Electronic MUSICIAN

JUNE 1987   A MIX PUBLICATION   U.S. $2.50/CANADA $3.50

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06

HOW TO BOUNCE TRACKS WITHOUT BOUNCING OFF THE WALLS.

The more tracks you record, the harder they can be to keep straight. And the more you'll appreciate the personal recorder that makes it easy to sound good.

The Tascam 246 Portastudio.

EVERYTHING'S UNDER CONTROL.

To begin with, the controls of the 246 practically explain themselves. When you want to assign any of its 6 channels to any of its 4 tracks, just push 1, 2, 3, or 4 on the channel strip.

That's it. The signal's routed. (And mixed, if you've decided to send other channels to that track.) No blind guesses about what's bouncing where. Or about levels, since the 246 has separate sets of meters for recording and mastering.

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The 246 cuts down on confusion. As well as hiss. In fact its built-in dbx gives you 10dB more broadband noise reduction than Dolby C.

And if you want to record Code or FSK, you can clear dbx out of Track 4 with the flip of a switch. What's more, you won't have to sacrifice Track 3 to guard against crosstalk, thanks to Tascam's advanced head design.

PUT MORE IN. GET MORE OUT.

Each channel of the 246 has line *and* mic inputs, complete with effects and EQ. Even Channels 5 and 6.

Which makes things much less frustrating when, for example, you're trying to mic a full drum set. Or use a multi-keyboard MIDI setup.

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Visit a Tascam dealer today and test a Portastudio.

No matter what you assign it, the 246 will keep things simple. While keeping you calm, cool, and creative.

TASCAM

Electronic
MUSICIAN

A MIX PUBLICATION

DEPARTMENTS

COVER

To make your own distinctive sound, sometimes you need to make your own distinctive equipment and accessories. More and more musicians are finding out it's not that hard to "do-it-yourself"—whether building an electronic device or writing a custom piece of software—and in the process, they're experiencing the satisfaction of personal accomplishment. Photo by John Helyar.

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Editor's Note

I hear a lot of complaints about the manuals that come with musical instruments and software. There's a reason for these complaints— many of them are justified. But sometimes it's all too easy to blame an otherwise adequate manual when the problem is due more to unrealistic expectations on the part of the user.

Some people expect that after reading a manual, all knowledge about a piece of equipment will be magically transferred into their brain cells. While this might be true for a superbly written manual about a simple piece of gear, today's synthesizers and software programs are *extremely* complex, and it takes practice to learn all of an instrument's nuances. It is impossible to absorb all of a program's capabilities at one sitting, no matter how good the manual.

This complexity also tends to work against logical organization. So many features are interdependent, and there are often multiple ways to accomplish the same function. Also remember that products are fine-tuned right up to their introduction, which makes it that much harder for the manual writer to learn how to use the equipment. In fact, if a manual accurately documents all of a program's features, that's quite an accomplishment in itself. As manual writers acquire more sympathy for the musicians who have to struggle through a learning curve, perhaps musicians will appreciate the trying conditions under which most manuals are written.

What brought this subject to mind was a Macintosh program I use that some people have told me was "full of bugs" and "poorly documented"— the two most common gripes I hear about software. When a friend of mine first started working with the program, he said it crashed all the time. After he got more familiar with the correct protocol for using the program, the number of crashes greatly diminished. It wasn't a case of the program fixing itself; rather, my friend learned to use the program *the way it was intended to be used.*

My experiences paralleled his. When I first got the program, I immediately learned the minimum of what I needed to get "on the air." Like many programs, if you don't know the proper way to implement a function, you can often figure out some roundabout way of doing the same thing...but because the program was not designed with makeshift techniques in mind, trouble can result. After I studied the manual in greater depth, surprise! There were all kinds of shortcuts for doing things in simpler, more efficient, and far more crash-proof ways.

A manual is not something you read once. Nor is it just something you refer to when you're in trouble. It's something that should be read periodically, from cover to cover, until you feel you understand every operational aspect. Underline sections, make notes in the margin, and when you discover neat little tips and applications that aren't in the manual, write them down—and send them to the manufacturer for when the manual gets updated.

Learning complex gear is a complex process. Sure, many manuals could be better, and some are downright terrible; but before you condemn the manual, remember that it can't *make* you an expert—only practice will do that.

(P.S. Many, many thanks to all of you who sent in your reader surveys. Your help has been absolutely *invaluable* in helping us chart the future direction of the magazine, and after we tabulate the mountain of responses, we'll print the results. Thanks again—your feedback is what keeps the magazine on course.)

Do Clone Around

The letter from Daniel M. Colopy (March '87 EM) in response to Jason Rich's article (Jan. '87 EM) about the dangers of buying an IBM clone is overly alarmist. Buying a clone is about as risky as buying canned corn at the supermarket. (For those desiring additional controversy, see Robert L. Sabaitis's letter in the April '87 EM).

I'm a PC software developer, editor of *Programmer's Journal* (a 15,000 circulation magazine for PC programmers), and I wrote The 48 Track PC, a MIDI software program for the PC. I haven't heard of a clone incompatibility problem for years. The clone has become the standard of the marketplace, and its popularity is the main reason IBM's profits fell by billions last year. My old IBM sits in the corner, or is used as a loaner. My clone is bigger, faster, and cost $550.

There is absolutely no reason not to buy a clone. You should probably stay away from clones above 8 MHz until they become widespread, but that is the only caution I can think of noting. The software problem that Colopy mentions, that of the software coming up with zero memory available, sounds as though it is caused by an ancient and quirky method of determining available memory.

