the digital realm *outside* the unit.

What's really exciting about this development is that it now appears possible, or will be in the near future, to assemble a system out of separate components that will have at least some of the flexibility of the workstations, but for a lot less money. Whether or not this becomes possible for you depends on three things: compatibility between digitized audio sent by one unit and received by another (which we'll talk more about next month); your understanding of the basics of digital audio (which we'll dwell on this month), and

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your pocket book (which isn't any of my business anyway).

But please understand, we're *not* talking about MIDI, which is really a coded set of gestures – nor sound itself – passing from machine to machine. Rather, the advent of direct passage of digitized audio from device to device involves transmission and reception of the audio data itself – that which can then be modified and/or performed by MIDI instructions. A chief significance of this is that keeping audio in digital format from your system's input to output involves none of the losses commonly associated with the conversion of analog signals to digital and back again. To understand these losses, let's start the discussion with some review of analog processes.

Analog

IN A WAY, you can never really "hear" anything. Now give me a minute – don't write me off as completely nuts yet. Hearing is really a personal perception, rather than a direct experience. If the

classic tree in the forest falls, there's a physical disturbance, causing the air molecules to move, thus creating a sound wave. When we are in the forest and hear the tree, we are still several steps removed from actually having direct contact with the motions, actions and reactions of its fall. What we're really experiencing, or "hearing," is merely the compression and expansion of air moving against our eardrums. It is not the literal tree we are hearing.

In each stage of the hearing process, one form of energy in a particular medium is translated into another form of energy in another medium. What is common to all these stages is the replication of energy changes (amplitude, polarity, time factors, etc) from medium to medium, sound source to listener. Unfortunately for the energy, the mediums themselves are in a constant state of flux (even if viewed over very small periods of time), and therefore have continously changing characteristics. It follows that each energy transformation causes inaccuracies in translation, so that

some of the original information will be lost, or spurious information be added.

So, to take a Webster's stock definition of analog – something similar to something else – hearing something on analog tape ends up being an analogy of what really happened. To be fair, it's an almost exact replication, offering a real similarity to the real (I use the word loosely) thing, but there are going to be differences.

Analog Audio

IF SOUND IS an energy event that varies continuously with the passage of time, what's the most logical way to transmit it? Well, by some means that can be made to vary in a way that's similar to the soundwave, of course! Old Alexander Graham did this by varying the spacing (and thereby resistance) of carbon granules, thus modulating a current in step with air pressure changes. Edison made a permanent record of these changes by using sound pressure to cut a wavy groove in a wax cylinder, a groove that was similar in contour to the invisible fluxation in air pressure occurring as he said "Mary had a Little Lamb."

But there are a host of problems besetting this kind of approach, many of which take the form of noise and distortion. Unfortunately, most noise sources (such as dust in a record groove or hum from a turntable motor) have either a physical contour that varies continuously (such as the scratch in a record) or an electrical characteristic that does the same (such as white noise or radio interference). Systems sensitive to reception of the variations in sound waves are also receptive to the characteristics of noise sources. That's why you have to take care in handling your LPs, and also why companies make so much money on devices that can lower the noise level and increase the signal level in transmission, processing and recording systems.

But there's a limit to how high the signal level can be increased. There's no way you could engrave a groove in a record or saturate a tape with magnetic energy in a way that is really similar to the amount of energy emanating from a 16" cannon on a warship, for instance. The physical limits of the media prohibit such high energy levels, just as low level noise sources compete with signals on the other end of the loudness scale. And when you add the fact that these problems increase with each copy you make, analog methods don't seem like such a good bet after all. So what to do?

Digital Audio

SUPPOSE THAT INSTEAD of conveying the continuous variations of a sound wave as changes in another form of energy and medium, you make a series of many
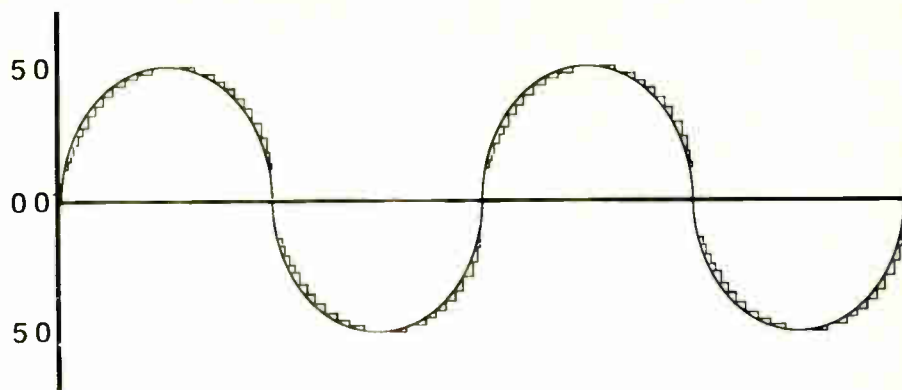
Figure I. Quantizing a l00Hz sine wave.

Figure 2. How a sample and hold circuit breaks up a sine wave. Shaded gray area shows error.

measurements of these changes at precise points in time? Then you can convey this information in a way that won't let analog noise and distortion be recognized by your system. This is precisely what digital systems do. They sample the analog waveform many times a second and quantize each measurement made during a sample period, so that a stream of discrete, unambiguous numbers is the result – numbers which represent changes in the sound wave's amplitude and polarity over time.

You're probably wondering just how in the devil you can accurately represent a constantly varying event with a fixed number of measurements, and I'll get to that shortly. Back to the digital signal itself.

Because each sample is a particular number, these numbers lend themselves to being encoded in binary form, as One and Zero bits, or if you will, as trains of On and Off pulses clocked at a consistently accurate rate through the hardware. This scheme is ideally suited to digital transmission, storage, and manipulation (just think about how quickly a light can be switched on and off). Outside analog noise ends up being too low in level to register as 1's and 0's on its own (okay – unless it's present at the input, right before the signal is digitized – but that ain't digital's fault), and therefore doesn't get registered at all. And achieving representations of really loud sound events is simply a matter of generating a larger number.

The sound wave can be looked at as a stream of numbers, thus, processing a number affects the audio. For example, making it greater and thereby making the sound louder is a matter of multiplication; making it less, for a lower sound level, is a matter of division. This explanation greatly oversimplifies, but I think you can see the advantages – computers are great at math, and (if proper care is taken, and big enough
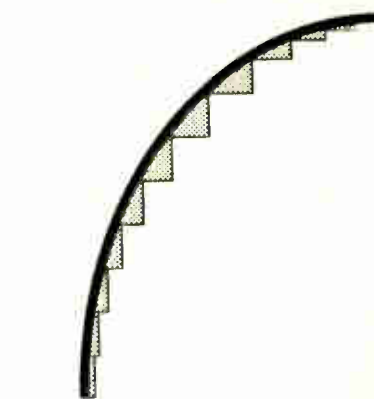
numbers used) performing math on a signal while it is in the digital domain is less prone to errors and noise than dealing with it in the analog domain. Some of the math that can be performed produces the equivalent of filtering, EQ, mixing, reverb, pitch shifting, delay and the like. As a matter of fact, almost any analog process can be analyzed and an algorithm, which consists of a program, be devised which can then direct a multipurpose set of hardware. Neeto, eh?

Sampling/Quantizing

IMAGINE A PIECE of graph paper divided into one hundred segments vertically, and one hundred horizontally, with a horizontal line drawn midway across. The vertical scale is divided into one hundred voltage divisions consisting of 50 positive and 50 negative, with the midway line being the zero voltage point. Adding the total of positive and negative voltage equals one volt. The horizontal scale represents one hundreth of a second, divided into one hundred parts.

Now imagine one cycle of a l00Hz (100 cycles per second) sine wave drawn on the graph paper, with its zero crossing exactly midway on the horizontal midline of the graph. (See Figure I.) Let's describe this ▶

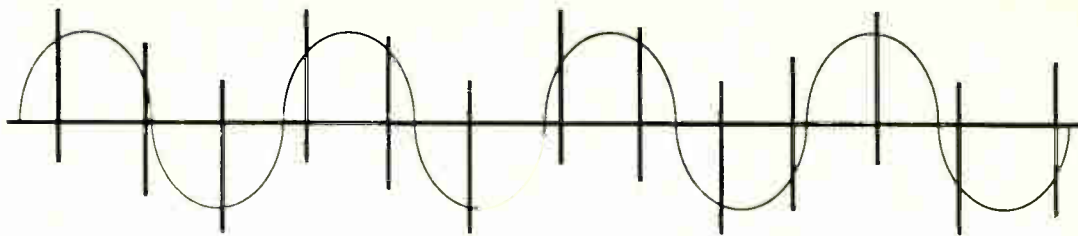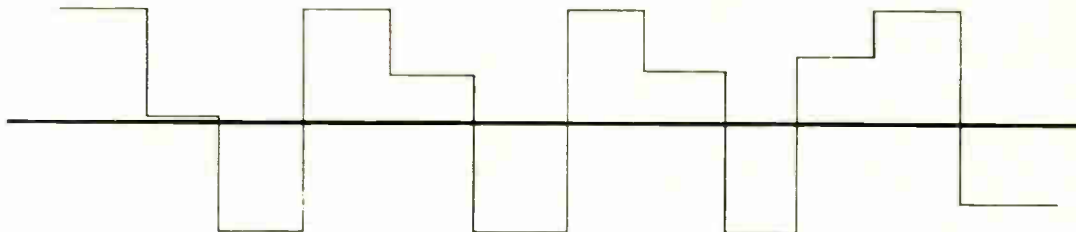**Figure 3A. Trying to sample a sine wave more than twice the sample rate . . .**

Figure 3B. . . . results in an alias frequency lower than the original.

waveform in strictly numerical terms, in base ten, decimal notation.

To do this you have to relate time to voltage, and come up with a "data stream." It's fairly easy, if rather tedious, to do this. Simply correlate each time value on the horizontal axis with its corresponding voltage value on the vertical axis, being careful to note the polarity of the voltage read-off. Read upwards from a time value to its intersection with the waveform, then across to the voltage level that comes the closest to also intersecting at the same point on the waveform. Note that time and voltage intersections don't always coincide exactly on the waveform line. This is called quantizing error, which is not of as much significance as you might think, save for generating a small amount of white noise-like interference.

When you're done you should have two columns of one hundred numbers each, both starting with zero. The decimal representations of voltages can be relatively easily changed to base two (binary) notation, consisting of strings of Ones and Zeros. Once in that form, they can be easily stored as pulses/no pulses on a tape or disk, or conveyed by a wire, fiber optic cable or whatever.

Modern digital systems will sample at 44.1kHz, or 48kHz, and in a 16-bit linear format offer 65,536 measuring, or quantizing, increments. This large number derives from allowing 16 bits, each of which can be on or off, to form a byte, or word, describing a particular voltage measurement taken during each sample period. Since each bit can have two states, and there are 16 of them, you have 2 to the 16th power, or 65,536. An 8-bit word by comparison is 2 to the 8th power, or 256 measuring increments. This disparity

creates a considerable discrepancy in dynamic range between 16- and 8-bit systems.

Rapidly varying (high) frequencies and transients require more sampling intervals than low frequencies, and defining very loud versus very soft sounds requires more number choices. Also, the more measuring increments you have available, the less likely it is that a small change in the analog waveform will occur between two measurement increments, assuming also that the sampling periods coincide so as to catch the change. When a small change like this does occur, the bipolar nature of change in the analog waveform is not represented by a change in the voltage levels recorded by the digital system, and distortion results, which will be reproduced unless certain measures are taken. But I'll get to these later.

It's important to realize that any analog-to-digital conversion is always subject to quantizing error, because you're trying to pin down a continuously varying process

hybrid analog waveform that consists of discrete "stairstep" voltages that rise and fall with the contour of the sampled waveform, much like stairs going up and down a hill. The ratio of stair "riser" to "tread" measurements varies with how long a given voltage value remains for a given number of sampling periods, but the form of these stairs could be considered to be fairly symmetrical for a pure sine wave.

The sample/hold circuit forms this staircase by capturing the voltage in the waveform present at the beginning of each sample period and then holding this value until the next sample period. This hold function creates the "tread" in the stair step. At the next sample period, the value of the next held voltage changes, because the analog waveform has itself changed value, continuously, between sample intervals. This change in voltage level to that captured and held during the last sample period creates the "riser" in the stair step. (See Figure 2.)

The important point here is that the

"In a way, you can never really 'hear' anything. What we're really experiencing, or 'hearing,' is merely the compression and expansion of air moving against our eardrums."

with stable numbers. So how important is this error, especially at high frequencies where, for example, you would have roughly two samples (44.1kHz rate) taken within one cycle of a 20kHz waveform?

Sample and Hold

TAKING THE 100HZ wavecycle we graphed earlier as an example, let's imagine what happens when it enters the sample and hold circuit of an A-to-D converter. This circuit is analog, in that it creates a

stair step voltages more or less follow the contours of the analog waveform. I say more or less because quantizing errors are always present when the actual value of the waveform doesn't exactly coincide with a quantizing (measuring) increment at the start of each sample period. Nevertheless, the stair step waveform can be decoded back into a quite accurate recreation of the original waveform it represents by the simple expedient of filtering.

You see that the stair treads really are a bunch of square waves (or in some cases

52

rectangular waves). As you may or may not know, square waves are formed when all possible odd harmonics are combined. This can be thought of as a collection of sine waves starting upward from a given fundamental frequency (another sine wave). In the case mentioned, that of a 100Hz waveform with an amplitude of one volt peak to peak, there would be 480 such square wave "treads" following the contour of each wavecycle, and 48,000 of them per second. This assumes we're now sampling at the 48kHz professional rate rather than the rate implied in our example with the graph paper. If we pass this stair step wave through a filter with a "brick wall" stop band that cuts in at, say, 22kHz, all the stair step components will be deleted. And what will be left will be the 100Hz sine wave whose contour they were rising and falling with.

In the case of a 20kHz sine wave, as long as it rises and falls within at least two sample periods and crosses at least two quantizing intervals, a square wave representation will be formed by the sample/hold circuit. By definition, this square wave will contain odd order harmonics higher in frequency than the sine wave it represents; but when it runs into the filter, the square wave will lose all harmonics that are odd order (180kHz, 9th harmonic; 140kHz, 7th harmonic, etc) and become the 20kHz sine wave it wants to be!

If by chance this sine wave is more complex and contains a harmonic "wiggle," this harmonic would be far too high to hear – and would be filtered out in any event. It should be noted here that this filtering happens at the system output, after the waveform time/voltage measurements stored in binary form have been reconstructed back into a stair step waveform. This filter is often called an 'anti/imaging filter, for it also removes high frequency multiples of the input waveform's frequency response curve (spectrum).

It should be obvious by now that the more quantizing increments you have available, and the more samples taken per second, the more likely it is that the varying line of the analog waveform will land on the intersection of a sample period and a quantizing increment, or at least come very close. Manufacturers have decided that a 44.1kHz sample rate and a 16-bit quantizing word are sufficient (for now – progress marches ever onward) to accurately encode and decode signals in the normal audio range. Large amplitude and complex signals relate randomly to any errors that are generated, and the errors, when reproduced in a digital system's analog output, are low in level, resembling white noise. When the signal level being quantized/sampled drops, however, the errors become correlated to these smaller excursions of the signal, for there are times when the signal's change in amplitude does

not cross more than one quantizing increment.

For reasons too complex to describe here, the above generates a nasty form of distortion known as "granulation noise." One way around this is to introduce a small amount of white noise to the incoming signal called dither. This becomes superimposed on the waveform, and effectively causes low level in the waveform to cross two quantizing increments instead of one. When the resulting data is recreated and filtered, the result is the original waveform sans distortion, but with a negligible amount of white noise added. This situation often occurs when the minute harmonic "wiggles," riding on the larger excursions of a complex wave, don't quite cross from one quantizing increment to another between sample periods.

Aliasing and the Nyquist Theorem

A FELLOW NAMED Nyquist came up with a very interesting theory, which he

"When it runs into the filter, the square wave will lose all harmonics that are odd order (180kHz, 9th harmonic; 140kHz, 7th harmonic, etc) and become the 20kHz sine wave it wants to be!"

proved experimentally, and which has since come to be the basis of the design of all digital systems. Stated briefly, he said that the highest frequency one can convey by a digital sampling system must always be at least one half the sampling frequency. Thus, if you wish to record 20kHz, your sample rate must be at least 40kHz. If you try to record any frequency higher than the 20kHz limit, the analog-to-digital converter (ADC) will not be able to take two samples per cycle; meaning it won't be able to accurately show the bipolar nature of the waveform and its frequency.

Remember that two samples must be taken and two quantizing increments must be crossed to create a square wave (via sample/hold) of the same frequency as the wave sampled. Beyond this Nyquist limit, the sampler will take samples of the rapidly varying amplitude of, say, a 30kHz waveform, but catch samples at points not really related to each cycle of this waveform. The result will be descending frequencies ("aliasing" frequencies, since the original is being represented as something else) which take the place of the original harmonics. (See Figures 3A and 3B.)

For example, if S equals the sample rate of 48kHz, and F equals the 30kHz "defiant" harmonic you wish to capture, then $S-F$ equals the spurious harmonic added to the legitimate frequencies below the Nyquist limit. In this case, you would suddenly find yourself with an 18kHz signal never present at the input. This and other frequencies formed by similar interactions cause aliasing distortion in the audible signal. To avoid this, a filter with a flat bandpass and very sharp stop band is inserted ahead of all other elements in the digitizing system. In some ways it's similar to the anti-imaging filter at the tail end of the chain, and it's usually called an anti-aliasing filter.