If software has trouble on a clone, don't shelve the machine—call the developer! That developer is missing a market of millions of machines, and believe me, will be most interested in bringing the software up to date. You can spend a week in Hawaii on the $700 you've saved on the clone while the developer works on the update.

Robert Keller
Eugene, OR

PC Fans, Please

I am interested in communicating with anyone who is using the IBM PC family for the production of electronic music. I can be reached by mail (2532 Highland Ave. #20, Cincinnati, OH 45219), or on BITNET (the academic network) at node UCCCMVS, user PCNERKG.

Daniel Mocsny
Cincinnati, OH

Using Your Head

In your reply to a letter from Bill Murray (Dec. '86 EM), you said that recording the same signal on all four tracks of a 4-track recorder would increase the overall signal level, but also increase noise by a corresponding amount; and also addressed the question of head misalignment on playback. The reply seems to have been written based on a "common sense" view of the situation, which in this case led to some problems.

First, although it is true that the increase in signal (when mixing more than one "identical" track's output) is indeed accompanied by an increase in noise, the two do *not* increase proportionally. The signal is correlated between tracks, and sums as a voltage,

while the noise is uncorrelated and adds as a power. Simply put: combining two tracks should increase the signal by 6 dB, but the noise by only 3 dB. In general, when you increase the width of a track by a factor of two, you gain 3 dB in the signal-to-noise ratio (all other factors being equal). So recording the signal on four tracks in parallel and using the sum of all four should increase the signal-to-noise by 6 dB, a worthwhile gain.

There is one difficulty. As the track gets wider, the azimuth alignment (to avoid high frequency rolloff) becomes more critical. However, in this case we are not dealing with a single playback gap, but with four (more or less) independent gaps. Thus, in this case the factor that could introduce high frequency rolloff (beyond what is normal for a single track) is not the azimuth per se, but the phase shift between channels. There are two distinct possibilities here: a static phase shift is likely if the record and playback heads are not the same. This could be worked around by trying various combinations of tracks, but it's possible (though not likely, I think) that none of them would be closely enough aligned.

Even if the same head is used for recording and playback, there will likely be a significant dynamic phase shift between channels, due to the constantly shifting skew of the tape as it travels past the heads. This causes a varying track-to-track phase shift during both recording and playback. For this effect there is little that can be done other than using the best transport mechanism available and a brand of tape that doesn't aggravate the problem (i.e. avoid cheap cassette shells), but even the best of both may not be sufficient to avoid the problem.

Ultimately, I would advise Mr. Murray (and anyone else with similar needs) to try the multi-track approach, but to test it first—record a known signal with significant high frequency content, and try various combinations of tracks on playback. It may or may not prove to yield better results than using a single track. In any event, your suggestion to record on all four tracks to allow selection of the best one is certainly worthwhile. To this I would further suggest recording at several different levels to allow later selection of the best compromise between low noise and tape overload, and/or for matching the levels between samples for different segments of the keyboard (if you're using multi-sampling).

Martin J. Maney
Chicago, IL

It's CZ When You Know How

As a new owner of a Casio CZ-101, I eagerly followed Alan Gary Campbell's advice and purchased a Radio Shack 277-1026 adapter. When I plugged it in, I found that it only made intermittent contact, cycling the synth on and off every time it was moved. For those readers with a similar problem, do not despair. Radio Shack sells a plug with the correct inner diameter, part number 274-1570, and I purchased two for 99 cents. I cut off the old plug, checked the polarity and found that the adapter wire with the white stripe attaches to the center connector. A little soldering, and the adapter is working perfectly.

Mike Epstein
Cheshire, CT

The Obsolescence Blues

I have been very troubled over the past few months by something that must be on the minds of other electronic musicians besides myself. Although I am only 16 years old, I have a considerable MIDI setup. I have earned every penny that went into the system and am very proud of it. But, every time I pick up the latest issue of EM, I am faced with the ever-changing world of electronic music. After reading your "Report From Japan" and the new DX-711FD advertisement, I won't ever be able to look at my CZ-1 again without saying "why...why...why didn't I wait?" And look at my Prophet, my $2,500 baby, which suffered the highest of all letdowns: *discontinued!* I see Emax or S-50 in the silver ball; how can I ease my appetite for the new thing? Please give me some answers—tell me my CZ-1 is not an endangered species and that my Prophet is worth more than $1,000 (but don't lie). Oh yes—and keep up the good work!

Dan Saltzman
Natick, MA

Dan—Ain't nothin' wrong with a CZ-1. Or a DX711FD, or an Emulator II (even if Emax has an edge in the specs department), or a minimoog, or an arch top Les Paul. For a quick return to reality, read E.A. Holley's "The MIDI Virtuoso" in this issue and you'll probably feel better. Meanwhile, think of all the "orphan" computer owners out there, and be happy that the instruments you have work and are still supported by their manufacturers. As to what a Prophet is worth, well, it's worth what someone will pay. If it's the older Rev 2 model, there are big-name Rev 2 fans who would probably be glad to pay more than $1,000 to have a backup around. And don't sell your CZ-1; wait for the expander box version of the DX711FD to come out, and integrate it into your

setup. Remember too that the obsolescence factor works both ways—a year from now, you'll be able to pick up last year's New Thing at a real good price.

The About to Be Obsolete Blues

I'm about to spend what is a lot of money for me on a rack mount sampler. I'm currently leaning towards an Emax rack. But given the rapid changes in technology, I don't want to blow my wad on an Emax and have a new level of sampler (or keyboard) technology emerge a month later.