Fading Out

SO MUCH FOR a few of the, uh, "fundamentals." Obviously there's a great deal I haven't covered or have had to glance over. I'm indebted to Ken C. Pohlman, from whose excellent book *Principles of Digital Audio* I was able to draw much useful information while compiling this article. I suggest you buy it (Howard W. Sams and Company is the publisher) if you want to learn more.

Next month I'll outline a complete PCM recording transmission system, discuss some of the more salient aspects of D-to-A and A-to-D converters, and cover some of the new all digital products coming on to the market. In particular, I hope to make clearer the do's and don'ts of digital interconnection.
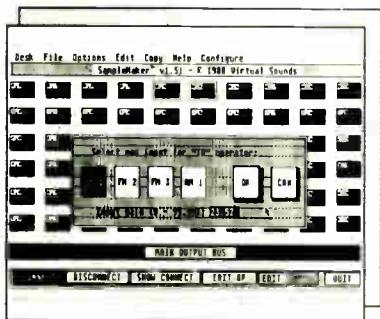
It's a whole new world out there. ■

COMPUTER
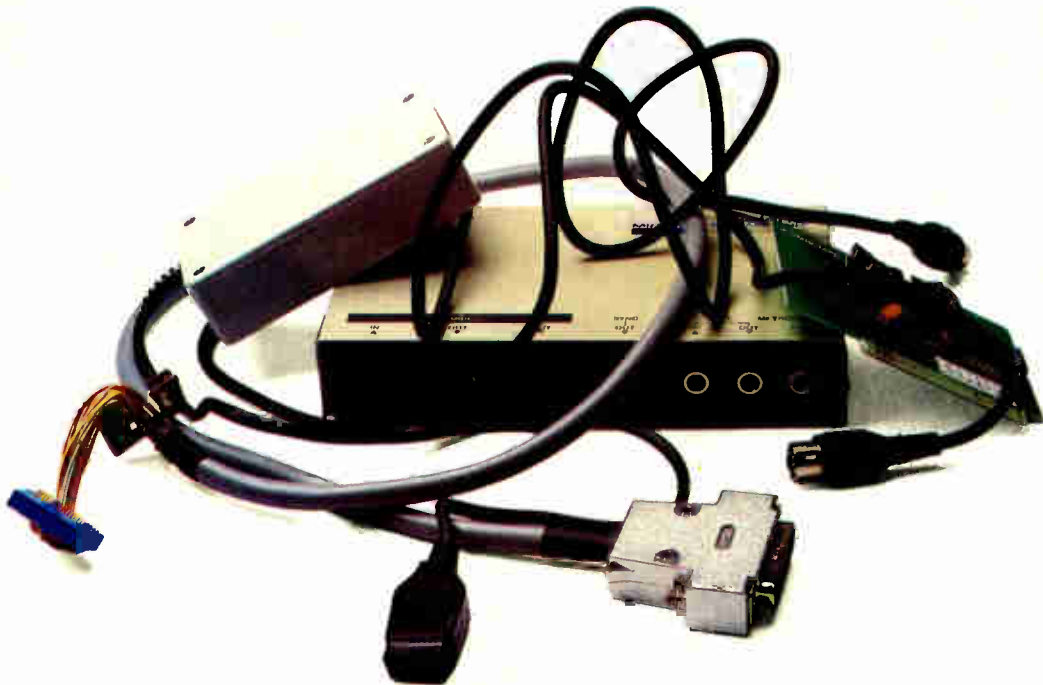## N O T E S

COPY PROTECTION
An Overview

REVIEWS
Graphic Notes
Dr. T's MPE
PC Desktop Studio . . . and more!

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

Of all the personal computers you can buy to make music, none makes it easier than Atari computers.

That's because, unlike the others, the Atari 512-kilobyte 520ST™, 1-megabyte 1040ST™, and 2- and 4-megabyte MEGA™ computers have more of what you need already built-in.

Here's what we mean.

MIDI. The Key to Electronic Music.

As you may already know, the MIDI interface is the key to electronic music.

If you're unfortunate enough to not be working with an Atari, you'll have to buy an interface separately.

And make sure it's compatible with the rest of your equipment, not to mention your software.

And then you'll have to make sure everything is installed correctly.
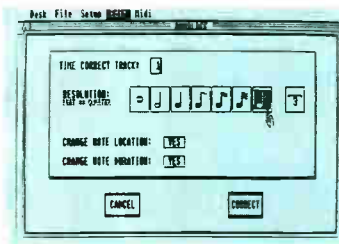
What's that like?

You know the song, "What are you doing for the rest of your life?"

Atari ST™ and MEGA computers, on the other hand, have a MIDI port built right into the back of the computer.
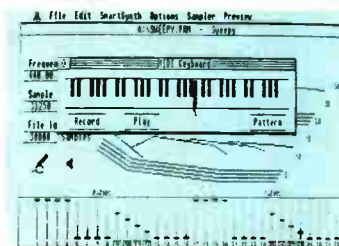
So you can connect all kinds of equipment—synthesizers, samplers, drum machines, SMPTE controllers, pitch-to-MIDI converters—as easily as plugging into an amp.
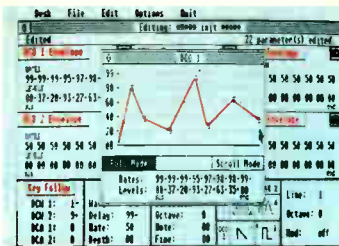
CZ-ANDROID™ Hybrid Arts™

MIDISOFT STUDIO™ Passport™

SOFTSYNTH™ Digidesign

THE COPYIST™ Dr. T's™ Music Software

A Musician's Music Box.

No other computer company has made the commitment to music and musicians the way Atari has.

That commitment, by the way, doesn't end with our hardware.

We're working in harmony with all the major music software houses to produce the software you want to make music with.

And building a distribution network of music dealers—not computer dealers—who know electronic music well enough to help you, no matter how much you know.

This Should be Music to Your Ears.

The Atari ST and MEGA computers are just parts of a full system. So there are lots of things you can add when you're ready.

Like our MEGA File 20™ 20-megabyte hard disk for storing your magnum opus.

And our SLM804™ laser printer for publishing it.

Plus one of the largest libraries of music software in the industry.

But perhaps the nicest thing about an Atari is how little it costs.

With what you save on an Atari, you could buy yourself a synthesizer. And some software.

Want to learn more? Write or call for the name of the Atari music dealer nearest you.

⅃ℿ ATARI®

World Radio History

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

News
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COMPUTER NEWSDESK

YAMAHA PORTABLE COMPUTER

Right before we headed off for NAMM, Yamaha told us to come by their booth and check out their new computer(!), the CI. The product of 2½ years of development, it's a laptop AT clone featuring a 10MHz 80286, dedicated SMPTE in/out, two assignable sliders on the front panel, and eleven MIDI jacks in all – two in, eight out, and one thru. The display is 640×400 square pixel backlit Supertwist LCD with normal 1:1 aspect ratio, and there's outputs on the back for external Hercules, CGA, or monochrome Multisync II compatible monitors. Extras inside include an extra timer for applications, a dedicated processor for handling the sliders and SMPTE jacks, the ability to handle all MIDI jacks at full bandwidth using a custom Yamaha communications chip, and a ROM with a music font built in. On the outside, you'll notice note numbers screened above the normal ASCII keys, two serial outputs, a printer port, and the ability to expand the internal 1Meg of RAM up to 2.5Meg. It runs normal IBM applications and comes with DOS, a pair of MIDI utilities (MIDI monitor and a generic patch librarian), and a carrying bag. There is extensive developer support (including an application toolkit), and several IBM music software companies will apparently have their wares running immediately on the CI.

Price for the two floppy version will be around $3000; for one floppy and an internal hard disk, around $4000. It looks like the Macintosh is in for some competition as *the* portable music computer.

MORE FROM Yamaha, 6600 Orangethorpe Ave., Buena Park, CA 90620. Tel: (714) 522-9011.

INEXPENSIVE NOTATION

Grandmaster, Inc. has just announced the release of version 2.0 of their "music processing program," MusicEase. A WYSIWYG (what you see is what you get) music notation editor for the IBM PC, it can accept note entry from either the computer keyboard or MIDI. It includes a 1-2-3-like menu interface plus a Wordstar-like set of commands, plus multiple (non-overlapping) windows. More eye-catching features include an unlimited number of connected staves; up to three lyric lines per staff; the ability to stretch a staff to an exact measured width; and the ability to transpose, shift, invert, retrograde, or add to each staff, block, or entire piece. Requirements are an IBM or workalike, DOS 2.0 or higher, 512K of RAM, CGA or EGA, an Epson or IBM Proprinter, and a Roland MPU401-compatible MIDI interface.

MusicEase – a WYSIWYG music notation editor for the IBM PC.

Price is just $99.95 for the un-copy-protected disk and manual; a demo disk is available for $5.

MORE FROM Grandmaster, Inc., PO Box 2567, Spokane, WA 9920-2567. Tel: (509) 747-6773.

PASSPORT MASTERS MORE COMPUTERS

Along with upgrading their popular Master Tracks Pro sequencer to level 3.0, Passport has ported it and its baby brother Master Tracks Jr. to a number of computers. New features in version 3.0 include support of two independent MIDI interfaces, synchronization to MIDI Time Code as well as song position pointers, a SMPTE window with continuous timecode readout, and a Marker window that displays each marker's song position and SMPTE time. These enhanced markers can act as a cue list that can be printed out, and can be placed anywhere within a sequence (to one clock accuracy) for sound effects work. Version 3.0 also includes MIDI controller chasing, which means it will send out the most recent continuous controller and pitch-bend values when started in the middle of a sequence. This makes it well suited for MIDI automation.

Both versions of Master Tracks run on the Macintosh, Atari ST, and now the IBM PC (under Windows). The Junior version also runs on the Apple IIGS (an all-too-rare new music package for that computer). Pro retails for $395; Junior retails for $149.95. Owners of older versions of Pro may update to 3.0 for $45.

Also, Passport has a free booklet (*Passport to MIDI*) for those new to MIDI sequencing.

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

MORE LIBRARIES THAN BOOKS

With the DX7 having been beat to death, the MT32 and ESQ1/SQ80 synths have been getting deluged with patch editors and librarians.

On the MT32 side, Rigamer Technology has just released MTED, which runs on the IBM PC and allows access to all those functions hidden by the MT's normal front panel. Features of the editor include holding two timbre groups of 64 patches along with an additional eight at once; transferring of any single, group of eight, or group of 64 timbres; patch mapping of all 128; and assigning of rhythm sounds or timbres to any key with its own level, stereo placement, and reverb switch. MTED requires an IBM workalike with only 256K of memory, DOS 2.0 or higher, and a Roland MPU401 (or compatible) MIDI interface. Cost is $79; a demo disk is available for only $5.

For the ESQ family, Playroom Software has announced MVP-ESQ (also for the IBM, requiring Windows 1.0 or 2.0). It allows editing of patches either as an ESQ patch sheet (with pop-up graphic envelope editing) or akin to the front panel flow chart (also with graphic envelope editing). The Syntrak function displays on the editor's screen any tweaks being performed at the ESQ itself. There is also the provision for selective patch randomization.

MORE FROM Rigamer Technology, 2470 Prentiss Avenue, New Orleans, LA 70122.

Playroom Software, 7308-C East Independence Blvd., Suite 310, Charlotte, NC 28212. Tel: (704) 536-3093.

QUIET LION; LOUD PRODUCT

Like the Volkswagen Bug, the Commodore 64 won't die. Quiet Lion has just released a comprehensive sequencer for it that is intended as a companion to their scoring program, Music Printer GS. Features include individual event editing, cut-and-paste type operations, programmable punch ins and outs, meters from 1/4 to 24/16, velocity scaling and muting, automatic fixing of stuck notes, and splitting a track above and below a particular note. Limitations seem to be only eight tracks, and a "plain" user interface that relies on alphanumeric keys to execute functions. It supports the Passport or Sequential MIDI interface along with the Commodore's

SID chip, and can read and write both Sonus and Syntech file formats. Price: $79.95.

MORE FROM Quiet Lion, PO Box 219, Sun Valley, CA 91353. Tel: (818) 765-6224.

ANOTHER LIVE ONE

RTX8 is a MIDI processor that runs on the Commodore 64 computer with a Sonus or Passport interface. Designed to split a master keyboard across multiple channels for live or layered performance, it allows the creation of eight assignable zones. Each zone includes the following parameters: MIDI channel, low note, high note, transposition, sustain pedal effect, and mapping of any continuous controller to the MIDI master volume control. Other useful utilities include a panic button and MIDI data monitor (either one worth the $39 list price). Up to eight setups reside in memory at once for instant selection.

MORE FROM Realtime Music Publishing, PO Box 8144, Loveland, CO 80537.

STRANGE SNAPS

Snap Software is making a career out of writing patch editors for otherwise unsupported products. They began with programs for the GM70 and GP8; their latest offering is for the Yamaha MEP4 and runs on the IBM PC. MEP4 Companion, as it's called, allows access to every MEP4 parameter on a pair of screens. Features include within-patch copying from one processor to another. Available only by direct order, the program costs $100; a demo version will set you back $10.

**MORE FROM** Snap Software, 1116 Janey Way, Sacramento, CA 95819. Tel: (916) 451-9914.

WIDER VISION

SampleVision, a visual sample editing package for the IBM PC, has been updated to support the Ensoniq EPS, Yamaha TX16W, E-mu Emax, and any sampler using the MMA Sample Dump Standard along with the original Akai S900. Aside from all the expected features, some special treats include equalization, inversion, reversion, color FFT's, and the ability to locate a place in a sample by sample count, elapsed time, file percentage, or any type of SMPTE timecode. SampleVision requires an IBM-compatible with 640K of RAM, a hard disk, CGA, EGA, or Hercules graphics interface, and a mouse. Retail price is $349.

MORE FROM Turtle Beach Softworks, PO Box 5074, York, PA 17405. Tel: (717) 757-2348.
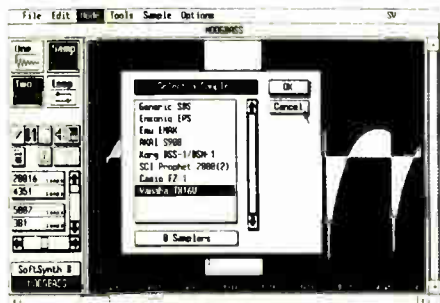
SampleVision, a visual sample editing package for the IBM PC offers some unexpected features.
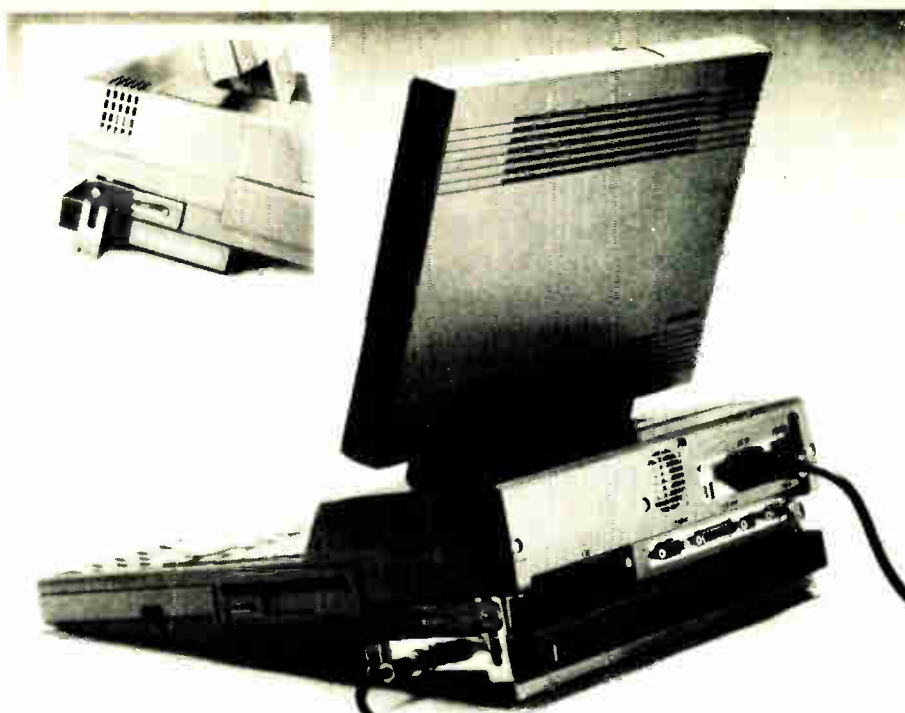
WonUnder, an accessory for Toshiba laptop IBM-compatible computers.

TOSHIBAS AND MIDI

Connect Computer has developed WonUnder, an accessory that bolts onto the bottom of the Toshiba T5100, T3100, T1200, or T100 Plus laptop IBM-compatible computers which allows any single ⅔-length (or less) expansion card to be used with that machine. This, of course, includes MIDI cards, for portable composition or live performance. The Toshibas will still fit back into their cloth carrying case, and will be less than two pounds heavier (and your wallet, $379 lighter) for the graft.

MORE FROM Connect Computer, 9855 West 78th Street, Suite 270, Eden Prairie, MN 55344. Tel: (612) 944-0181.
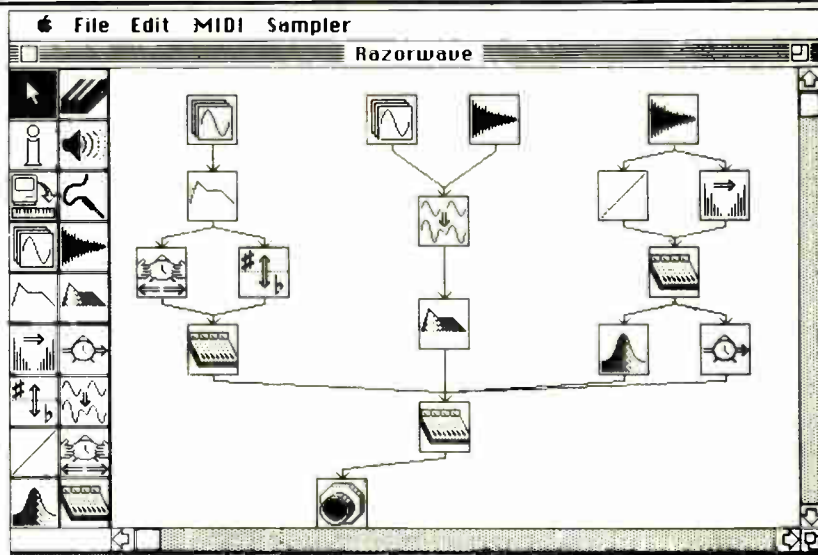
Digidesign's new TurboSynth – a palette of sound modules interconnect to create samples for nearly any available sampler.

BEYOND SOFTSYNTH

Digidesign was the first company to prove that sample editors, and then software synthesis packages, were viable products in the marketplace. Whereas SoftSynth concentrated on additive synthesis, their new TurboSynth package gives a mouse-and-windows treatment to the patchable modular synths of yesteryear. A palette of sound modules (which include sample playback and FM, additive, and waveshaping synthesis techniques, along with ten signal processing modules from spectral inversion to digital delay) can be interconnected in any fashion desired to create samples which can then be downloaded into nearly any available sampler. Running on the Macintosh and costing $349 list, TurboSynth will also be compatible with Digi's Sound Accelerator card for the Mac II and Mac SE. This combination will allow real-time synthesis and 16-bit performance of the sounds from the Mac via MIDI.

MORE FROM Digidesign, 1360 Willow Rd., Suite 101, Menlo Park, CA 94025. Tel: (415) 327-8811. ∎

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

News
Reviews
Previews
Applications
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C O M P U T E R

Graphic Notes
Music Publisher 1.01
Notation Software

A company from Down Under enters the notation fray with a Macintosh program featuring a dedicated piece of hardware for note entry. *Review by Wheat Williams III.*

GRAPHIC NOTES IS the name of an Australian music publishing company which has released the software it developed for its own use to the general public. Designed to meet the needs of a professional publisher, it is easy enough to learn and reasonably priced, and looks to be useful to the average musician in his or her basement studio.

A great deal of excitement has been generated by the releases of the new generation of music printing software for use on the Apple Macintosh with the Laserwriter and other Postscript printers. In my article last month, I described Music Publisher as an eagerly-awaited

entrant into this growing field. This month, I actually got a review copy. Does it deliver? Yes and no.

WHAT YOU GET
First, name confusion: Music Publisher is the name of the program. Although advanced ads mentioned the name "Fortissimo," this apparently refers to a multiple user version of the program which is still being developed for professional music publishers. The Postscript font of musical symbols included with the program is called Repertoire. The additional 36-key keypad supplied with the program is called

Presto. Also supplied with the program are two sheets of special clear Mylar stickers to apply to the Macintosh's keycaps. The only thing left in the package is a pair of nonstandard cables for interfacing the Presto keypad to a Mac Plus, and a more standard cable that will work serially with the Mac SE or Mac II's ADB port. Finally, the program's 152-page owner's manual is a very good one; adequately explaining each function in a logical order with lots of illustrations.

SETTING UP THE PROGRAM
Music Publisher will only work with Apple's System 4.1, Finder 5.5, and Laserwriter 4.0 or

higher, and requires at least a Mac Plus. For reasons discussed below, you would be much better off with a Mac SE or a Mac II and a large-screen monitor. (I'm hoping my Mac Plus will still be useful for something other than a doorstop in six month's time.) To facilitate configuring your working disks, Music Publisher comes with Apple's old Installer utility. Music Publisher's Installer script will reconfigure any disk you feed it by replacing your current System with Version 4.l (with the four sizes of the Repertoire screen fonts) and Finder with 5.5. It also copies over the Music Publisher application itself, and the Repertoire Postscript font. This utility is useful if you don't know anything about how to drag icons in and out of folders or use Font/DA Mover. Personally, I wouldn't touch Installer with a ten-foot SCSI cable.

If, like me, you already use a higher System and Finder than 4.l and 5.5, and are paranoid about letting an automated utility indis-criminately restructure the Blessed Folder on your hard disk (and all your hard-won fonts, DAs, and CDEVs), you can manually place the necessary applications and move the Repertoire screen fonts out of the System on the Installer disk with Font/DA Mover. Word is the program is now also directly Font/DA Mover compatible.

Graphic Notes felt that the best way for users to learn the layout of all the musical symbols which are entered from the Mac keyboard was to provide clear plastic keycap stickers with the symbols printed on them in blue, green, and red. These indicate which symbols are triggered by the normal, Shift, and Option key combinations, respectively. This makes a lot of sense, because you don't have to memorize long strings of command combinations (like those found in Passport's program Score! for the IBM) to typeset music. Installation is simple providing your keyboard isn't caked with pizza, and once the stickers are in place they stay there and do not obscure your view of the little QWERTY letters printed on them by Apple. You'd better be careful while applying them; they aren't removable.

HOW IT WORKS

Once hooked up (unfortunately, I had a prototype cable for connecting Presto that caused me no end of trouble – at press time I received new production cables which seemed to be a huge improvement), the first thing I no-ticed is that Music Publisher launches very slow-

ly. It takes almost 60 seconds to launch this 44lK program from a one-page document off of a hard disk.

Music Publisher assumes that you will be making firm decisions about how you want your pages of music to be formatted before you start entering the music. As soon as the program is up, you are presented with a page setup dialog box if you are starting a new document. Here, a major shortcoming appears. As soon as you de-cide on the size of your music page, set its mar-gins, and click "OK," you are locked in. You can-not change the layout of a page of music without completely trashing that file and starting over from scratch. Thus, if you had an elaborate piece of music formatted at 8½"×ll", and your friend needed it on smaller, narrower pages, you would be completely out of luck.

Next, Music Publisher presents you with a blank page in a single window at WYSIWYG size. Rulers at the top and left have moving lines that indicate the vertical and horizontal coordi-nates of your cursor movement in inches, centimeters, or points and picas. This is a nice touch. Music Publisher makes screen updates, such as when the image must be redrawn as you move symbols about on the page, rather slowly (it is not nearly as slow, however, as Mark of the Unicorn's Professional Composer program). Likewise, using the scroll bars to move around a page is painfully slow. This is why I believe Music Publisher will work most effectively on the faster Mac SE or Mac II with a large-screen full-page display monitor.

By clicking on a vertical location with the mouse, pulling down the Music menu and selecting 'Add A System,' you can select any of

several preformatted systems (combinations of connected musical staves) like Piano/Vocal (three staves), Melody (one), or SATB. Each system can only be used in one of four note sizes, the sizes of the four screen fonts for Rep-ertoire. This is a major goof. One of the main points behind using Postscript is that a Post-script printer can calculate and print any font in any point size imaginable. However, Music Pub-lisher only gives you four Repertoire screen font sizes. This is particularly ludicrous in light of the fact that Music Publisher will let you enter text and lyrics in any point size at all, in half-point in-crements.

The program also has a box available for designing your own custom system con-figurations. It is very flexible, and lets you design any combination of braced, bracketed and barred staves in any clefs you can think of – Alto and Tenor included. When you finish creating your new system, it is automatically added to the list under the Add A System box, and can be called up anytime. This is very, very good. Unfortunately, it is implemented terribly. While you are creating a new system, there is no way to save your work in progress. You can only save it by closing its window and exiting the customization mode. Next – you guessed it – there is no way to go back and re-edit your new system if you need to make changes. I was trying to painstakingly recreate a complicated twenty-stave orchestral system from a symphony when I discovered this – not good.

All of the other music entry functions are ac-cessed by clicking one of sixteen icons on a moveable palette, not unlike the one in MacDraw. ►

Andrew Imbrie: SHORT STORY

The familiar "hand" symbol is used to manually adjust the amount of empty space between staves in a system by simply clicking on an area and dragging to make it larger or smaller. This is effective, although for the sake of reproducible consistency it might have been wiser to have the user specify vertical spaces in point sizes.

ENTERING MUSIC

The most unique thing about Music Publisher is its method of note entry. Score formatting and most editing jobs are handled with the mouse and menus in the usual Mac interface fashion. But when it comes to entering notes and chords into the staves displayed on screen, the mouse is dispensed with altogether and replaced by keyboard commands. To facilitate this, Music Publisher will only work with the special Presto 36-key keypad, supplied. Since the program cannot be used without the special keypad, there is no need for copy protection on this package. Pirated copies of the software won't do anybody any good – and you can still install copies anywhere you want them, on floppy disks, hard disks, and networks – to your heart's content. Bravo.

The user enters the pitch values of notes with the right hand on the extra keypad, which has four rows of keys labeled C through B, and which place those corresponding musical pitches on the staff in various octaves. Durational values and sharps and flats are changed on the conventional Macintosh keyboard with your left. This makes for the fastest and least frustrating method of note entry I have ever tried in a score printing program.

When it comes time to do actual note entry, the "note" icon activates the dual keyboards for note entry. This is the one part of Music Publisher that fully realizes this program's true potential. With a little practice, music entry approaches effortlessness. You move from staff to staff with the Mac keyboard's arrow keys. Normal and percussion notes, rests, and grace notes from double whole notes to 128th notes are all right there under your left hand. Honestly, I don't ever want to go back to any other way of notating music on a Mac.

Of course, the usual Cut, Copy and Paste features can be used to duplicate and move whole blocks of notes.

While we are on the subject of notes, Music Publisher has a very sophisticated algorithm for beaming notes within a measure, which it does automatically as you enter them. When you specify time signatures for each measure, you can instruct the program how to subdivide the beams into groups. For instance, if you are in 7/8, you can tell Music Publisher that you want the eighth notes beamed $3+2+2$, $4+3$, $5+2$, or any other combination you can think of. And you can change the instructions for every measure if you like. I particularly like the way Music Publisher handles partial beaming, such as taking a row of sixteen connected thirty-second notes and creating four subgroups of four notes each while using the top beam to connect all of them. I didn't have time to investigate two

strange Music menu items labeled "Beam Me Up" and "Tie Me Up," but somehow I don't think they relate to either *Star Trek* or bondage games, respectively. (*They're actually for manual beaming and tying if you want to add some additional markings of your own – Ed.*)

About the only drawback to note entry is the fact that Music Publisher automatically places notes to occupy a fixed percentage of horizontal space within a measure based on the note's duration. If you ever need to move some notes closer to or farther from other ones to make, for instance, a brief burst of sixteenth notes in a 4/4 measure easier to read, you have to meticulously insert extra space between them in increments of one and four points. This is

Charles Wuorinen: A RELIQUARY FOR IGOR STRAVINSKY

tedious – simply being able to click on a note and drag it to a new position with the mouse would have been much better.

I'm quite confused as to why Graphic Notes chose to cram all its extensive text functions under a single menu with a number of internal sub-menus, while devoting a whole menu to displaying point sizes for a utility which determines the thickness of lines and boxes that you draw freehand with an arrow tool à la MacDraw. How often are you going to use that? Furthermore, the Font Size sub-menu does not indicate which screen point sizes of a particular font are installed. The standard way of doing this is by displaying those numbers outlined, but Music Publisher simply ignores the issue.

Two other tools on the Toolbox palette deserve special mention, and praise. Slurs and crescendos can be accurately drawn and placed by clicking on the exact points on the page where each will start and end. Most other programs will stick these symbols wherever the heck they feel like it, most often overlapping text or other symbols. Finally, there are icons in the toolbox which let you access features like piano pedal markings, glissandos, and repeat endings. Graphic Notes plans to implement routines for fingering symbols for tabulature and piano music, and guitar fret chord symbols.

All in all, Music Publisher is well suited to typesetting very complex and esoteric types of sheet music, and can perform some pretty advanced notational tricks that we don't have space to cover here.

PRINTING OUT

When it comes to printout, Music Publisher's scores look quite professional on an Apple Laserwriter. Graphic Notes tells me that they feel that the Repertoire font they have created is superior to the industry-standard Sonata font from Adobe, and they could have a point. I did not get the opportunity to try printing out poster-sized scores on a Linotronic, but it should be possible.

FINAL OBSERVATIONS

Currently, Music Publisher does not always let you save your work whenever you want to. The first time you select "Save" from the File menu, the "Save" command becomes dimmed, which means you cannot use it. It only "undims" itself and lets you save again after you enter more notes into the score. Thus, if you make a few formatting changes like moving some systems and adding clef changes, you can't save the document and quit. You have to perform some other operation which Music Publisher will recognize as a significant change before it will let you Save.

The program does not support MIDI or any method of importing or exporting files from other programs such as sequencers or other score printers, though Graphic Notes assures me that they are committed to implementing these features. The next revision, due out by July (which will be free to all registered owners), should support playback of the music

on the screen polytimbrally through separate MIDI channels. I certainly hope so.

CONCLUSIONS

From all of the above, you might get the idea that I don't like Music Publisher. This is not the case. Editing and entering music in this program is a joy, and the flexibility of on-screen text and symbol placement is truly impressive. Plus, the output looks great.

As for problems, the embarrassing one with the cable has apparently already been fixed. It will be harder to implement letting you reformat page size after you have started work on a piece of music. On-screen menu placement and window scrolling speed could be improved. Also, they had better remedy the inconsistent way in which the program saves files immediately.

You'll also notice in the intro that I pointedly did not say that the program could be useful for a home "MIDI" studio. Again, this program does not support MIDI – yet. Being able to read and correctly notate music before you start working with Music Publisher is simply a given.

Music Publisher is a very promising program that, unfortunately, isn't quite up to its advertised potential yet. But for the first offering of a company new to the Macintosh market, they are definitely on the right track.

PRICE $599

MORE FROM Graphic Notes, 200 Seventh Ave., Santa Cruz, CA 95062. Tel: (800) 336-6683.

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LEGAL ISSUES &
THE HI-TECH MUSICIAN *part one*

Copy Protection

Time to set the emotional opinions aside, and discuss just how copy protection works, and why companies do and don't use it. *Text by Stefan B. Lipson.*
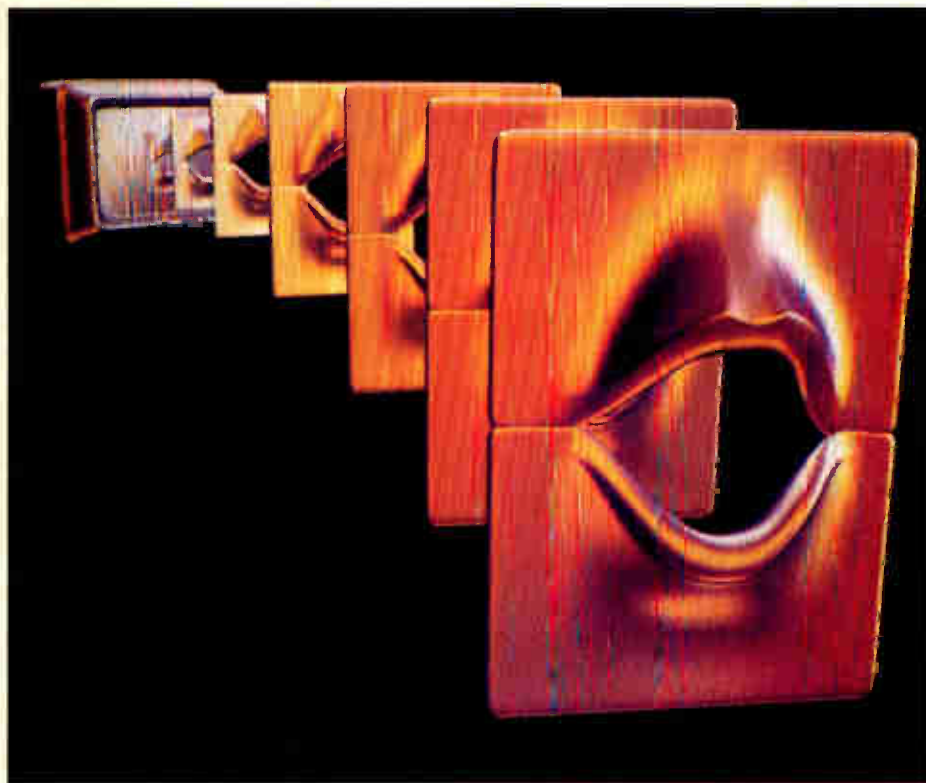
Illustration Colin Cantwell © 1988 Crystal Chip, Inc.

THE POSSIBILITIES BROUGHT about by advances in computer technology are exciting, but along with the technology have come innumerable and unforeseen legal problems. Both designers and end users of computer hardware and software depend on copyright and patent law to protect their work. As it stands, however, US copyright law has not seen a major revision since 1976. As such, there is considerable confusion in the computer industry as well as in the creative arts as to how the law applies to these technological developments. While not all courtroom decisions on computer-related cases have followed a consistent legal philosophy, a large number of the cases do set precedents in some way.

For people in the creative arts, the outcome of these cases is of considerable importance. Music software for sampling, sequencing, composition, and other applications as well as the output of these programs is governed by the same copyright laws previously mentioned. Ultimately, how the legal questions are answered in the courts is of great consequence to artists, musicians, video professionals, and others in the creative arts (not to mention computer professionals).