I'd greatly appreciate any info you might have on what's on the horizon (and when) that can compete with an Emax or even all samplers. Are any current samplers expandable enough to include any new features soon to be introduced? And are there any suggestions you might have before I blow my $2,300?

Rick Tavello
Oxford, CT

Rick—Meet Dan Saltzman, the writer of the previous letter. Regarding expandability, most samplers are subject to periodic upgrades that enhance performance, so in a sense they all are expandable to some degree. With respect to buying any instrument, I have the following observations:

✓ *No matter what you buy, something either better or cheaper will be available within six months to a year later.*

✓ *A good instrument doesn't become any less good because something better appears. In other words, an instrument will always have the features you bought it for.*

✓ *If you really can't decide on something, wait; eventually what you want will come along. The problem with this approach, of course, is you lose a lot of time where you could have had fun playing with an instrument. A two-year-old sampler is more fun than no sampler at all.*

Good luck with your search. For specific information on how to shop for a sampler, see Tim Tully's article "Choosing the Right Sampler" in the December '86 EM.

Kudos to O'Donnell

I wanted to write to congratulate you on Craig O'Donnell's 4-track article (March '87) which I think is most helpful and informative to anyone searching around for a 4-track. I am a subscriber and am pleased to see you publishing articles that can truly assist either amateurs or serious pros lost within a world of MIDImania. Keep up the good work!

K.C. Porter
Hollywood

Yugoslavian Addendum

I have some corrections and additions to "Electronic Music Behind the Iron Curtain" (March '87 EM), as follows:
1. Yugoslavia is not an Eastern Block country, and its treatment of pop music is entirely different than in the Eastern Block countries. Most of the major rock acts (Rolling Stones, Talking Heads, Wings, Jethro Tull, etc.) toured the country at one time or another, which is

not the case with the Eastern Block countries!
2. There is a large number of *privately owned* and operated recording studios in Yugoslavia, and a great number of recently released records were recorded in these privately owned studios. There are some basic similarities in the process of getting signed, one of them the privately financed (by the artists themselves) LP recording which is then shopped around since there is a choice of labels to choose from and offer the tape to.
3. The correct spelling of the last artist mentioned in the section on Hungary is Gabor Presser (not Pressor), and just for your info he was a member of Omega until 1970. A couple years ago he appeared with the "old" Omega members in a reunion-type concert which was recorded live and released as an album. While he was with Omega, he recorded three albums before forming Locomotiv GT, the first truly Hungarian super group. He even came to the States a couple of years ago, and composed music for a stage play directed by a Hungarian director, somewhere in the Midwest. (Incidentally, both the guitar player Tamas Barta, and drummer Jozsef Laux, from the first edition of Locomotiv GT, emigrated to the West. According to some Hungarian emigrees, who I met in Munich four years ago, Tamas came to the U.S. where he died from an overdose, and Jozsef Laux works in a California music store.)

I was born and raised in Yugoslavia and was a freelance composer for the Novi Sad Radio TV station, so I know how things were. My hometown is in Northern Yugoslavia just 20 miles from the Hungarian border; during the early '70s, while working as a DJ for the local radio, I had a chance to interview most of the pop groups coming to town, which is how I met Omega and learned of their internal changes.

Tibor Spiegel
Elizabeth, NJ

Better CZ Patch Librarian

Many thanks to Tim Dowty for the excellent CZ Patch Librarian (February '87 EM). In response to Mr. Dowty's call to "speed up the disk saves," the following changes will reduce the disk access time during a save from approximately 75 seconds to 15 seconds:
1. Delete lines 5010 through 5070.
2. Move lines 550 and 555 to 5200 and 5210.
3. Add the following lines:

```
5010 fori=43to52
5020 poke 900+i, peek(i)
5030 next
5100 poke43,0:poke44,136
5110 poke45,1:poke46,152
5120 poke47,1:poke48,152
5130 poke49,1:poke50,152
5140 poke51,255:poke52,159
5150 poke55,255:poke56,159

5200 print:printtab(5):inputff$
```

```
5210 if ff$= orff$=then5300

5220 save ff$,8

5300 poke43,peek(943):poke44,peek(944)
5310 poke45,peek(945):poke46,peek(946)
5320 poke47,peek(947):poke48,peek(948)
5330 poke49,peek(949):poke50,peek(950)
5340 poke51,peek(951):poke52,peek(952)
5350 poke55,0:poke56,127

5999 return
```

Lines 5010-5150 change the pointers which tell BASIC where to find its program text and variables and point them to the patch storage area. Line 5220 merely saves that portion of memory as if it were a BASIC program. Lines 5300-5350 restore the pointers to their original condition upon entering the routine. Lines 5200 and 5210 must occur here and not at 550 and 555 since the variable ff$ must be input after the pointers are changed, or else BASIC will not know where to find it. For that same reason, the values of the pointers are stored in the cassette buffer (lines 5010-5030) and not as variables so that they may be retrieved while BASIC thinks its program text is the patch storage area.

I hope that readers will find this modification useful. I can't live without it!

Gary S. Karpinski
Director, Lab for CAI in Music
City University of New York
Brooklyn, NY

Another Attempt to Break the Speed Limit

C-64 disk output can indeed be sped up using machine language. Fortunately, the ML already exists in the machine's operating system. See the following re-written Write routine.

```
5000 REM •••••••••••••••••••••••••
5001 REM *NEW WRITE TO DISK*
5002 REM •••••••••••••••••••••••••
5010 CLOSE 15
5015 SYS 57812FF$,8,1
5017 REM NOTE THE SYNTAX ABOVE
5020 POKE 193,DS-INT (DS/256)*256:POKE
     194,DS/256
5030 POKE 174,DE-INT (DE/256)*256:POKE
     175,DE/256
5040 SYS 62957
5050 OPEN 15,8,15
5060 GOSUB10000
5999 RETURN
```

First, a ROM routine is called to set up the file name and device number. Next, the start address is POKEd into locations 193 and 194 in low-byte high-byte format. Then the end address +1 is POKEd into locations 174 and 175 and another ROM routine is called to perform a SAVE. The program then reads the error channel to check that all is well.