In this our first of an occasional series, I'll look at copy protection, the software developers' way of preventing illegal diskette duplication.

WHAT AND WHY

Copy protection is a means of preventing a user from duplicating a piece of software for use on more than one machine at a time. Computer programs come with license agreements which are intended to stop (or at least warn) the user that installation of the software on more than one machine for simultaneous use is illegal. Developers felt, however, that written warnings alone were inadequate in addressing the problem – and therefore devised copy protection schemes as a way of preventing illegal duplication.

A number of different copy protection schemes exist, ranging from the storage of key information on an additional undetected diskette track to drilling holes in the diskette to act as a signature. Some copy protection schemes have vulgar side effects, such as spewing expletives across the screen when the user attempts to run a bootlegged version. Some developers have even discussed retaliating against bootleggers by using a copy protection scheme to format and/or destroy various system components such as hard disks. More often than not, however, a copy protection scheme simply will not allow the user to install the program properly, which in turn prevents the program from being used.

From the software developer's point of view, protection is necessary because it stands as one of the few ways to protect their investment. Developers, after spending thousands, hundreds of thousands, or perhaps even millions of dollars on development, must recoup these costs to remain profitable and to generate revenues for further development.

The cost of software piracy is substantial. Hypothetically, if a corporation were to purchase one copy of a program that retails for $200 and copy it onto their 2000 work stations, the developer is deprived of $400,000. Developer's fears of such piracy are not based on the paranoid fears of greedy software executives, either. As evidence, a previous study by The Association of Data Processing Services found that approximately half of the installed copies of Micropro's Wordstar, a word processing program, were illegal.

A CHANGE IN POLICY

For a number of reasons, copy protection has all but vanished in the world of corporate software. One reason for this change in policy is

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that developers did not see the proliferation of hard drives and other hardware peripherals, which have generated a number of additional problems for copy protected software. If a user, for example, copies his or her program to a hard drive and the hard drive fails or crashes, not only does the user lose the hard drive, but the copy protected software is lost as well. In such an instance the user must attempt to secure a new copy from the software manufacturer (hopefully

> *"A number of different copy protection schemes exist, ranging from the storage of key information on an additional undetected diskette track to drilling holes in the diskette to act as a signature."*

for free). Even if the manufacturer is sympathetic, the process of getting a new copy still results in lost time and productivity.

An additional problem surfaces with laptop computers which use 3.5" diskettes but employ an architecture traditionally associated with machines that use 5.25" diskettes, such as IBM and compatible machines. The user of an old IBM XT who purchases a new Toshiba (IBM compatible) Laptop, for instance, is faced with replacing software due to obstacles in copying the 5.25" diskettes to 3.5" diskettes. If the software is protected, there may be no way to transfer the program, short of writing to the software developer.

Another reason for removing copy protection can be traced to stringent policies on illegal duplication that many firms have implemented. These policies have been known to include immediate dismissal for employees caught with illegal software.

In the corporate world at least, the implementation of these policies coupled with the outcry from customers experiencing the aforementioned problems has helped convince developers to remove copy protection schemes.

BEATING PROTECTION

To a great extent, copy protection became irrelevant in 1983 when Central Point Software made available the first of what was to become a wave of "protection bashing" programs. Central Point's star program, Copy II PC, allowed users to duplicate most copy protected programs for IBM PCs.

Every duplication program such as Copy II PC includes a disclaimer on the opening screen reiterating that the reproduction or duplication of software for anything other than archival purposes is expressly forbidden. Developers understandably expressed doubt that such a warning would make any difference to pirates (remember, one of the reasons they used copy protection in the first place is concern that their own warnings would not be heeded). To some, the warning has as much substance as the sticker on cigarette machines warning minors of the criminal penalties for smoking.

Programs which break copy protection are double edged because they address the needs of

both the honest user who wants only to back up diskettes for archival reasons and the pirate who wishes to avoid paying for the product. Meanwhile, affected developers were angered that another developer would create a program that so compromised them. In spite of the developer's sentiments, the program did tremendously well and a variety of such programs subsequently appeared, allowing Apple, Commodore, and Atari owners to duplicate copy protected software. Copy II PC was so well done, in fact, that in 1986 Central Point Software won PC Magazine's Award for Technical Excellence.

THE HOME MARKET

While copy protection for software in the corporate world has almost completely vanished, it is still commonly included with software targeted for the home user. Developers fear piracy by home users because unlike large corporate buyers, they feel that home users cannot guarantee (or coerce) honesty. They feel that these users would not hesitate in copying a program for one or two friends, thereby depriving developers of a significant portion of their revenue stream.

There are other differences between the corporate market and the home market that developers cite as reasons for leaving copy protection on home products. In contrast to the corporate market, for example, there are few disk conversion problems for the home market and until recently, fewer hard drives for home users. Not to be overlooked is the fact that corporations which purchase hundreds of copies of a program or negotiate a site license have considerably more influence with a developer than a home user who purchases a single copy of the program.

ALTERNATIVE PROTECTION

Many users complain that they only want to back up a diskette for archival purposes. Archival duplication is acceptable to developers but it is not archival duplication that they are trying to stop.

Some companies have removed copy protection all together. Others, as a compromise, have adopted a hardware form of copy protection, sometimes referred to as a security block. A security block is a hardware device that fits onto the parallel port of the computer. A special identification code is assigned to the block by the manufacturer. When the program runs, it checks the port to see if the block is in place and if the code is correct. If the block is not in place or the identification code is incorrect, the software will not run.

Security blocks solve the problem of archival diskette duplication by allowing the user to back up the actual software any number of times but preventing the use of multiple copies. This form of protection is costly, however, both to the developer who must pay for the blocks in advance, and to the consumer who ultimately pays the additional hardware cost.

There are a few other alternatives to copy

protection. One such alternative is shareware. A shareware developer encourages dissemination of a shareware program with the request that if the end user likes the program and uses it, then a specified fee should be sent to the developer at the specified address. This appeal to the user's honesty and sense of fair play has resulted in considerable success for a number of such developers.

Another alternative for developers is to provide potential customers with demonstration versions which allow the user to use the system but not save any work. This gives the user a chance to test drive the program without actually getting an illegal copy. Both shareware programs and demonstration programs are often found on networks such as GEnie, Compuserve, or PAN where they are made available to a wide audience.

SUMMARY

Developers feel robbed by illegal program duplication. The complaint that software developers offer sounds similar to that voiced by the recording industry, which tries to prevent duplication of records and tapes. One distinction which developers make, however, is that many developers are independent programmers and their programs may represent many months work with little or no remuneration. Further, most programs cost many times more than a record or tape, and so the duplication of a single program represents a larger loss. Add to this the dilemma of developing for a narrow market which inherently offers a limited return.

Finally, developers express concern for the feasibility of creating new and more powerful products if the revenues needed for development are not forthcoming. Development can be tremendously costly and illegal duplication impedes further development by

> *"To some, the warnings on duplication programs about illegal copying have as much substance as the sticker on cigarette machines warning minors of the criminal penalties for smoking."*

reducing available development funds. Nine months of product development cannot be justified with inadequate financial returns. Ultimately, it is the end user who loses out.

Granted, copy protection has to some extent helped developers reduce their losses due to piracy. Copy protection can be bypassed, however, and it causes an additional set of problems for the end user (and developer) as well. Further, it represents an incomplete solution to the underlying problem of protecting an idea or a creative work that can be electronically copied. Right now there's simply no easy answers. ∎

We're curious as to what you think of the issues raised in this series of articles. The issues do after all, affect you and your livelihood directly

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MICRO REVIEWS

Geodesic MIDISynergy 1.0

A rewritable MIDI sequencer for the Amiga. *Review by Stefan B. Lipson.*

MIDISynergy™ I
Version 1 0

Geodesic Publications

MIDISYNERGY IS PRESENTED as a simple MIDI sequencer/recorder for the Commodore Amiga. There are a few distinctions to be drawn with this product, however. First, MIDISynergy is probably best viewed as a learning tool both for the folks at Geodesic and a learning tool for new programmers interested in building music applications software. The diskette includes all of the C source code for the program (Geodesic wrote Synergy under the Aztec C68 Manx Compiler) so that you can see how various functions are implemented. Because the source code is provided, you have the option to change the program or customize it if you so desire. To do so, you need to know C and you will need a compiler if you want to recompile the code. (The professional system from Manx

is $199. The Developer System is $299 and comes with Unix utilities.)

Geodesic sees itself as providing a jumping off point for people who are interested in doing their own coding. With that philosophy in mind, the program provides an interface that is really just skeletal (Geodesic has included the source code for the interface on the diskette as well if you are interested in seeing how they implemented it), providing access to a basic set of sequencer functions such as play, record, erase, and rewind. There is also a window that allows you to monitor the MIDI data stream in real time, but there are no pull-down menus or other features commonly associated with more powerful sequencers. The documentation consists primarily of installation information, a
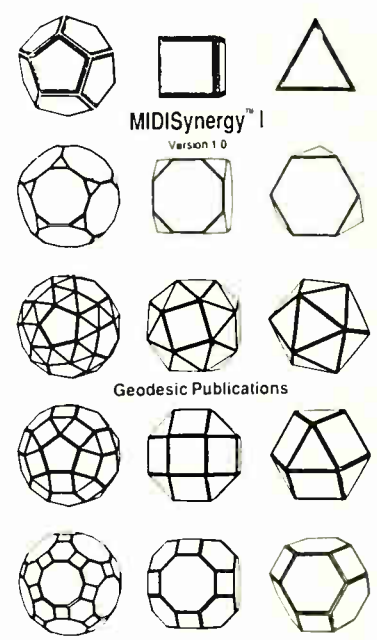
Steinberg's The Ear

Ear-training software for the Atari ST. *Review by Mihai Manoliu.*

COMPUTERS CAN BE an ideal tool for music education: combine endless patience with exact repetition and perfect memory, and you have the ideal teacher. Steinberg's The Ear for the Atari ST goes a long way towards filling the need for an ear training program that is both simple to operate and fairly comprehensive. All you need is a MIDI polyphonic keyboard and about fifteen minutes with the manual, and you will soon be on your way to more accurate hearing.

The program is separated into four sets of exercises: Intervals, Scales, Chords, and Melody Dictations. Exercises consist of the computer playing a certain chord, interval, or scale and the student's response; the correct answer appears once the student has made a response. It is possible to re-listen to the passage before responding, and the computer keeps track of the results of a particular exercise as well as cumulative results. These statistics are displayed at the bottom of each exercise screen.

The Intervals section consists of three separate exercises: one in which you are asked

Desk Intervals Scales **Chords** Dictations Options

Hear chords of four and five notes and inversions !

Which kind of chord ?

Major	Major seventh	Major ninth w/o fifth
Minor	Major w/ raised 7	Major with added sixth
Diminished	Minor w/ dim. 7	Diminished w/ dim. seventh
Augmented	Minor seventh	Major ninth

Which inversion ?

| No inversion | 1. inversion | 2. inversion |
| 3. inversion | 4. inversion | |

| Again, please | Again, as arpeggio |
| Next chord, please | WRONG ! |

Stop it !!!

Attempts: Mistakes: in %: altogether Attempts: Mistakes: 0 in %: 0 This exercise

to decide which of a pair of intervals is larger; the other two ask you to name the interval played (one plays the notes simultaneously, the other consecutively). The Scales section is also made up of three exercises. The first will play one of these five scales: Major, Harmonic Minor, Melodic Minor, Natural Minor, and Hungarian Minor. As in all exercises, roots seem to be

selected at random (the range of possible notes along with the tempo can be set in the Preferences screen; you can thus set the range to be within your keyboard's optimal sound). Another exercise adds the Dorian, Phrygian, Lydian, and Mixolydian modes to the previous five scales. The last scale exercise adds the Whole Tone, and Chromatic scales to all the

brief description of the interface and a specification for a standard MIDI file format. All in all, Synergy is intended for hackers who are interested in the capabilities of the Amiga.

The code is the most interesting thing for prospective customers. After looking at some other files on the diskette, it appears that the folks at Geodesic are interested in experimenting and learning about the Amiga. That would explain the presence of several undocumented programs on the diskette including U7, a primitive graphics program that takes advantage of some of the Amiga's graphics capabilities. There is also a very primitive game called Triclops which features crude animation. There were a few more of these programs available and the source code is also provided so that you can see Geodesic's approach to graphics coding. Granted, the programs aren't relevant to music applications, but if you're interested in hacking with an Amiga, they might be of interest. I have to assume that the folks at Geodesic like playing with the Amiga's graphics as well as the music and thought that others might be interested as well.

Geodesic is interested in what people do with their code and are open to including customer additions with future releases. MIDI Synergy is not copy protected. Hack away. ∎

PRICE $37
MORE FROM Geodesic Publications, P.O. Box 956068, Duluth, GA 30136. Tel: (404) 822-0566.

others. You are asked to decide which one of these eleven scales has just been played.

The Chords section consists of four exercises: triads with and without inversions, and a combination of triads and more complex four- and five-note chords with and without inversions. You can choose to have the chords re-played as arpeggios (which really helps to recognize inversions). Among the more complex chords are minor, major, diminished and dominant sevenths, sixth, major ninth, and the minor-major seventh.

The Melody Dictation section is by far the weakest part of the program. A random atonal melody is played and you are given its starting note. This exercise would be more musical if it was based on random but scalar melodies.

The Ear can make ear training a lot easier and much more fun. However, some improvements could be made: a better Dictation exercise, a computer key for the Again function (or even a way to control Again and Next functions from the MIDI keyboard), and a way for the user to define more scales and chords. Such refinements would make The Ear an excellent companion for all musical levels. ∎

PRICE $99
MORE FROM Steinberg/Jones, 17700 Raymer St., Suite 1001, Northridge, CA 91325. Tel: (818) 993-4091.

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Dr. T's KCS with MPE, PVG, & Copyist
A software environment for the Atari ST

Take a popular sequencer, enhance it and the accompanying scoring program a bit more, add algorithmic composer-like editing tools and wrap it in a multi-program environment, and you're going to be busy for a while. *Review by Lorenz Rychner.*
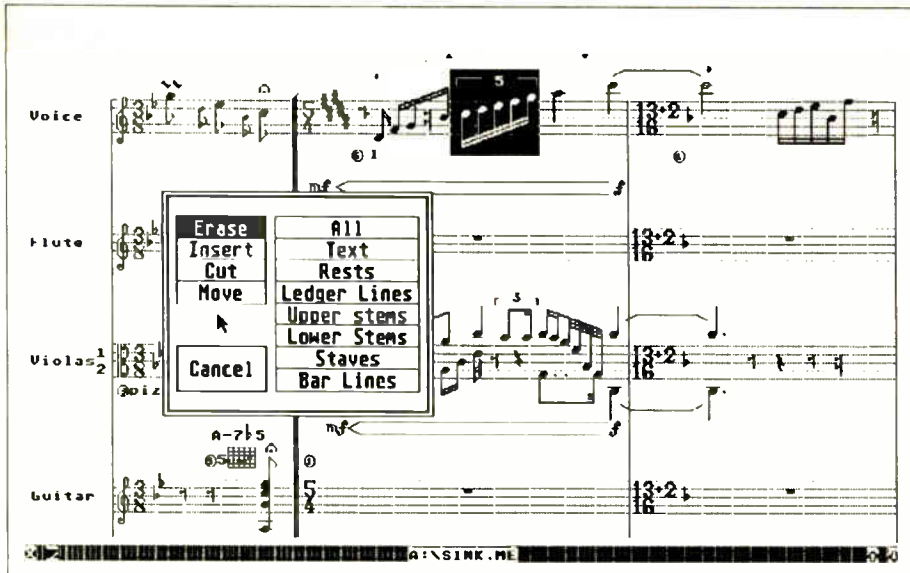
IN THE BEGINNING, Dr. T released KCS (the Keyboard Controlled Sequencer) for the Commodore 64 and Apple II. It featured Open Mode – which worked quite unlike anything that was available at the time. Now, a few years later, KCS works on several more computers as both a normal and unusual sequencer, and has grown several arms: MPE (the Multi-Program Environment), PVG (the Programmable Variations Generator), and Copyist (a transcription program). Time to get the prescription renewed.

AN OVERVIEW OF KCS

The Keyboard Controller Sequencer, of course, has evolved somewhat over the years. It is currently up to version 1.6 *and* Level II, the latter of which has some new sections – namely, the Master Editor and the Programmable Variations Generator, or PVG. There's room here for only a cursory overview of KCS, so I'll give you a little idea of what Open and Track modes are all about before describing the Dr.'s latest wonder drugs. (For a more thorough explanation see the KCS review in MT February '87.)

In track mode you record up to 48 tracks, somewhat like on a multitrack machine. The length of track 1 (you set it, and change it later

at will) limits the length of all other tracks. Each track has its own screen, and all the features that you would expect from an advanced program are there, including Step Time Entry. There's no looping of individual tracks in Track mode, however; they either all loop or they don't. Repetitive phrases have to be in the track as linear data. You play it in once, then append it to itself, which takes up memory.

If your piece calls for a structure other than a short segment played once straight through, you'd probably use Open mode. In this mode, you can record all your music right there into sequences that don't have to be confined to the length of any given other sequence. Anything can be a sequence: a single note (a sample trigger? a crash cymbal?) or a whole bunch of merged tracks, a control sequence that lists all the other sequences, including transpositions of pitch and velocity, repeats, and last-minute data like volume (controller #7) for a perfect fade-out of a whole merged song – you name it.

THE MASTER EDITOR

This collection of magic elixirs is only available from Level II. It brings up six screens, and only the currently selected one will affect the data in the selected sequence.