Richard Curcio
Brooklyn, NY

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

STRUCTURE

S=SYNTHESIZER SOUND GENERATOR
P=PCM SOUND GENERATOR
R=RING MODULATOR

OUTPUT MODE

1 UPPER / LOWER — REV — U OUT / L OUT
2 UPPER / LOWER — REV — U OUT / L OUT
3 UPPER / LOWER — REV — U OUT / L OUT
4 UPPER / LOWER — REV — U OUT / L OUT

P-ENV

L1 · Sus L · CENTER PITCH · T1 T2 T3 T4 · L0 · L2 · End L · KEY ON · KEY OFF

TVF/TVA ENV

L1 · L3 · L2 · Sus L · T1 T2 T3 T4 · End L · KEY ON · KEY OFF

EQ

Lf · Hf · BOOST · LOW FREQ · HQ · Lg · HQ · Hg · HIGH FREQ · CUT
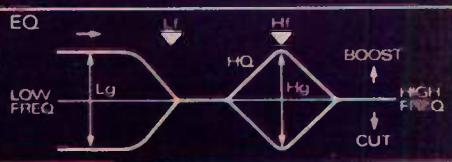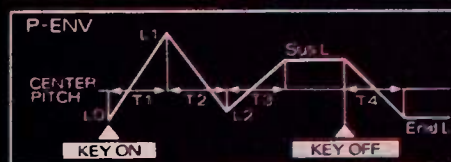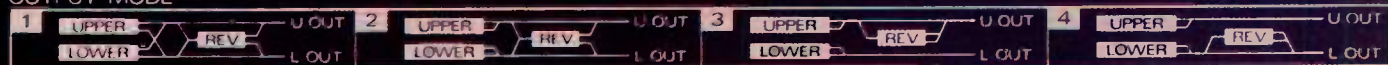
INTRODUCING THE D50
THE BOLD NEW FORCE IN DIGITAL

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

FIGURE 1 PATCH CREATION

predictions, analysis and control capabilities. Roland engineers have spent years developing a new highly sophisticated LSI chip, code-named the "LA Chip," that utilizes a linear arithmetic technique to digitally synthesize sounds. The "LA Chip" is the heart of the D-50.

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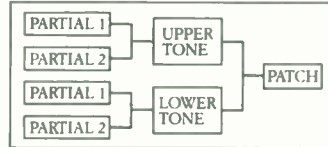e a Tone, and two Tones are combined to create the Patch. (Figure 1) The D-50 can hold 64 Patches and 128 Tones. Each of the two Tones can be processed individually by on-board signal processing that is sophisticated enough to rival a rack-full of equipment, and includes digital reverb, digital parametric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

LINEAR SYNTHESIZER
SOUND SYNTHESIS TECHNOLOGY

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

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

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

FIGURE 2 STRUCTURES

PCM Sampled Partials/A Partial can also be more than a digitally synthesized signal, it can also be a PCM sample. Resident in the memory (ROM) of the D-50 are over 100 carefully selected 16 bit PCM Sampled Wave Tables which can be used by themselves, combined with Synth Partials or combined with each other. The PCM Partials are carefully selected, and digitally processed so that they combine well with other Partials. Some of the sounds include a wide variety of the attack portions of percussive sounds: marimba, vibes, xylophone, ethnic instruments, grand piano hammer attack (with the fundamental removed), a variety of flute and horn breaths, a range of different string plucks and bows, nail files, guitars, and many more. The Wave Table library also includes Loop sounds and long samples, such as: Male and female voices, organs, pianos, wind and brass instruments, and also Harmonic Spectrum sounds, which are created by removing all of the fundamentals of a sound, isolating its harmonic components.
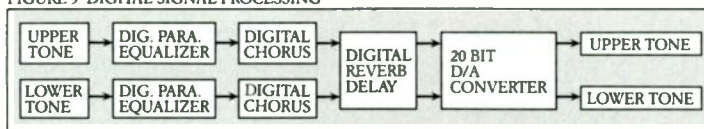
The sounds created by the D-50's PCM Waveform Generator are far superior to wave table samples found in other synthesizers, which are usually only one looped cycle in duration, and are usually no more than 5 milliseconds. In contrast, many of the PCM Partials on the D-50 are up to 256 milliseconds.

Structures/The combination of the Partials' operation modes can be set by selecting one of the seven Structures. (Figure 2) By choosing one of these Structures it is possible to combine two Synth Partials, or two PCM Partials, or a combination of the two in several different relationships. In addition, the Partials can be cross-modulated by the digitally-controlled Ring Modulator, which helps to create the complex harmonic environment for the resulting Tone.

FIGURE 3 DIGITAL SIGNAL PROCESSING

Unlike ring modulators of the past (which tended to be interesting yet unpredictable), the Ring Modulator in the D-50 is designed to track with the keyboard, ensuring the proper harmonic relationships as you go up and down the keyboard.

Built-In Digital Effects/The final routing of the signal before it reaches the output is through the digital effects circuitry. (Figure 3) But, far from being merely an add-on, the D-50's effects are as carefully thought-out as the rest of the instrument, and likewise just as integral to the creation of new and unique sounds. The first effect is the digital Parametric Equalizer, used to contour the equalization curve for the tone before it passes into the digital Chorus, or we should say Choruses, as the D-50 fields an arsenal of eight chorus circuits—all available simultaneously, configured in any of 16 modifiable presets such as panning chorus, tremolo, flanging and much more. Within each chorus there are parameters set up as to how these choruses

PG-1000 PROGRAMMER

interact for maximum effectiveness. Lastly, the signal passes through the digital Reverb, which can also function as a digital Delay, offering various room and hall sizes, gated (non-linear) reverb, reverse, stereo panning effects that can be routed to either or both of the stereo outputs. The awesome power of these built-in effects means that the D-50 requires literally no outboard effects processing. And just as important, because all the D-50's effects are processed in the digital realm, they are completely noise free.

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

M-256D MEMORY CARD

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

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Dazed and Confused

Help. I am trying to learn how to set up and operate a simple MIDI system. I have a Roland TR-505 drum machine, Ross 4X4 cassette deck, 8-channel stereo mixer, and a 512K Macintosh. My goal is to set up a MIDI system where I can program a rhythm track from the 505 and bass/melody lines from a keyboard, and have them play back on command for both live performance and gigs. How do I do this? Do I need a sequencer or can I use the Mac? Will it hold up on gigs? How about a recommendation for an inexpensive keyboard? I'm lost! Tell me where I can learn the basics of MIDI and please suggest some of the best starting keyboards that are inexpensive (Yamaha DX100/Casio 1000 range). I'm sure there are others who would welcome EM's recommendation

David Engles
New Orleans, LA

David—For starters, I suggest a copy of MIDI for Musicians *(available from Mix Bookshelf, 2608 Ninth St., Berkeley, CA 94710;* ☎ *415 / 843-7901—write or call for free catalog). I wrote it with the beginner in mind and it has received much positive feedback from users and educators. Also see E.A. Holley's article on "Maximum MIDI for Minimum Money" in the April '87* EM. *Now for your specific questions.*

A Mac becomes a sequencer when you insert sequencing software. The "Big Three" sequencer packages for the Mac are from Op Code (Sequencer 2.5), Southworth (Total Music, which has been supplanted by MIDIPaint), and Mark of the Unicorn (Performer); all three have their proponents and detractors. Which one is right for you is a matter of personal preference, so go to stores and get demos. In addition, "Masterpiece" from Sonus will be out soon, and as we go to press Passport has just released the beta test version of an enhanced "Master Tracks" for the Mac.

Regarding keyboards, even though it's going to cost you more bucks I strongly recommend a keyboard with velocity response (i.e. dynamics) and preferably aftertouch as well. These will be able to help you record more expressive parts into your sequencer—even program drum parts that will play back through the 505 with dynamics. Two keyboards that might do the job are the Ensoniq ESQ-1 and CZ-1, but there are many other contenders; as with sequencers, only you can decide which synthesizer is right for you, and this will take some testing. To expand your system, I recommend the Yamaha TX81Z or FB01, or the Casio CZ-101. These are multi-timbral, which means that you can get multiple timbres out of either box when it's being driven by a sequencer. This will expand your sonic arsenal at a reasonable cost. Also note that the ESQ-1 is multi-timbral, and it even has an on-board sequencer that could be used instead of, or in conjunction with, a Mac sequencer.

Most importantly, keep reading, go to stores and local clinics or shows, join a user group, and expand your knowledge. Assembling a MIDI sys-tem is a long-term process that requires a lot of research, but the results are worth it.

More About Sampling Limitations

I received my March issue—another in a long line of great ones. The 4-track comparison is invaluable.

I wanted to make the following comment in reference to Bolman's letter on page 13. Bolman says that he has tried 42 kHz sampling rates and drastic EQ when sampling with the Prophet 2000, but has still not obtained the kind of high frequency sound quality he wants. Therefore what must be frustrating him is not the number of bits, as you speculated in your response, but probably the fact that when the 2000 is sampling at 42 kHz, there is a rather steep 16-kHz anti-aliasing filter sitting in front of the ADC. At 31 kHz sampling rate, the filter is set to 12 kHz. So there is quite a difference between nominal sampling rate and frequency response. This must be more or less true for many samplers.

I think there may be a widespread misconception (which manufacturers are not anxious to refute), that the frequency response is automatically just half of the sampling rate. Digital "brick wall" filters are still too expensive and analog attempts mutate the sound. For sampling to approach the real-world fidelity that Bolman expects, he will need a 60-kHz machine, so that affordable filters can be moved up to about 22 or 25 kHz. (I'll be right in line behind him.) As I understand it, the number of bits has only a marginal effect on frequency response. Most of the fidelity benefits of 16 bits comes from increased dynamic resolution, as well as a reduction of the omnipresent quantization noise which reduces sonic details to stairsteps. After all, all 12-bit samplers have a maximum dynamic range of only 72 dB—I have a cassette deck with dbx that can beat that!

Stanley Jungleib
Sequential Circuits

The IBM MIDI Card

As we go to press, we received the following letter regarding IBM's new MIDI card. William H. Post has worked for IBM for 28 years and is also an amateur musician.

IBM's Music card attaches to the new Personal Systems computers and previously existing PCs, XTs, and ATs. The Music card is not Roland MPU-401-compatible, but it performs the same basic functions. The card also includes an 8-voice synthesizer with 240 preset voices and 96 programmable voices. (Yamaha was consulted for the creation of this card, and the synth section looks a lot like an FB01.) The card is warranted for one year and lists for $495.