The first screen is called *Blend*; it lets you make changes to a track or sequence while the computer reads the data of another sequence (the reference sequence) as the model. This can be done by aligning pitches to those in the reference sequence, or velocity values, durations, rhythms, channels, or you can autocorrect some events to match exactly the times of events in the reference sequence. Of course, the model doesn't have to be copied exactly for all of these parameters, and the computer will just approximate them if you say so. The manual suggests this as a way of producing swing feel, and many other uses come to mind.

Chords is the next screen. If you played some chords in a slightly arpeggiated way (and who doesn't?) and want them straightened out, then the 'Deflam' feature is the tonic for you. 'Arpeggiation' goes the other way – it spreads out a chord over time. The 'Sort' function could be seen as a preparation for Arpeggiation – it rearranges the notes in a chord in the order of their pitches, either ascending or descending, and with a skip (skip every other note first and play the skipped ones later) if desired. Velocity and duration can be also changed in several ways, and the 'Orchestration' feature makes it possible for the notes to be assigned to a number of MIDI channels.

Controllers is a screen where MIDI events other than notes can be treated. After you define the controller, select 'Split' if you want that controller's data to be extracted and moved to the next available empty sequence; select 'Erase' to simply delete it; pick 'Thin' if you need to thin out the density of controller data. As an example of using Thin, say you've used more aftertouch than necessary, and the sequence is flooded with AT events. Experiment with different Fraction Kept values (2 strips half the events, 3 leaves only one third, etc), and set a Maximum Time to ensure that events close to note events will be left in (one of the most considerate Thinning functions available overall).

Track Utilities offers changes to all tracks: 'Deleting' a range (shortens all tracks), 'Erasing' a range (leaves a blank, like rests), 'Insert' adds a ▶

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range of rests. Let's just look at an example for the latter: say you need to extend a verse by one bar where you'll later add a solo drum fill. You set a Range Start and an Insert Amount in clocks, and all tracks will play nothing for a measure at that point. Back in Track mode you'll record the fill, give its first note the time value of the downbeat of that measure (there's a calculator back on KCS's Edit screen), and you're done.

Pitch Map is the next screen. If your sampler had the crocodile bark assigned to the note D3 when you recorded it, and you want to change it to the snake hiss that is triggered from B5,

MPE "By clicking on the Ext function within KCS you can simultaneously load up to four of Dr. T's Editor/Librarian programs without the need to shut down the sequencer."

maybe with a change in velocity, you're in business. You can even strip an individual note out of the whole sequence by giving it the command 'Channel / Delete 0.' A pitch mapping screen lets you reassign pitches over the range of 7 octaves and 3 semitones. Just be sure to realize that Dr. T chose to go against the MIDI standard with the note octaves versus MIDI Note Numbers – in all his programs, C4 is MIDI note number 60 (middle C), as opposed to C3 in MIDI (he's in powerful company, since Korg, Roland, Kawai, Digidesign and others do the same).

Oh, and finally, tempo commands can be erased or changed up and down in percentages, and accelerando/decelerando values can be converted into absolute tempo values.

TWO PLACES AT ONCE

"MPE" stands for "Multi Program Environment." On the track and sequence Edit screens it shows as Ext (for – you guessed it – external). Click on Ext – without the need to shut down the sequencer program – and you can load up to four of Dr. T's Editor/Librarian programs. The scoring/notation program The Copyist is about to be updated for MPE. While in the editors, you have access to SysEx and to the disk drive of your choice (as defined on the System Screen). When you select To KCS, you return to the last KCS Edit.

Of course, nothing's free – loading these programs into memory results in a reduction of available free events. The Copyist, in particular, will take up a lot of space. In fact, unless you have a Mega ST (either 2 or 4), the good Doctor points out that you won't have very much room for files if you load the Copyist and editor programs simultaneously.

The great part about this integration is that you can listen to the current sequence play the patch you're working on in the editor program. Plus, the time and aggravation that this arrangement saves alone are well worth the trouble of getting the editor programs in the new MPE versions.

It's not readily apparent from inside KCS 1.6 that you can also unload the programs – there's no message to this effect, but there is one in

KCS Level II: shift-click on the last program loaded to remove from memory. This restores all the memory, and worked flawlessly from the KCS 1.6, once I had gotten wise to it. It didn't work quite as smoothly with the KCS Level II and the Copyist, because the latter was a pre-release version that hadn't been cleaned up yet.

SEQUENCES TO NOTATION

The Copyist has been reviewed in these pages before (see review MT June 1987). To teach a computer how to deal with the many variables that experienced arrangers and copyists deal with routinely must be an awesome task; therefore, I always have some slight sympathies for their shortcomings. As with all notation/conversion programs I've seen, you'll soon learn how to edit any rhythmically interesting sequences just so that the computer can display the music in a readable manner (there seems to be a rule that if it looks right then it'll sound dumb, and if it sounds right then it'll look dumb). There's a playback mode from the notation screen, which is nice, but gets hurt by the aforementioned rhythmic side effect of applying this balm.

To give a capsule review and summary of The Copyist, if your music is destined to be read and played live and you have your copying chops together, you'll probably win over the computer hands down (and studio players dislike computer printouts passionately, anyway). If it's for shopping your songs around, go for it – the result can look very professional, and the link with the sequencer worked like a charm.

THE FINE PRINT

The new programs come with owner's manuals that are a far cry from the tiny books with miniature print that Dr. T's distributed before. They are very readable, ringbound in sturdy plastic, with indices and table of contents. The Copyist's manual left me stranded a lot, but the others made learning a breeze. The disks are copy-protected, by the way. Hard disk installation is possible, but the original disk must be in drive A during the boot-up, serving as a key disk.

THE PVG

This Escher-like space is also only available from within the KCS Level II. The basic idea behind PVG is to take some of your existing sequencer material and create variations on it which you control by setting values for a variety of parameters. Looking at it from a broader point of view, it's the most powerful editor you're ever likely to see. Downright intimidating, actually – particularly when you come up against screens that look like they're test papers in math, with terms like Set Values, Rotate Values, Gaussian, Macros, Consecutive Mults, and In-betweens. And the manual waxes

on about "Chaos making a new Science" and Mandelbrot diagrams (not in my dictionary . . .). We're getting *out there*, folks. And that's just where this program will take your music if you let it. But you're in control at all times, although it'll take you a long time to get a sense of how much control. Give it as little or as much music to work on, define the direction in which to take that music, set up restrictions, and see what comes up.

What you'll be building to operate on your phrase is a set of definitions called a Preset. The program can store up to 80 Presets. For more complex procedures, you combine up to 16 Presets into one of 20 Macros, so that the computer can combine the commands of the Presets in one operation.

The first screen in PVG is called *Changes*. A number of data fields are ready for entering parameter values that will randomly change pitch, velocity, duration, time, shift, and interval, based on the data on the Edit screen when you clicked on PVG. If you had a range (block) defined at that time, then that's the material the PVG will work on. It could be as little as a single event, or a whole song. You must further tell the program how many variations it should perform, and what the likelihood of varying one parameter over another should be. I hear you say "how should I know?" Well, take a stab in the dark. Fill in some numbers under AMT (amount) and WGT (weight), set a number for Changes per Vary, and see what gives.

For an experiment I used the default C scale from sequence 1 in the Open Mode, which moves by quarter notes from C4 up to C5. In the field 'Change by Constant' I entered Pitch Amt 2, Wgt 1, with Changes per Vary at 1, asking for 10 variations with evolving Mults, not overwriting the Original. Under 'Restrictions,' where you delimit things that the program should *not* create, I entered the pitches of A, A#, C#, D#, E, F#, G#, with pitch limits of C4 and C5 for upper and lower limit (in other words, keep to the range of the original). The result: the Program wrote 10 new tracks, each with 8 notes picked from the C scale, at quarter note intervals (since I didn't ask for a time change), all between C4 and C5. They read like this: CCEFGABC, CCEFGGBC, CCDFGGBC, CCDFFGBC, CCDGFGBC, CCEGFGBC, CC DGFGBC, CCCGFGBC, CDCGFGBC, and CDCFFGBC. The manual states that "the WGT value determines the likelihood that a change will be made by that particular amount." Low weighting, small changes.

Instead of Change by Constant, I could have chosen Gaussian, where the Amt column is replaced with SD (standard deviation, which the manual defines as "a statistical measure of the likelihood that a given change will be of some specified size"). It seems to mean that whatever you enter there won't always come up, but more often than not it will. And, I could have entered some numbers under 'Time' and 'Shift,' which would have affected the rhythm and length of the scale. Selecting 'Staccato' or 'Legato' would have played with the duration of the notes that would have been changed

rhythmically. 'Signed' values is the third option next to Constant or Gaussian, giving you more control than the others, since the changes seem to happen more gradually, and in the positive or negative direction that you can input.

Another screen lets you assign *Swap/Copy* functions, where the source material isn't altered, just rearranged. Again you enter numbers, and you set up a protection scheme. Set Values will set a randomly chosen event to the set value irrespective of its starting value. Interval, Delete, and Erase work here, too.

Global Changes allow you to make changes to the unprotected notes of a sequence, not necessarily at random. This is a screen where you can make Edits in a predetermined way, by giving only the global change any weight (meaning no other parameters, even on other screens), and by setting the Changes per Vary to zero. This works for transpositions or inversions of pitch, velocity, duration, time reversal, delete, or erase options. Split/Pattern looks for patterns that you input (a certain interval or other relationship) and splits it off into a new track. By careful setting of the parameters and of the protection options you can split off pretty much what you want as opposed to random values.

In-Betweens looks at two existing sequences and creates new material according to some mathematical formula, while taking into account any Autocorrect values (for similarity in rhythm) and Scale Positions Restrictions that you have input. Another screen produces *Ornaments* (embellishment figures), including notes that weren't part of the source sequence. You can set the timing of the Ornament with Offset and Delay values. Other parameters contribute to the amount of time the Ornament gets played, or split the Ornament off onto another track.

PVG *"The basic idea behind PVG is to take some of your existing sequencer material and create variations on it which you control by setting values for a variety of parameters."*

Add Controllers is a way to introduce random controller data, not unlike the Ornaments which are random note data. You choose from Delay, Type, and Value; consider Looping the result, or mixing the effect of several controllers at once. A fixed Program change can be added, and the introduction of new controllers can be stopped with Next Note Lim, Extend, & Duration. Controller events can be further modulated according to parameters like the pitch or velocity of notes. If you want to feel that you at least started the process, *Vary Controllers* needs some controller data to be present in the original in order to do its job. You specify which controller data should be varied, with restriction and protection parameters. These screens are interactive, and care must be taken to clear data before setting up another effect.

CONCLUSIONS

I have by no means touched on every parameter: there are just too many. And given the interaction between screens, with the variety of effects that can come from them, you'll win the lottery long before you come close to exhausting the possibilities that the PVG offers. With every change in the source sequence the outcome can change, even if you have kept the input values to a minimum and if you use just one Preset. Can you imagine what could happen if you wrote a Macro with 16 Presets, and lots of Changes per Vary in each? I didn't have the time to do it, because inputting the right values (and keeping track of them) is quite time consuming.

How far you'd want to take KCS is largely a matter of taste. I doubt that musicians who are plugged into the realities of the commercial music scene will take the PVG portion of the program very far, while the excellent sequencer lends itself very well to real-life projects. But for the explorers out there, this must be paradise. The ability to jump to other programs with MPE just adds to the versatility. Cancel your vacations . . .   ∎

PRICE KCS I.6, $249 (registered owners can upgrade from I.5 for $25, if I.5 was bought after 9/1/87, or for $40 if the I.5 was bought before 9/1/87); KCS Level II, $325 (registered owners of I.5 can upgrade for $100, registered owners of I.6 can upgrade for $75); The Copyist costs $249 and will soon be available with MPE; updating editor/librarians to MPE level costs $25 if bought after 9/1/87, $40 if bought before 9/1/87.
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perspectives

SEQUENCERS, MEET THE HUMANS

In our new regular feature for alternative points of view, our Technical Editor decries the narrow-sightedness of current sequencer design. *Opinions by Chris Meyer.*

YES, YES, I'LL agree – sequencers these days are truly amazing, wonderful things. They're immensely powerful, very flexible, and work as great songwriting tools.

They're also narrow-minded and unaware of how to work with humans.

For some (maybe even the majority), composition and music is a premeditated thing. But for the rest of us, a good deal of composition comes out of just playing around at the keyboard (or on the guitar, or wherever). Also, some of us prefer jamming with a few friends over learning a few songs to play back by rote (and some good ideas may come out of those jam sessions, for playing back by rote later). Therefore, I'd like to see MIDI sequencers start to accommodate those of us who want to just jam.

The top of both my Christmas and birthday lists is what I refer to as a "free time" sequencer. This would be something I could flick on in record mode (akin to a tape deck) and just play into (notes *and* program changes, as I get inspired by various sounds). And I want to do this *without* plotting ahead of time what time signature or tempo I'll be playing in. After jamming for awhile, I'd want to be able to go back, mark the parts I want to keep around for editing, and throw the rest away. Then, when I isolate a fragment inside one of my rough edits that I like, I want to be able to mark downbeats (either by isolating the notes on the screen, or tapping them in while playing it back), and say "Okay – keep those four bars, and that was in 5/4." The sequencer then replaces the old bar markers (which were probably something weird like three beats and 27 clocks per measure) with real ones where they belong, and calculates the tempo. Now, I can edit and quantize the fragment I played as if I had premeditated the tempo and time signature, and overdub against a metronome click that's now in time with what I played.

The above can be applied to recording a MIDI'd band jamming. To do this, a sequencer would need to take in multiple MIDI inputs (why hasn't Southworth's JamBox/4 been copied more?) and be able to record on multiple tracks (or at least multiple MIDI channels). It would be easier to place the downbeats if the sequencer could read a particular MIDI note on a particular channel (say, the equivalent of the drum or snare) and automatically map those as half notes (or whatever). Then, just edit in the missing or edit out the redundant beats to make it all fall right. I had hoped that something like this could be done with a drummer driving a Human Clock or Master Beat, but they're always a few beats behind a drummer's changes, and therefore the autocorrect goes wacky (quantizing behind or ahead of the beat) during those bars.

My next dream relates to jamming as a form of playing. Imagine that a few members of your MIDI jam band have devices that require clocking, and can sync to MIDI. However, maybe not everybody wants to have their devices

> "I'd like to see MIDI sequencers start to accommodate those of us who want to just jam."

running from the start of the jam – either they decide to switch from keyboards to arpeggiator 20 minutes into the set, or they have to go out to disk to load a new sequence fragment that they want to play along with. There has to be a way to constantly tell everyone on the MIDI sync line where the downbeat falls. Older Sequential Circuits devices used to send out a special message that marked where it was.

Something that can be done generically is to again specify a note on a particular MIDI channel (say, MIDI note 0 on channel 16) that marked the 1. Then, either one player is designated as the master clock and builds this note into all his/her sequences, or some ultracheap sequencer is set up as the tempo master under some player's control for tempo and time signature (the latter of which could be changed on the fly by running in

drum machine mode, and cueing up different one bar patterns of different lengths to loop over and over). Then (wishfully), the slaved devices could be told "Okay – start playing this on the next downbeat signal you see." Some sequencers almost allow this to be done within themselves, but they tend to expect the player to start alternate patterns precisely (instead of cueing up for a bar marker) or to premeditate how things will be chained before the sequence even starts.

Of course, one would want the ability to transpose. Most sequencers allow this; again, it would be nice to designate a master channel, and the master player could transpose everybody (or that could also be built into a master guide track for the master to allow chord progressions to be followed). Additional jam features would be giving the player/controller some ability to adjust dynamics and timing while playing. For the former, just the ability to expand or compress velocity levels per track in real time would help a person orchestrate during a jam. For timing, the ability to slide tracks slightly ahead or behind the main beat in real time would let people play with the feel. Both of these would be particularly nice to have under a hardware slider control while playing (as opposed to a mouse and ASCII keyboard); perhaps MIDI controller numbers and channel could be specified ahead of time so these parameters could then be altered from a master keyboard or something like the Yamaha MCS2. (The recently announced Feel Factory from Future Lab may also fit the bill here.) The last simple trick would be a half/double time switch, for moving effortlessly into half or double time.