Depending on the software, you can enter music either through a MIDI keyboard or the computer keyboard and do all the standard computer tricks like saving to mass storage or transmitting via modem. Editing functions include changing voices, transposing keys, and changing notes and/or volume. A number of software manufacturers have signed up to provide software for the Music card before June, 1987. They include Yamaha, Jim Miller, Passport, Electronic Arts, and Dr. T's.

The Music feature includes two hardware elements. The *Music card* is a 13-inch full-length computer card; using two cards (more aren't allowed) provides 16 voices. Preset voices include wind instruments (flute, clarinet, saxophones), brass (trumpets, tubas), percussion instruments, piano, synthesized instruments, wind, rain, and lots of fun sounds. The card includes the following:

✓ A MIDI interpreter, which interprets the contents of the MIDI data stream;

✓ An FM sound generator that produces the 336 sounds;

✓ A digital-to-analog converter which produces signals that can be played through external stereo amplifiers;

✓ An audio preamplifier, with sufficient amplification to drive headphones; and

✓ Stereo audio output ports. A sound can be directed to the left or right channel.

The *Converter box* connects to the card through a cable and a D-shell connector on one side and provides MIDI In, Out, and Thru.

Optional packages available from IBM include Hardware Maintenance Supplement and Diagnostics diskettes, and a Personal Computer Technical Reference supplement. Set up instructions shipped with the card indicate that the customer will have to set some switches on the card to identify whether it is the first or the second, and so on. Failures generated during the initial set up will result in a screen message which references the user to the Guide for Operations.

It looks like IBM has ventured off into a new environment with a product that could become an industry standard. In any event, it will certainly help bring down the cost of getting into musical electronics for those who own IBM PCs or true compatibles.

William H. Post
Buena Park, CA

Error Log

In "Build the Alpha Digital Drum" (February '87), C9-C13 should be 100n instead of 100μ, the 12 inverters in the lower right hand corner are 4049s (IC6 and IC7), and VR2 (tune) should connect only to R11 and ground as shown; the third terminal should not connect to anything, instead of connecting to +5V. We regret any inconvenience this may have caused.

In "Opening the Door to Music Math" (Dec. '86 EM), page 64, the third and second from last sentences should read "Also, major thirds are not 'pure,' that is, the relationship between A and C# is not a ratio of 5/4 which is the intervallic ratio for a pure major third. A pure major third above 440 Hz would be 550 Hz (440 × 5/4 = 550) and not the 556.88 we see in **Fig. 2**." **EM**

ACCESSORIES

▶The **P-12** compact disc player ($899) begins as a hand-selected Philips CDB-650, then is rebuilt, retested, and burned-in to give improved response. Features include four-times oversampling (176.4 kHz), 16-bit D/A conversion, digital output jacks (for coming CD applications—CD-ROM, digital sound processing, sampling), 758 selection programming and phase linearity of ±0.15% from 20 Hz to 20 kHz.

American Audio Laboratories
PO Box 588
Greer, SC 29652
☎ 803 / 271-4252

▶New **Mapper** updates allow two simultaneous "virtual maps" so that keyboards, sequencers, or other MIDI controllers at each of the two MIDI inputs can be mapped separately (e.g. with two keyboards connected, the Mapper can set up a complex series of key splits controlled by one, while simultaneously setting up a completely different series of splits and harmonizations for the other). Also, the second input can now pass MIDI clock and song pointer information. Field updates are available for Mapper owners; new Mappers include the updates. Finally, custom software services that address specific system problems are also available for the Mapper. PAN electronic mail: contact AXXESS.

Axxess Unlimited
PO Box 8435
Ft. Collins, CO 80525
☎ 303 / 482-5656

▶**Power Loss Shutdown** (priced from $150) protects microcomputers operating with uninterruptible or standby power supplies by powering down an entire computer system if commercial AC power has not yet been restored when UPS or SPS batteries have discharged. Models with adjustable AC line dropout voltage level are available to accommodate the inherent brownout protection capabilities of most UPS systems.

Electronic Specialists, Inc.
171 S. Main St.
PO Box 389
Natick, MA 01760
☎ 800 / 225-4876

MICROPHONES

▶The **Model SM15 Head-Worn Condenser Microphone** ($275) uses a completely new unidirectional electret condenser car-
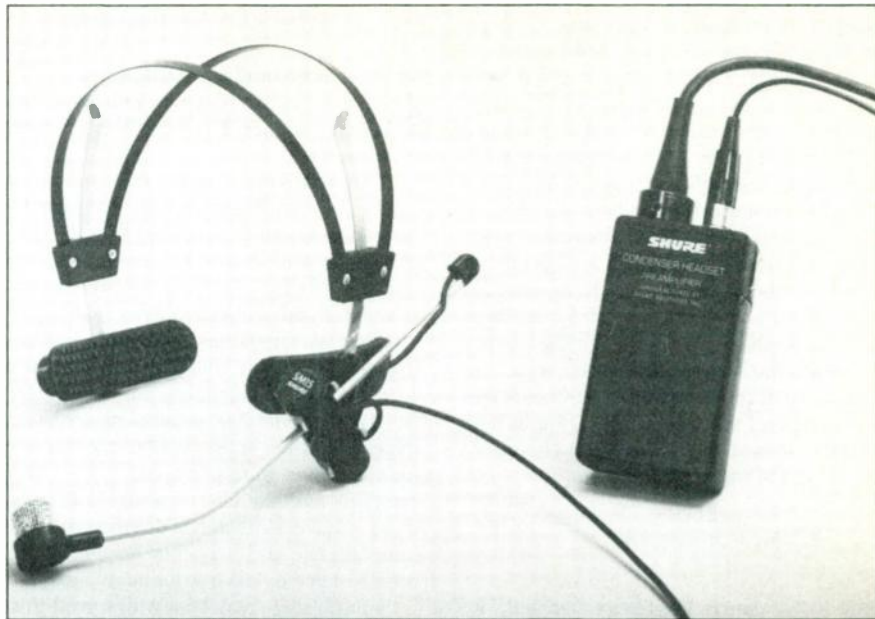
Shure SM-15 headset condenser microphone