So, whatcha think – isn't it about time machines started dealing with us on *our* own terms? ∎

A Good Idea Got Better
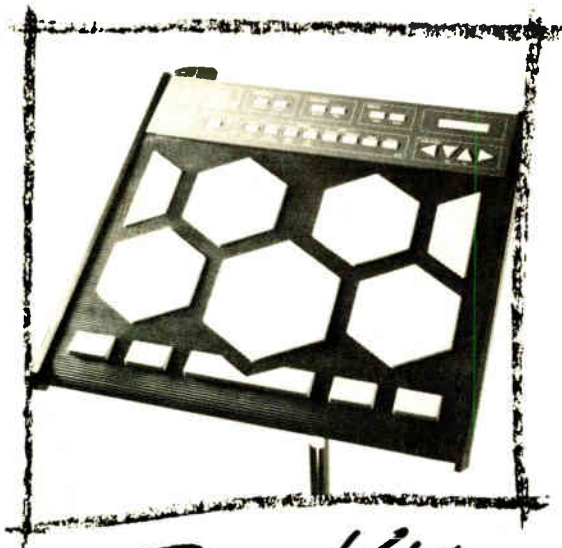
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SEQUENTIAL PROPHET VS
ArabTech
Yung Dragen, Santa Monica, CA

This is a refinement of a sound developed by the VS's random patch generation function. Yung (normally, our Readers' Tapes reviewer) named it this "because the random name the VS gave it looked like Arabic to me." It's a slow attacking atmospheric patch where the amplifier, then filter, and finally waveform mix slowly take their turns bubbling to the front. Yung describes it as "a colored chrome organic view of an alien planetscape." A special (after) touch is pressure affecting both chorus rate and depth, "which is like pouring ammonia into the atmosphere." Yes, it is tonal, and no, we won't inflict him on you again. ■

VOICE CONTROLS

—	42	100

KEYBOARD MODE SINGLE

CHORUS

ON	ON	16	44

LFO GROUP

	Shape	Rate
	TRI	82
	TRI	16

VOICE PAN

ARPEGGIATOR

MODULATION

Source	Amount	Freq	Filter								
	0									YES	
	0		A-C B-D								
	51										RATE + DEPTH
			B+B			YES					
			A-C B-D								
						YES					

OSCILLATOR GROUP

Oscillator	Wave #	Freq
	32	00.08
	74	11.97
	37	00.00
	62	00.10

MIX ENVELOPE

Point	Rate	Level
0	—	5/37/52/6
1	76	19/67/11/3
2	55	38/16/4/32
3	63	46/28/10/9
4	54	83/0/0/17
5	76	—

Loop	Repeat
1→3	1

FILTER ENVELOPE

	Rate	Level
	—	0
	53	61
	43	29
	55	44
	42	23
	63	—

Loop	Repeat
2↔3	CONT

FILTER GROUP

Cutoff	Resonance	Env Amount
20	11	

AMP ENVELOPE

	—	

ENSONIQ ESQI
BlVerb
Craig Thomas, San Francisco, CA

Craig describes this interesting patch, which is short for Bell Reverb, as a "new age sound with a distinctively Asian character." It has a somewhat percussive attack which expands into a windbell-like sound with a slow release. The "reverb" portion of the patch comes from the ambience created by this slow envelope. Craig has given the patch a slightly off-center panning which you may want to tweak to taste. Happy contemplating. ■

	OCT=	SEMI=	FINE	WAVE	MOD #1	DEP #1	MOD #2	DEPTH
OSC 1	+2	00	00	PRIME	LFO 1	-01	ENV 1	+12
OSC 2	+0	00	05	EL PNO	LFO 1	-03	ENV 1	+12
OSC 3	-1	07	06	BELL	OFF	+00	OFF	+00

	LEVEL=	OUTPUT=		MOD #1	DEPTH	MOD #2	DEPTH
DCA 1	63	On Off		OFF	00	OFF	+00
DCA 2	63	On Off		OFF	00	OFF	+00
DCA 3	63	On Off		ENV 1	44	OFF	+00

	FREQ=	Q=	KEYBD=	MOD #1	DEPTH	MOD #2	DEPTH
Filter	38	00	32	ENV 3	63	ENV	20

	Final Vol (ENV 4)	PAN=	PAN MODULATOR	DEPTH
DCA 4	63	12	LFO 2	63

	FREQ=	RESET=	HUMAN=	WAV=	L1=	DELAY=	L2=	MOD=
LFO 1	16	On Off	On Off	TRI	10	03	00	WHEEL
LFO 2	14	On Off	On Off	TRI	00	31	63	OFF
LFO 3	—	On Off	On Off	—	—	—	—	—

	L1=	L2=	L3=	LV=	T1V=	T1=	T2=	T3=	T4=	TK=
ENV 1	+63	+00	+00	63	00	00	06	00	00	09
ENV 2	+63	+00	+00	00	00	00	00	00	00	00
ENV 3	+63	+32	+03	44	20	00	40	48	32	15
ENV 4	+63	+63	+00	13	38	32	40	63	48	09

	SYNC=	AM=	MONO	GLIDE=	VC=	ENV=	OSC=	CYC=
Modes	On Off	On Off	On Off		On Off	On Off	On Off	On Off

	Split/Layer=	Split/Layer Program	Layer=	Layer Program	Split=	Split Program	Split Key=
Split/ Layer	On Off		On Off		Off Lower Upper		

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PATCH
WARE

REVIEW: If you're looking for new sound offerings for your D50, you should check out the new **Alternative Soundscapes, Vol. I-III** from **Day Star Enterprises**, which contain some very good additional sounds. As in most offerings of this nature (despite the programmer's best efforts it seems), there are some redundant sounds, or simply sounds individuals won't find useful. It's probably more like a ". . . you can't please all the people all of the time . . ." situation, because none of the sounds are of poor or unusable quality. It's just that everyone will find a few which naturally fall into daily or at least routine use, quite a few that are used occasionally and then there are the 10 or so that never seem to find a place on any track.

Volume I contains the basic all-round selection of sounds, leaning more towards pianos, basses, brass and woodwinds. The expected D50 percussion patches are here as well as some nice pads and leads.

Volume II was the one I personally liked the best and the one which I think really shows off the strengths of L/A synthesis. The pads were unusual enough that they added a different texture to a track, but not so wonky that they weren't useful. A lot of patches were built for chase mode, and take advantage of a variety of reverb selections, not just the huge Hall patch. There are also more percussion sounds, brass and analog patches, and although everything is undeniably D50 programming, there are enough unique sounds to set this cartridge far above the other two.

Volume III is geared more toward acoustic imitation sounds, and they don't come across as well. It's not that they're bad, and if you're looking to collect more D50 sounds this volume is certainly worthwhile, but to my ears (and maybe it's due to listening and playing 128 previous patches) they just didn't have the same creative magic of the second set.

The packaging is quite nice, with a sound chart of every patch on thick, plastic coated card stock – making for quick reference. Sounds can be purchased in a variety of formats; from RAM cartridges provided by DayStar (at $145 per) to easy-to-use data sheets ($15). If you supply your own RAM card, DayStar will load the sounds on them for $15 a piece, or at $20 they'll load them on either Yamaha's MDFI disk, Roland's MC500 or a variety of Macintosh librarian programs. Application notes are also included for each Volume, giving the user valuable hints about how to get the best response from the available sounds.

Overall, the Alternative Soundscapes series of D50 patches are very good, and the pricing structure makes them affordable for just about anyone. Although I can't say any new ground has been broken here, you'll certainly find a few gems to enhance your tracks. For more info contact: *Day Star Enterprises, PO Box 2387, Amarillo, TX 79105.* ■ *Chris Many.*

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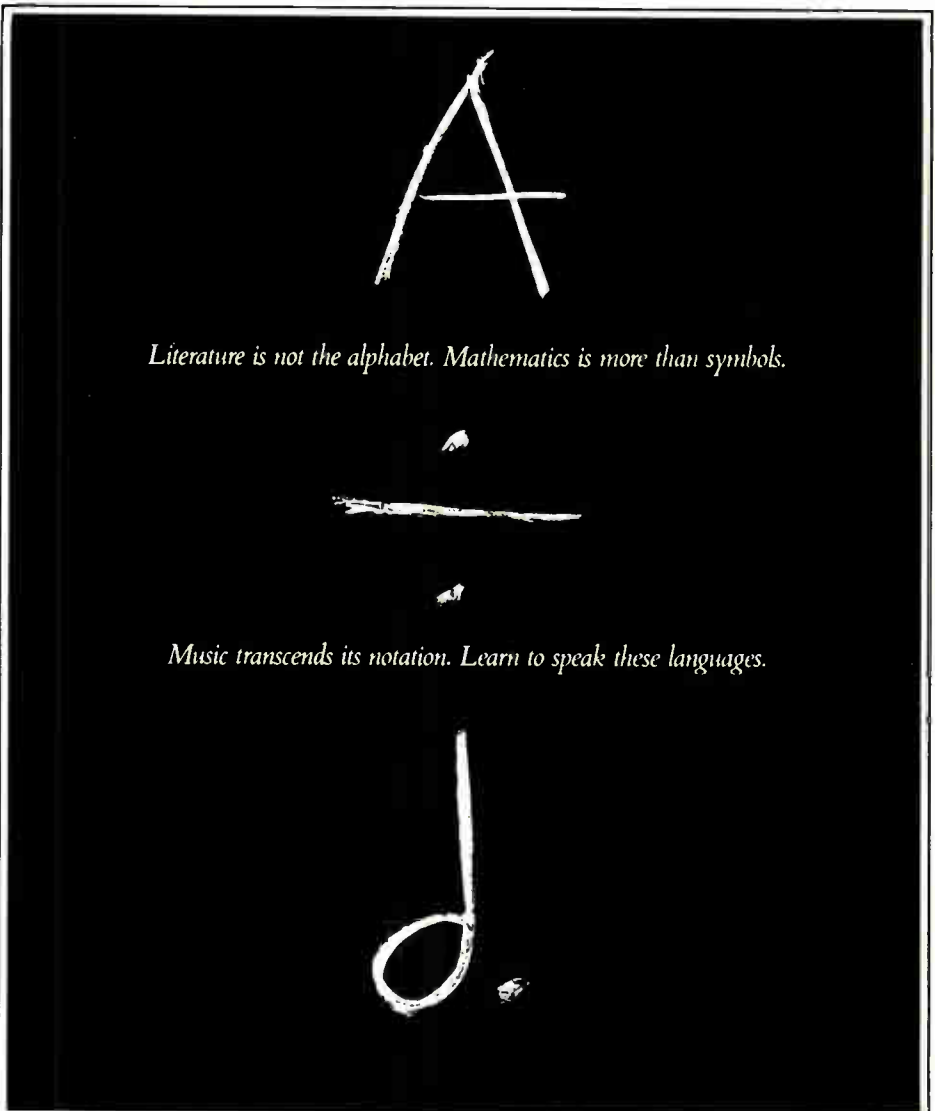
Begin with Basics   Foundation for the Advancement of Education in Music

L ISTEN TO HIS solo with Santana on 'Soul Sacrifice' from Woodstock. You can hear that Michael Shrieve has always been a drummer who played around the edges, whose rhythmic pulse was suffused with melodic and harmonic invention, at least as much as you could get on an acoustic drum kit.

"When I look at that Woodstock solo now, which is not very frequently I must say," he adds quickly, "but every now and then it comes out on cable and I check myself out and see myself playing these Max Roach licks, which rock drummers weren't doing. Also I was playing with a guitar player like Carlos Santana who was so melodic that it always created a desire in me to think melodically on the instrument."

Shrieve had to wait several years before electronic drums, sampling and MIDI put the same tonal palette at his sticks that keyboardists had had at their fingertips for years. In 1988, Shrieve is ready to establish the state-of-the-beat for the 1990s.

Shrieve is sitting in The Site recording studio, just up the road from George Lucas' Skywalker Ranch in Marin County. Across the control room glass, his partner, percussionist David Beal, is doing the final mix on their electronic percussion ensemble album, *The Big Picture*. Another record, with synthesist Steve Roach, was released in June. He's readying plans for a jazz album with Patrick O'Hearn, Andy Summers, David Torn, and Mark Isham and preparing for a Santana 20-year anniversary reunion tour.

Shrieve's career has the logic of a Walter Payton downfield run, full of missteps, spins, sprints and stutters with occasional drops for losses. If you've been wondering whatever happened to Michael Shrieve after Woodstock, some updating is necessary. "I always go through this," says Shrieve exhaling a stream of cigarette smoke that serves as a resigned sigh. "People are always saying, 'so what have you been doing since Woodstock.' I was just telling my wife the other day that we could've been living in Hawaii for ten years for all people know."

In 1969, Michael Shrieve was one of the hot drummers on the block. His name was often spoken in the same breath as Ginger Baker, Mitch Mitchell and Keith Moon, musicians who expanded the concept of rock drums beyond simple timekeeping. He joined Santana while they were recording their debut album, right after the original drummer walked out. "I was still living with my folks and I was about 17," he laughs.

After seven years with Santana, Shrieve struck out on his own, working on projects that ranged from adventurously obscure to commercially forgettable. At the later end was Automatic Man, a hi-tech funk group that recorded two records for Island in the late seventies. "The idea was like Power Station," explains Shrieve. "It had R&B

the STATE *of the* BEAT

From the basic latin back-beat of Santana through the complex world of sequencers and samplers and back again, drummer/percussionist Michael Shrieve marks out a distinctive beat.
Interview by John Diliberto.

underneath and rock 'n' roll on top. It didn't work." Neither did his other rock ventures, Novo Combo or HSAS with Sammy Hagar and Neil Schon.

But while he was with Automatic Man he worked with Island label mate Stomu Yamashta, an iconoclastic avant-garde classical percussionist, lately turned synthesist. In the late seventies Yamashta put together Go!, a multi-media project with vocalist Steve Winwood, guitarist Al DiMeola and synthesist Klaus Schulze. They recorded three albums, *Go!*, *Go Live!* and *Go Too!*, that careened from space jams to Winwood's soulful singing and DiMeola's speed demon guitar.

Out of Automatic Man and Go! came two events that would shape much of Shrieve's music for the next 12 years. On the debut Automatic Man album, he recorded for the first time with electronic drums. "In '76, I was looking into electronic percussion and invested money into the first electronic drum company called Impact," says Shrieve. "In fact, I used those on an Automatic Man album in '76 and that was one of the first recordings of electronic drums. Then I got out of it."

From Go!, he began a long-term relationship with German synthesizer legend, Klaus Schulze. Schulze already had a formidable, albeit underground reputation as a pioneer in space music. His synthesizer opuses like *Timewind*, *Moondawn* and *Picture Music* are considered precursors of the New Age.

"I was really loving Klaus' music except for the drums," says Shrieve sardonically. "I said, 'Listen, I really like your music, but I can't listen to it because of the drums. I'm a drummer, and every time I put it on and

the drums come in, I have to take it off. Nothing against the guy (Harold Grosskopf), but let me come over and do some things so I can listen to the records.' " Shrieve laughs at his own audacity.

Schulze, himself a former drummer, admits that it was Shrieve who got him back into percussion. "I had stopped the drums because I couldn't hear them anymore," says Schulze. "But then I met Michael Shrieve and he taught me how to use the rhythm in pieces like my own music without destroying the mood or the feeling."

Shrieve and Schulze recorded many sessions that wound up as albums, including *Trancefer* and *Audentity* (Brain Records). Shrieve discovered that he could play freely within Schulze's modal maze of interlocking sequencer rhythms and spiraling synthesizer refrains. "He was using sequencers, and sequencers are tight and precise and I can play that way," proclaims Shrieve. "I had played a lot of latin music where there were a lot of rhythms going on, so it was the same thing as a sequencer, because you could stop playing or go into it. It was in and out of the groove, sort of like skipping rope. You come in and you're either messing up or you're not. You're floating or you're not."

Shrieve floated on his own album, *Transfer Station Blue*, one of the most popular albums on the new age Fortuna record label. The music was derived from sessions he'd recorded with Schulze, including pieces from the Wahnfried album, *Tonwelle*. Shrieve had taken the 16-track tapes, re-mixed them and in the case of the title track, interpolated R&B rhythm breaks

that were as misplaced as stick figure drawings on a Rothko canvas.

The record made an unusual crossover from the new age into funk audiences, with DJs using it as rap background. It also found an audience with film directors. "Every film I've ever gotten has been because of *Transfer Station Blue*," says Shrieve. "Directors love that."

SINCE HE DRIFTED apart from Schulze, Shrieve had moved to New York and began experimenting heavily with electronic percussion and doing session work, including a bit with The Rolling Stones and then on Mick Jagger's 'She's The Boss.' It proved to be a sort of psychological epiphany for Shrieve. "I went to Nassau for ten days with Mick and Jeff Beck," he recalls. "We were staying at Chris Blackwell's house, and Mick was playing me songs on the acoustic guitar and they didn't sound that great. And I realized as he was playing them for me that they were going to end up sounding great. This is Mick Jagger playing and he's written classics. And it just hit me that I should start believing in myself, writing-wise, and just write and not doubt what I don't know, but have confidence in what I do know."

Shrieve began working again with electronic percussion, which had taken quantum leaps since the Synare. With digital samplers and Simmons drums, he now had the melodic abilities he once envied in Carlos Santana. "Now it's gotten to the point where we can actually play melodies with the drums," he says with relief. "But lo and behold, now what do you do with this technology and what do you do with the ability to play melodies on drum sticks? Well you have to play melodies or play music with it. So it comes back to the same thing. It has to be good music."

Shrieve embraced drum machines and in 1984 recorded *In Suspect Terrain* (Relativity Records), an all-percussion solo record centered largely around the Linn drum machine with other acoustic and electronic percussion added in. "There's an art to programming drum machines," explains Shrieve. "You can make it sound like a human. I did a record called *In Suspect* ▶

Terrain where I tried to do that. The only problem with that record was that it came out four years later in America. By that time there were new chips for the Linn, so it sounded, perhaps, old. But I still feel there was some very hip programming in the sense that I programmed one piece, 'Ratatouille,' that had the influence of Tony Williams, and people like that. It's an exercise that I wanted to put on record."

But it sounded like an exercise – brilliant

rhythm tracks in search of a context. The melodies would come as he began scoring films. In 1986, he and synthesist Patrick Gleeson scored Dino De Laurentis' movie, *The Bedroom Window*. For the sessions, Shrieve brought in a young drummer named David Beal. "Just before leaving New York for California to do the film, I did a small session with David Beal, kind of a new guy on the scene," says Shrieve admiringly. "He came to the session with all these samples. He was one of the first guys in New York to have it so together with that scene. So I hired him to work with me on the percussion for this soundtrack. We had such a good time doing it, I said, 'Let's create a situation where we can do something percussion wise.' It felt time for me to get back into something that was percussion music. I've always wanted to do something that was percussion music besides songs or playing on a piece of music, but actually percussion music."

If Michael Shrieve was still a hippie, David Beal would be his worst nightmare, a post-punk apostle, replete with spiked black hair, black clothes and black pants. During the interview, his girlfriend, Catherine Rush, sat on the studio floor sewing metal studs into another pair of black pants.

When Michael Shrieve was flailing the drums at Woodstock, David Beal was beating on pots and pans. He was five years old. But in these cross-cultural days, they've found a common ground. Although Beal is a session player, he has performed extensively in jazz and symphonic music before making the leap to rock 'n' roll. He studied percussion at the Cleveland

Institute of Music, and played in the Baltimore Symphony, Cleveland Philharmonic, North Texas State Jazz Band, in late incarnations of Blood Sweat and Tears and more recently with Joe Cocker and Peter Gabriel.

"David is the perfect modern man," exclaims Shrieve, while Beal sits in the control room putting the finishing touches on their album. "He really has the drum stuff down, plus the computer and sampling stuff more than anybody I've ever met. And we're completely different. He's a rocker and I guess I'm not."

Beal is a musician who enjoys his role as a pop music mercenary, playing music to order. Yet he's able to open up to something different and experimental. "It was a new experience for me because coming from a rock 'n' roll standpoint, I'd always had to work with a song and the song had to fit radio programming formats," enthuses Beal, through a hoarse, high-pitched voice. "It had to be a certain length and have a hook, verse and chorus. When we did the film we were able to do seven minutes of music that didn't fit into anybody else's category. It was probably one of the most expressive things I'd done."

Together, they call themselves Big Picture, and their album, *The Big Picture* (Fortuna), is a percussion duet full of digital samples and acoustic drums. They create a global percussion dance ensemble that shifts from African drum choirs with strings to avant-garde percussion ensemble with a groove.

"Everything is played on sticks," claims Shrieve, even if the sticks are triggering strings, brass, basses and every percussion device from a bass drum to a wok. "Once you get your hands on these pads and you get sounds up that you like, you don't want to stop playing them. You can start off a piece of music and just keep going on and on. On something like the Octapads you have a limited amount of notes, so everything sounds good but you have to be careful about it getting boring."

Using a pair of Roland Octapads each

with DW footpedals made by Drum Workshop and an MSB Plus for MIDI patching, Shrieve and Beal triggered an Emulator III, Emulator II, Roland S50, a Mirage, an Ensoniq ESQ1 and Roland drum machines. Although almost everything was played, most of the basic tracks were sequenced on Macintosh computers with Performer software as well as Intelligent Music's M, UpBeat, and Donnie Blank's Drum File. "On most of the record we would sequence a foundation and usually we went back and replaced them track by track," says Beal. "You can spend hours and hours programming a sequencer to play like a human, but we're players – so why bother? The sequencer was mainly an arrangement writing tool. That makes the record sound more exciting to me. In the pop world they like everything sequenced, but this is the one world where you can get away with a good sloppy fun groove. So why not do it?"

Despite all the computer and electronic technology, the album has a primal, organic feel, like it just danced out of an African jungle. "One thing sampling has done is turned you on to world acoustic instruments," says Shrieve. "When we use tablas in the music or African drums, granted we try not to pretend that we're Indian tabla drummers; we play as we always play, but use those instruments' sounds. And that has got to have its own uniqueness."

"That was the most exciting thing about electronics," agrees Beal enthusiastically. "As a percussionist you're only as good as your collection of percussion, so a guy from rock 'n' roll who could express himself in ethnic music, like African music, his rhythms could never cross over because he couldn't play instruments like talking drums and bata drums. They take years to learn how to play. But once you get into sampling, you can take that bata or talking drum, sample it and put it on your Octapads and all the different ethnic communities of drummers have now crossed over.

"In *The Bedroom Window*," he continues, "we had African tribes, with 20 or 30 African drums that were all sampled and we could just play track by track and create things that drummers like Michael and I could never express on our own."

It also changes the way they had to approach the drums, taking into consideration properties of sound with which drummers aren't usually concerned, like sustain and pitch-bend. "With the authentic percussion there are so many subtleties on the drums, and subtleties in the way people play with sticks," explains Beal. "The sound you're playing, like a talking drum, may have a pitch sweep to it, so when you're playing those rhythms you're taking that pitch sweep into consideration. You want to leave space for

it. Before, the rhythm you might play on a cymbal bell, that could never be played on an orchestral bass drum, because of the physical aspects. Now, that cymbal bell can be an orchestral bass drum so you're getting a whole different rhythm sense."

SHRIEVE'S CAREER HAS been like a complex rug weaving, with patterns re-emerging and threads leaving off and picking up. But one of the most interesting threads is Steve Roach, one of the few American synthesists to create his own sound out of the German, Klaus Schulzian school of electronics. Roach is an admitted Schulze disciple and when you walk into his studio, the cover of Schulze's *Timewind* album occupies a prominent overview of the room. So perhaps it's only natural that he and Shrieve would connect.

They initially played together at The First Annual Palo Alto Space Music Festival in May of 1987 and immediately found an affinity in each other's music. Later that year, Roach flew Shrieve in from New York to lay tracks on his *Dreamtime Return* album. Those tracks were never used, but Shrieve wound up staying for six weeks, not only recording tracks for *Dreamtime Return*, but laying down the basis for an album of Shrieve-Roach duets called *The Leaving Time* (RCA-Novus).

For Roach, it strengthened his spiritual connection with Klaus Schulze. For Shrieve, it picked up another dropped thread. They locked in synchronization together, writing most of the music in that six-week period, jamming on the sequencers. "We'd be programming at the same time on the drum machine and the sequencer and just building and changing and shaping in an intuitive way," says Roach fondly. "Michael was also using the Ensoniq sequencer (with the ESQ1) so we had that common language between us. And the drum machine programming, of course, there's no question about his involvement there."

But both decided it needed something different, so they enlisted electro-jazz guitarist David Torn. "It felt like it needed more than just the synths to me," says Shrieve. "I had done that with Klaus and I was beginning to feel like we had to open it up."

Roach felt the same way, and David Torn's distended guitar appears on every track, adding harmonies far outside of the new age milieu normally associated with Roach. "His knowledge of harmonizing melodies and chords is another element that took the album to another level of harmonic sophistication," agrees Roach. "It was amazing hearing these chords and melody lines that were on the border of being way out there, tonality wise. At times it took my ears awhile to adjust to this new sound."

The Leaving Time is a new sort of space-fusion, resonating with synthesizer textures,

ambiences and the synchro-mesh sequencer patterns of Roach, pushed by the rhythmic drive of Shrieve, and shaped and bent by Torn's liquid guitar leads.

But after all this electronic innovation, Shrieve is looking again at his acoustic drums. Even on *The Big Picture*, he and Beal, after laying the electronic tracks, went back in with double acoustic drum kits.

Now he's working on a solo project that he calls a jazz record, using a traditional jazz drum kit and employing David Torn, Patrick O'Hearn, Andy Summers and Mark Isham, doing tunes by Gil Evans and others. "It's not going to be like one of those new age records," says Shrieve, with visions of fuzak rearing its head. "I want it to be new music and I love what these guys are doing, so obviously there's going to be an ambience to it. But I want to play drums on the record and I want to play freely on it as well. So I name those names and a certain thing comes to mind and that will be there. I want it to be improvisational drum-wise, and not sequenced or programmed.

"I feel like I've come full circle now," says Shrieve with an air of affirmation. "There was a period there in New York where drums weren't enough. Maybe I was hanging around with the wrong people, but I was really bored with the drums, and now I'm really back into them. The direction I'm going in is more of a jazz direction, because it has more to do with the art of drumming than anything else."

Since Shrieve has been known to shift gears at a moment's notice, it remains to be seen how the project will turn out. And a new wrinkle could be the Santana 20-year Reunion that will take place later this year. The only sure thing is that Shrieve will follow his own instincts and that it will all, somehow, be interconnected. "I learned from Stomu that if you do interesting things, interesting people find it," he says philosophically. "You don't have to worry about doing something that everybody might like. Do what you feel and the people who like that will come towards it and they will be interesting people." ∎

MT AUGUST 1988

I·N·D·E·X

to features from Inaugural issue to July 1988

TRYING TO REMEMBER what issue the D50 review was in? Need to do some research on MIDI Time Code? Well, if you've ever needed or wanted to know what we've covered in MT over the last two years, here it is: the definitive, all encompassing index. We hope you find it to be a good reference source; it certainly proved to be an eye-opener for us.

Our thanks go to intern Susan Leach, who spent many long hours inputting and revising the following list. Without her help it probably would've taken another year to get it out. ∎

MUSIC

INTERVIEWS

REVIEWS

COMPUTER NOTES

MT AUGUST 1988

T E C H N O L O G Y

I N D E X

On Circuit
EMMETT CHAPMAN
Le Café, Sherman Oaks, California

This month's pick performer, also the designer of The Stick and The Grid, shows off what he can do with his own creations.
Report by Chris Meyer.
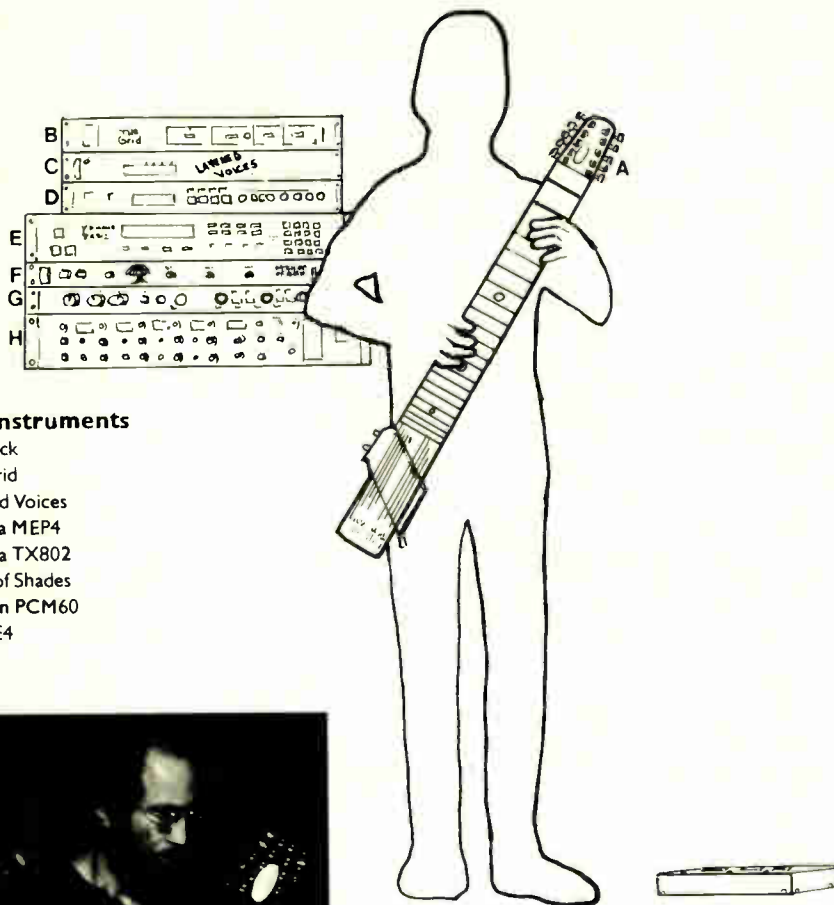
Key to Instruments

A The Stick
B The Grid
C Layered Voices
D Yamaha MEP4
E Yamaha TX802
F Patch of Shades
G Lexicon PCM60
H Toa DE4

THE EVOLUTION OF the guitar has been a slow one. Most of you probably didn't know it used to have just four strings; the addition of a fifth a few centuries ago was heralded as the perfection of the instrument . . . For the past couple of decades, Emmett Chapman has experimented with adding even more strings, eventually settling on the ten-stringed Stick – a fretboard instrument with five bass and five treble strings meant to be tapped, as opposed to plucked, à la Stanley Jordan.

About a year ago Emmett took an IVL pitch-to-MIDI converter intended for the pedal steel guitar and experimented with fitting it to a Stick. After going "hyper" (all strings the same light gauge and tuning to improve response and tracking) and some custom software to match up to the Stick's dynamics, the Grid was born – a synthesizer controller which Chapman likens to "10 tiers of keyboards." Aside from normal guitar-like tricks, the IVL system also allows different tunings to be called up from footpedals (since all the strings on The Grid are the same, tuning can become arbitrary).

Chapman also designs signal processors, attempting to show sensitivity to the needs of a guitar-style player. His "Patch of Shades" is a pressure-sensitive footpedal that can be used as a volume pedal, an effects return level controller and a sort of wah-wah pedal in its own right. His latest creation is a MIDI processor called "Layered Voices." What this box is about is taking in a couple of channels of information and then allowing the user to switch in and out various parallel lines under footswitch control to layer up the performance. A simple concept, but well suited for its players.
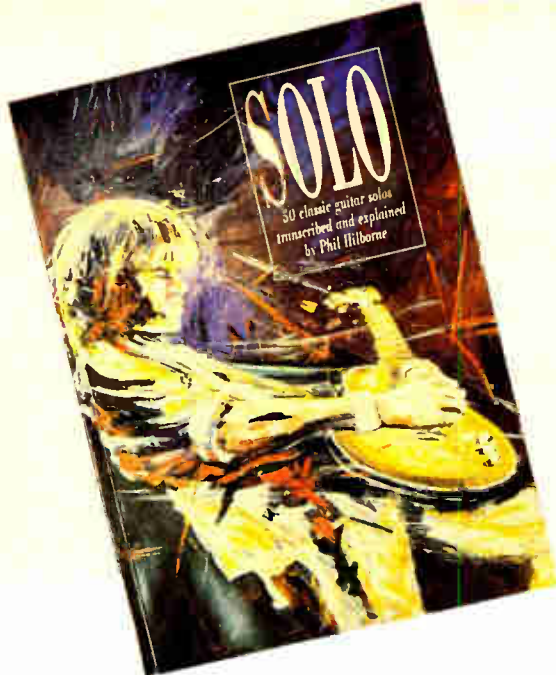
For the Le Cafe gig, sounds were provided by a Yamaha TX802. Percussive FM voices seem well suited for the basically percussive style of the Stick and Grid. A PCM60 provided basic reverb ambience and echo; a phase shifter behind the rack was applied to the Stick's bass for some motion. The mixer was a simple Toa D4 that occasionally flickered its clipping lights, but no clipping was heard (aside from some woofer flap out of the Toa 30SD on the first number). Le Cafe is very small and only required a couple of speakers and Walter Woods 350 watt amplifiers (selected for their durability and headroom). Unfortunately, a nasty buzz crept into the Stick's pickups that was never tracked down (and tentatively blamed on the light dimmers).

The concert ended up being a very friendly, personable sort of affair, with Emmett switching between Grid and Stick every other number, and explaining to the audience what was going on between numbers. Unfortunately, he had to reach back to the TX802 to change patches during the tunes on the Grid, which hurt some of the flow. The mostly improvised performance ended up being a cross between easy listening, jazz and mutated bluegrass, often devolving into blues. The Patch of Shades and Layered Voices performed their jobs subtly and effectively. Emmett's technique (sliding and bending mixed in with the hammer-ons) was also a subdued piece of showmanship. I didn't always agree with the selection of patches on the TX for the Grid (such as plinky treble versus strong bass), but the phrasings left me musing that a keyboardist wouldn't think of splitting notes between the left and right hands as a Stick player would.

It's fun to see an inventor get a chance to stretch out in person, and explain why he's doing this in the first place. ∎

Casio PG380
Guitar Synth

Interested in guitar controllers but not in the mess of cables and converters? Casio's latest may be the answer: a synth built into a normal-playing guitar. *Review by Steve Shepard.*

SINCE THE INTRODUCTION of the commercial keyboard synthesizer in the early 1970's, guitar and keyboard technologies have evolved in very different directions. The electronic keyboard family has grown from an amplified mechanical keyboard and electronic organ to include analog and digital synthesizers, and the current breed of hybrid and sampling instruments. The evolution of the electric guitar, on the other hand, hasn't been nearly as dramatic. The state of the art electric guitar in the late 1980's is basically a refinement of the

old Fender Stratocaster, with modest improvements in electronics, materials, and machining. A wide variety of effects processors are also available for the guitar, including the new generation of digital multiple effects processors.

Essentially, the keyboard has evolved into a family of sophisticated electronic instruments, while the electric guitar has not, and there has been little or no attempt to integrate more sophisticated electronics into the instrument itself. Surprisingly, the advent of the MIDI standard didn't do much to close the gap between guitar and

keyboard technologies either. However, it did allow the guitar to serve as a MIDI controller for a sound module or external synthesizer. To date, guitar MIDI controllers have been plagued with tracking problems, due largely to the inherent slowness of pitch-to-voltage conversion at low frequencies (remember, at least two cycles of a pitch must typically be sampled before that pitch can be unmistakably identified).

The Casio PG380 Guitar Synth is the first (with all due respect to the Vox Guitorgan) serious attempt to go beyond the guitar MIDI controller, and into the unique realm of the self-contained guitar synthesizer. It is part of Casio's ongoing battle to gain some respect in the marketplace, and is closely linked to the new VZI keyboard synthesizer. The onboard synth allows a guitarist to simply plug in and play, without the usual rack-mount array or pedal matrix. This really opens up the concept of the guitar synthesizer for the working guitarist; ie. one who might play consecutive rock, show, or solo gigs on the same instrument, as opposed to a studio musician who can (sometimes) take the time to get just the right sound.