tridge design that offers a smooth, natural-voice frequency response (50 to 15,000 Hz). The SM15 also features a newly designed double-braced headband with special grip pads that allow the user to twist and turn without fear of losing his microphone. The SM15's preamplifier can easily be clipped to a belt or waistband, and is powered by a readily available 9-Volt battery or by phantom power. The SM15 comes complete with 4-foot microphone and 10-foot amplifier cable, a windscreen to protect against wind noise and explosive breath sounds, and a custom carrying/storage case.

Shure Customer Services
222 Hartrey Avenue
Evanston, IL 60202
☎ 312 / 866- 2553

PUBLICATIONS

▶**Microtonality** will be featured in the Spring 1987 issue of *Computer Music Journal*. Single copies are $7; a one-year, four-issue subscription is $26 (institutions $55). Also, a CD, "New Computer Music" ($15.95 plus $1.50 p/h) features a variety of new computer music by international composers.

MIT Press Journals
55 Hayward St.
Cambridge, MA 02142
☎ 617 / 253-2889

▶The *Perfect Vision* ($22 per year; single copies are $6.50) is a quarterly publication dedicated to pursuing a quest for

quality in video reproduction and playback. Edited by Harry Pearson (publisher of *The Absolute Sound*—a high-end audio magazine), it focuses on issues addressing the home video market, technical observations, opinionated reviews of video software and hardware, and the sound quality of each element.

The Perfect Vision
PO Box 357
New York, NY 11579
☎ 516 / 676-2830

SIGNAL PROCESSORS

▶The **MP100** programmable instrument preamp ($999) allows use of 100 preset sounds of an almost unlimited variety, all with one convenient footswitch. It features simple pushbutton control of 29 parameters, and includes three internal distortion circuits, split 14-band graphic EQ, compressor, MIDI In/Out/Thru, and three programmable stereo effects sends and returns.

Analog Control Technology
225 Parsons Street
Kalamazoo, MI 49007

▶The **Tube Driver** ($139.95), an overdrive preamp with EQ, powered by a 12AX7 Preamp Tube, responds to the player's touch like a real tube amp and offers exceptional warmth with superb harmonics.

Chandler Industries
PO Box 15-B, Station "A"
San Francisco, CA 94115
☎ 415 / 626-1742

▶dbx has developed technology that allows use of any music signal to analyze frequency response so that broadcasters and sound contractors will be able to analyze frequency response using the music being played, instead of test tones. Product availability is expected within approximately one year; a **Professional Real Time Analysis System** is projected to sell for under $10,000.

dbx
Professional Products Division
PO Box 100C,
Newton, MA 02195

▶The new **FX-42 Bass EQ** ($94.95) is a six-band graphic equalizer with a level control and extended low frequency reach for bass. Up to 18 dB of cut or boost is available for the six bands (50, 100, 200, 400, 800, and 1600 Hz). The **FX-82 Compressor/Sustainer** ($74.95) is optimized for bass. Since the loudest and quietest portions of the notes are compressed, notes sustain longer, the sound seems tighter and clipping going into other effects or the amplifier is avoided.

DOD Electronics Corporation
5639 South Riley Lane
Salt Lake City, UT 84107
☎ 801 / 268-8400

▶The **Model 464A Co-Operator™** ($959) is a stereo Gated Leveler/Compressor/HF Limiter/Peak Clipper that automatically rides gain, controls excessive high-frequency levels (with selectable pre-emphasis), effectively limits peaks and is switchable for stereo-tracking or independent
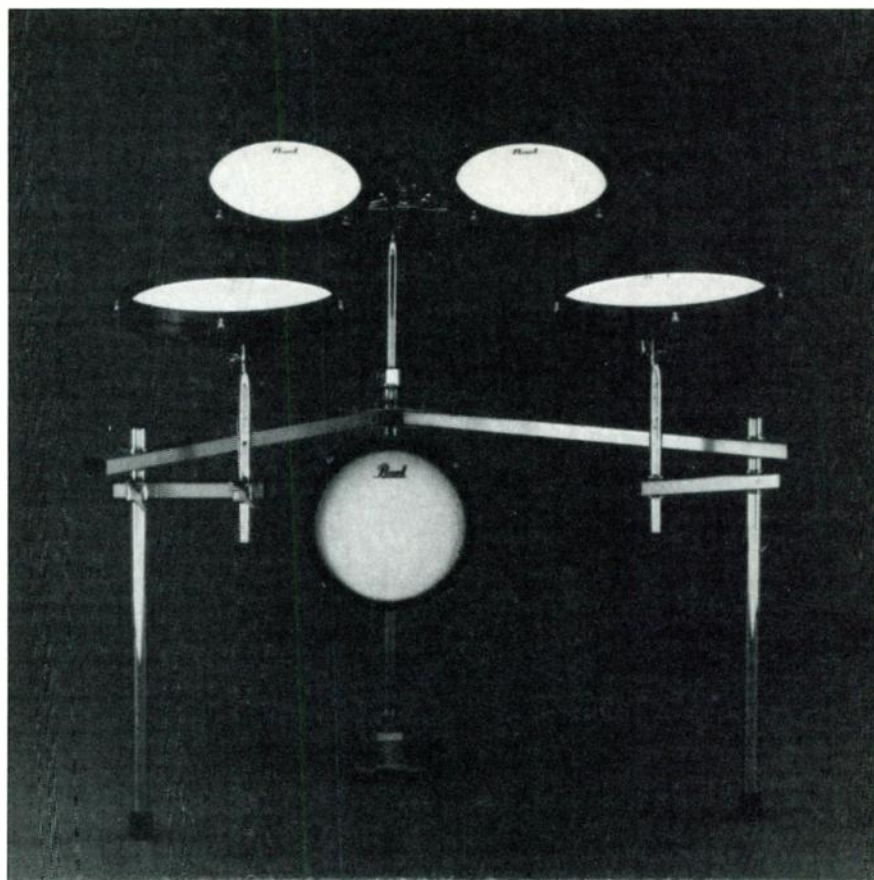
Pearl TK-5 practice pad set