Luthier Talk

THE GUITAR ITSELF is a Strat copy with two single coil pickups and a bridge pickup that can be switched from humbucking to single coil operation with a push/pull adjustment of the guitar volume control. The maple fingerboard and ebony neck are well crafted, and the guitar is quite easy to play. The overall quality of the instrument is much higher than its predecessor (the Casio MG510 Synth Controller). A Floyd Rose tremolo allows deep bends without loss of intonation. This tremolo has the tradeoff that to change strings it requires an Allen wrench (included) and a pair of wire cutters (not included, but necessary for cutting the ball end of the strings before seating them in the bridge). It is also, in my opinion, somewhat disappointing that Casio didn't follow Ibanez's lead and allow the tremolo arm to change the pitch (or whatever else was desired) digitally.

Another aspect of the PG380 that may take some getting used to is the close proximity of the volume controls and hexaphonic synth trigger pickup to the bridge. The trigger pickup is right next to

the bridge, and the guitar volume control is just ahead of the trigger. If you are used to playing close to the bridge things get very crowded, because there is really no room to rest your hand. However, these are relatively minor (and highly subjective) points, as overall the PG380 is an excellent guitar.

A built-in tuner stays in operation whenever the synth electronics are switched on (these don't have to be on for use in normal guitar mode). The chromatic tuner consists of two LEDs which indicate sharp or flat strings. When both LEDs are on, the string is in tune. If you are used to a tuner with a meter and needle indicator,

number which indicates the relative trigger sensitivity when a given string is plucked. According to the manual, numbers below 85 indicate that sensitivity should be increased, while numbers above 99 register as "oL" (over level), indicating that sensitivity should be decreased. The adjustments are made by turning one of six set screws with a Phillips head screwdriver. The process is pretty straightforward, and you probably won't want to change this once you've found the settings that match your technique.

The aforementioned set of DIP microswitches allow the user to set the MIDI bend range, channel, and mode

Sounds *"The quality of the preset voices is quite good. Several of the piano and organ voices are outstanding, as are the flute, vibes, marimba, and steel drum sounds."*

you may be somewhat wary of tuning with only two LEDs. However, I was able to tune quickly and accurately, and once it is in tune, the guitar stays there for a long time (my review model came out of the shipping carton in perfect tune).

Harboring Fugitive Electronics

THE BACKSIDE OF the guitar is partitioned to hold compartments for batteries, microswitches, trigger pickup sensitivity controls, and the ROM/RAM card slot. The electronics are powered by six 1.5V AA batteries or an optional AC adaptor (the standard Radio Shack 9V negative tip adaptor works just fine). The battery compartment is covered by a plastic plate that is held in place by two screws. These screws can be manipulated with a coin, but for a quick battery change, the arrangement is somewhat awkward. Also, the hinged plastic cover of the ROM/RAM card slot tended to pop open from time to time on the unit I tested.

A set of batteries kept the setup powered up for approximately 12 hours, with no problem. When the batteries begin to weaken, the LED display will flash for 15 minutes and then the power automatically cuts off (an analog battery condition indicator would be nice, to prevent the guitarist from being caught in the middle of a set of music with dead or dying batteries).

The pickup sensitivity controls and microswitches are easily accessed by removing rubber covers from the compartments. The sensitivity controls allow the synth trigger pickup to be adjusted for each string. A sensitivity check mode is initiated by turning the power on while simultaneously depressing the octave up and down keys on the front panel. In this mode, the front panel LEDs display a

(poly/mono), as well as the exact pitch of the "A" note of the built-in tuner (four settings are available, from 440-443Hz). There are eight bend range settings, ranging from 2-48 semitones. With the "chromatic" front panel setting enabled, the bend range is automatically set to zero.

Two standard ¼" output jacks are available on the PG380, which allow four possible combinations of guitar and synth sounds. The iPD/mix output yields a monophonic mix of the straight guitar and synth sounds, with the mix determined by the guitar and synth volume controls. The guitar output yields either a straight, mono guitar sound, or both the guitar and synth simultaneously (if a stereo plug and cable are used). By using both output jacks, separate mono iPD and guitar output signals are obtained. I found the guitar output to be significantly louder than the iPD/mix output, but was able to get a satisfactory mix using only one standard guitar cord.

The 64 preset synthesizer voices available on the PG380 are selected by entering the number of the desired voice onto an eight button keyboard. The current voice number is displayed on a two-digit LED readout which is mounted at an angle, allowing a clear view from the player's perspective. The data keyboard arrangement takes some getting used to if you want to change voices quickly. One inherent problem with a self-contained guitar synthesizer is that a guitarist doesn't usually have a free hand to make adjustments with, the way a keyboard player might, which is why many previous systems have relied on foot pedals for control. After some practice, I was able to make voice changes reasonably quickly. Additional pushbuttons allow voices to be shifted up or down one octave, and also select smooth or quantized treatment of string bends. There is also a front panel ►

pushbutton to select optional RAM/ROM cards if any are installed (none were available in time for this review).

iPD Synthesis

YOU MAY RECALL that after the introduction of the DX7 and the widespread acceptance of FM synthesis, Casio introduced phase distortion synthesis with the CZ family. The 64 preset voices are generated by a new technology Casio calls *interactive phase distortion* (iPD), which is incorporated in the new VZI keyboard synthesizer.

With the iPD technology, it appears that Casio is trying to provide the best of both worlds. Eight modules, each consisting of a digitally-controlled oscillator (DCO) and amplifier (DCA), generate independent waveforms which can be used to modulate output from other modules, or to hook directly to the outputs. These modules are set up in four pairs called "internal lines." The waveform pair generated in each line may either be mixed or used to modulate one another for ring or phase modulation. An "external phase" modulation mode allows the output from one line to modulate one of the modules in another line.

The importance of understanding any of this depends on whether you plan on buying a VZI along with your PG380 so that you can program your own sounds on the VZI, and then transfer them to the guitar via RAM cards – they fit in a compartment on the back of the instrument. Without a VZI, you have 64 preset voices and virtually no capacity to alter them.

Fortunately, the quality of the preset voices is quite good. Several of the piano and organ voices are outstanding, as are the flute, vibes, marimba, and steel drum sounds. Some of the sounds (the basses in particular) are more interesting when the octave keys are used to shift them up or down an octave out of their usual range. A few of the piano sounds tend to glitch during fast passages, and there are several novelty sounds (explosion, ambulance) which will be of little use to most players. These are not serious problems, but if you have 64 voices and no means of altering them, you would like them to all be winners.

Tracking and Slaves

THE REAL TEST of any guitar controller or synthesizer is its tracking. This is one area in which the Casio is quite surprising. To put it simply, it tracks better than anything else I've tried. The reason for this isn't exactly clear. Casio's promotional literature claims that the PG380 will "track its own internal sounds with no MIDI delay." This is undoubtedly true, but MIDI delay is not the source of the tracking problems that have plagued just about every

controller on the market to date. The tracking problems are a by-product of pitch-to-voltage conversion, and Casio has managed to keep them to a minimum. The tracking on the low E and A strings, although a bit slower than that of the higher strings, is quite good when compared to other units on the market. It has become a more or less standard trick for guitarists using pitch-to-voltage MIDI controllers to replace their low E and A strings with lighter gauge strings tuned up an octave to improve tracking. This won't be necessary with the PG380.

To use the PG380 as a MIDI controller, all that is needed is a MIDI cable and an external synth or sound module. The onboard synth continues to operate when the PG380 is used as a controller, so it's quite simple to get an extremely rich, layered sound by combining the guitar and onboard synth sounds with the external sound source under MIDI control. The unit does track its internal sounds somewhat faster than it tracks an external sound source, supporting Casio's MIDI delay claim, but it's quite impressive either way.

The PG380 may be used in either Poly or Mono modes. In Poly mode, all six strings control the same MIDI voice, while in Mono mode, each string can be set to control a different source. A DIP switch on the back of the guitar is used to set the desired mode.

Program changes are sent by entering the appropriate number with the pushbuttons on the guitar. Unfortunately, the appropriate number is not the actual program change number. Because the guitar keypad has only eight pushbuttons, an alternate numbering system must be

used to access all of the numbers from 0-63. Thus to send MIDI program change 3, the number 14 would be entered on the keypad and displayed, or to send 13, 26 is entered. Got that? Furthermore, program change numbers 64-127 can only be sent if an optional ROM card is installed.

Living with it Live

I HAD THE opportunity to "road test" the instrument on a number of gigs, and the results were definitely positive. It takes a while to get used to having a dead-on Hammond B3 or Rhodes sound at your fingertips, or playing convincing classical flute duets with a flute player, but many of the 64 preset sounds are quite impressive. The synth tracking is very good and the majority of the onboard sounds are glitch free.

One thing that did strike me as odd is the lack of sustain on any of the voices. If

you hold a note, which is a natural thing to do when you have an excellent organ or cello sound, it is quite disconcerting to have the sound cut off abruptly after 1.5 seconds or so. The problem is that when the amplitude of the string falls below a certain threshold the synth's circuitry reacts as if it received a note off command. Without some kind of external sustain pedal there seems to be no way around this problem because the threshold is not adjustable.

The lack of sustain will also pose a problem for rock players, as the guitar itself is a fairly "hot" Strat copy, set up for rock playing, yet the synth provides no sounds that sustain as well as the guitar. Hopefully, Casio will address this problem in future updates or accessories.

Twang for the Buck

THE COST OF the PG380 is deceptive. Its list price is in the neighborhood of $1500, which might sound shocking at first. However, you don't have to buy anything else to use it, so there are no hidden costs. Compare this to some of the other MIDI controllers on the market, which require that you buy separate trigger pickup and pitch-to-MIDI converter units at a price comparable to the PG380 (not to mention the price of the sound source you wish to control, and a guitar to mount it all to), and also consider the fact that the PG guitar is an excellent Strat copy, and you see that the Casio is actually a pretty good buy. It's easy for guitarists not to see what all of this would cost them with other systems (and going hi-tech guitar isn't cheap, no matter how you look at it).

Guitar synth "The PG380 is the first serious attempt to go beyond the guitar MIDI controller, and into the unique realm of the self-contained guitar synthesizer."

To summarize, the Casio PG380 is a unique instrument that affords guitarists the opportunity to create a wide variety of synthesized sounds without the use of outboard accessories. As a self-contained synthesizer it is versatile and easy to use. It is by no means an ideal instrument, but it is the first of its kind, and Casio is to be commended for introducing this innovative new product. The PG380 is certain to generate a great deal of controversy and competition in the marketplace. ∎

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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

ANALOG 2600, Odyssey, Micromoog, Axxe, Solus, MicroComposer, Synare II, Rhodes, Polysix, Maestro, more. Sale/trade. Complete list, Richard, Tel: (206) 364-7881.

CASIO CZ101 programmable synth, brand new w/box and receipt, bought but never used. List $500, sell $270. Tel: (714) 985-4307.

KORG EX800 synth module, $225. Will ship COD. B.T., Tel: (904) 576-1893.

MOOG MEMORYMOOG synth, w/ MIDI and built-in sequencer. Owner's manual, schematic, foot pedal included. Mint cond, $900. Tel: (919) 489-8430.

MOOG MINIMOOG, fine tune, accessories, spare parts. Moog Liberation, case all new, mint cond, best offer. Tel: (914) 725-3023.

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

ROLAND JUPITER 8, classic analog, mint cond, anvil f/case, MIDI interface, cables, $2000 obo. Peter, Tel: (713) 790-0738 (earlier is better).

ROLAND MT32 with Sony MDR-40 headphones, $400. Listen 2.0 software, $500. Dave, Tel: (805) 964-7724, X385, days.

SCI PROPHET T-8 MIDI synth, 76 keys with polyphonic aftertouch, $1100. Akai S612 sampler, disk drive, disks, $350. Tel: (205) 356-8987.

SCI SIXTRAK, interface, sequencing, patch editing software for CBM4, $500 obo. Tel: (918) 847-2480.

YAMAHA CP70 electric grand, $1200 obo. Trade for Ensoniq Piano, Prophet T-8, Yamaha KX88/76, any controller. Tel: (718) 966-5829.

YAMAHA CS70M polysynth, mint cond, w/road case, $750 obo. Tel: (818) 888-6846.

YAMAHA DX21, exc cond, home use only, $470. Roland TR505 drum machine, $190. COD accepted. Tel: (916) 923-9387.

YAMAHA KX5, Yamaha FB01, and Casio RZ1. Complete performance setup! $875 obo, trades negotiable. Jason, Tel: (516) 242-3352.

YAMAHA TX816 rack, 8 DX7s, studio use only, excellent cond, $2700. Leslie Mizell, Tel: (803) 254-4463.

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AKAI X7000, $625, perfect! Kawai K3, $599. Korg 707, $599. Boxes, manuals, I pay shipping. Tel: (314) 741 7894.

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YAMAHA QX1 version 7, disk drive, 80,000 note, home used, 50 disks

included, mint cond, good sequencer, $995. Tel: (615) 824-6391.

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KORG DDM110 SuperDrums, 9 sounds, 32 programmable patterns, good cond, $100. Tel: (918) 847-2480.

LINNDRUM w/MIDI, owner's manual, factory patch cassette, perfect cond, $450. John, Tel: (302) 654-1805.

LINNDRUM w/MIDI, expanded memory, mint cond, $800. Tel: (601) 452-4593.

OBERHEIM DX, timing pedals, accessible tuners, $395. SCI TOM, reverse sounds, MIDI, $375. Synare II w/sequencer, $475. Trades. Richard, Tel: (206) 364-7881.

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APPLE IIE CLONE, complete, Roland MPU401, Passport/Roland sequencing software, more, $695. Macintosh: Performer, Drumtraks, Jam Factory, more. Richard, Tel: (206) 364-7881.

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PERSONNEL

GIFTED MUSICIAN seeks producers with ear for music for professional opinion. Contact: D Ewing, 8251 Roselawn, Detroit, Michigan 48204.

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MISC

DS0 ENTHUSIAST seeks same to trade patches and exchange information and tips. Write: Bruce Pelley, Box 444, 50 Lancaster, Mass 01561.

IBANEZ FRETLESS BASS, Casio CZ1, Fostex 350, Boss Micro FX, Roland Spirit 40A guitar amp, others. Write: F Chirife, 145 4th Ave #5C, NYC, NY 10003.

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MIRAGE RACK w/ filter and VES for ST, $750. TX7 and DX-Android, $450. ADA 2FX multi-effects, $150. Tel: (501) 394- 6027.

PERFORMER 1.22, paid $360, asking $180. Dokorder reel-to-reel 4-tr, excellent cond, $375. 20×4×2 console, needs work, $375. Tel: (201) 572-5917.

SOUND WORKSHOP 1280B, Calzone case, Yamaha CX5M computer, 64 Fender Precision Bass, EMG pickups, will sell cheap. Tel: (303) 925-5660.

SOUNDCRAFT 200BSEQ mixer, 4mths old, excellent cond, $5900. Lexicon PCM60, $450. Contact: Bruce Siekmann, RD#1, Box 167-B, Mertztown, PA 19539.

STUDIO EQUIPMENT for sale. Send name and address for info sheet to: Hughes, 496 La Guardia Pl, Apt 169, NYC, NY 10012.

WANTED

ARP 2500 modular synth: documentation, parts, information and help are requested. Michael Simmons, 211 Lambert Palo Alto, CA 94306. Tel: (415) 322-1661.

BIZARRE and unusual demo-tapes for review in new apocalyptic art magazine: Caffeine Magazine, 150 Haight St, Apt #105, San Francisco, CA 94102.

STOLEN: EMAX, #1233, and Korg DW8000, #2868. Reward for information leading to their return. Tel: (303) 755-9633 or (303) 977-4328.

TEAC 3340S tape deck manual. Ray Rideout, Sound Creation, 900 John Nolen Dr, Suite 130B, Madison, WI 53713.

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CLASSIFIED

IPS-33 Smart Shift

Intelligent harmonizer

The DigiTech Smart Shift represents a breakthrough in digital pitch shifters. It's an intelligent harmonizer that can add one or two extra notes to each one you play. Notes that are clear, pure and perfectly synchronized to as many as 41 different harmonies. And the IPS-33 Smart Shift is priced far less than harmonizers of comparable quality.

Smart Shift is different from most other pitch shifters for two reasons. First, the pitches it generates are crystal clear and free from distortion. In fact, they're clean enough to be played without accompaniment.

Second, it offers intelligent, user-defined harmonies that synchronize to 41 different scales. These scales include chromatic, natural minors, harmonic minors and pentatonics, all the way to Dorian, Phrygian and Mixolydian.

The IPS-33 Smart Shift is actually a 16-bit real time sampler and a computer rolled into one. The powerful technology it contains allows you to program harmonies by either defining the key and scale type to be used, then selecting how far above and/or below your note you want the others to play; or by assigning up to two different notes for every note you play, from A to G#.

Regardless of which programming method you select, Smart Shift is a simple way to recreate the classic pop guitar harmony styles of legends from Les Paul and Mary Ford, to Duane Allman and Richard Betts. Best of all, you instantaneously generate each flawless note without studio overdubs and multiple guitarists. That leaves you free to create complex, three-part guitar harmonies—in the studio or on the stage—with only one guitar.

The Smart Shift offers the musician a list of powerful features:
☐ Superb sound quality.
☐ Capability to produce two intelligent harmonies per note played.
☐ Two separate user-defined intervals allow a musician to program two discreet intervals for harmonies. Intervals are displayed in semitones from –12 to +12.
☐ 41 harmony types means the Smart Shift will accommodate virtually any style of music that exists.
☐ A displayed value specifies the MIDI channel to be used for both the reception and transmission of program changes.
☐ Smart Shift capability allows a musician to save and access presets at will.
☐ Effect bypass permits user to defeat the harmony effects either at the panel or with a footswitch.

The Smart Shift delivers virtually unlimited power and flexibility to create complex guitar harmonies.

DigiTech
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