dual channel operation. Two LED bar graphs in each channel simultaneously display gain reduction and peak output level.

Orban Associates Inc.
645 Bryant St.
San Francisco, CA 94107
☎ 415 / 957-1067

DOD Effects Pedals for bass

PERCUSSION

▶The **TK-5**, a new five-piece drum practice pad set, features tunable drum heads, making it possible to practice silently without sacrificing the feel of an acoustic kit. The pads also actually have rims, making it possible to play rimshots.

Pearl International, Inc.
PO Box 111240
Nashville, TN 37222
☎ 615 / 833-4477

SOFTWARE

▶The **Lighthouse** ($249.99), an 8-track sequencer for the IBM PC, features real time input, step editing, overdubbing, transposing (both music and channels), pull-down "Macintosh" like menus (operates from keyboard or mouse) and full playback tempo control. Operates in either internal, FSK or MIDI sync for use with multi-track tape recorders or drum machines. Hard disk owners have full subdirectory capabilities within the program. Requires a 384K IBM PC, monochrome or color monitor, DOS 2.1 or

higher, MPU-401 and MIDI-equipped instruments.

Bering Data Systems, Inc.
31-15 32nd Street
Box 18, Long Island City, NY 11103
☎ 718 / 626-2046

▶FB PRO ($129) for the Macintosh is a voice editor/librarian that allows full voice and configuration editing. Three Voice Bank windows each display all 48 voices in a bank; voices can be easily pasted and copied from window to window.

Digital Music Services
23010 Lake Forest Dr., Suite D334
Laguna Hills, CA 92653
☎ 714 / 951-1159

▶Master Tracks Pro™ ($299.95) is a professional MIDI recording and editing package for the Macintosh. Its five main modules provide 64 tracks of real time and step time graphic input, song and step editing, a system exclusive librarian and keyboard control mapper all in one package.

Master Tracks Pro is also available for the Apple IIe, IIGS and Commodore 128, and is faster and much more powerful than the original Master Tracks program.

It gives the user real time, step time and song mode sequencing with expanded memory (up to 325,000 notes on Apple IIe). Multi-track songs can be created in sections or as continuous works.

Score™ ($495), a desktop music publishing system for the IBM PC, allows a user to input, edit and print any piece of music on dot matrix or laser printers. Its graphic editing and page layout commands provide complete control over the placement, size, shape, spacing and output resolution of music notation up to 400 dots per inch.

MIDISOFT Studio ($99) is a low-cost, full-featured MIDI sequencer for the Atari ST family that forms the heart of a home MIDI recording studio for making demos, composing new songs, or just playing around. It has the same features of the best-selling sequencers at a cost-effective price.

Two new Macintosh Librarians for the DX/TX and CZ synths ($69.95 each) save and load voices and banks of sounds quickly on diskette. The Librarian displays preset sounds organized as on-screen cartridges; you can have multiple cartridges open at once and rearrange

the data by "dragging" voices between cartridges.

Passport Designs, Inc.
625 Miramontes St.
Half Moon Bay, CA 94019
☎ 415 / 726-0280

▶P-EV: Piano Evaluator ($395) allows students and teachers to build a library of keyboard exercises that can be practiced at the computer. Each performance can then be evaluated by the computer for pitch, touch and rhythmic accuracy.

Piano Evaluator comes complete with a selection of exercises, ranging from simple to difficult. As the student plays an exercise, the computer can evaluate accuracy of the performed notes, rhythms, or articulation (when used with a velocity sensitive keyboard). In addition, the sensitivity with which each performance is evaluated can be selected (beginner, amateur, advanced, or virtuoso) to suit the ability level of any student.

Exercises are entered into the Piano Evaluator by using a simple step-time editor which displays the music in standard notation on the screen. Those wishing to practice an exercise can see the music on the screen, which "pages" or

scrolls as it is played.
RolandCorp US
7200 Dominion Circle
Los Angeles, CA 90040
☎ 213 / 685-5141

▶**Cakewalk™** ($150), a MIDI recorder/editor for the IBM PC/XT/AT (256K memory and Roland MPU-401 required), features 256 tracks of unlimited length, a pull-down menu interface, a detailed Event View for editing MIDI parameters, and extensive global editing commands. Edit regions can be marked by ear and further refined using Event Filter criteria. A demo disk is available for $10.
**Twelve Tone Systems**
PO Box 226
Watertown, MA 02272
☎ 617 / 924-7937

OTHER NEWS

▶The **Center for Electronic Music** is a unique non-profit corporation chartered by the State of New York to provide inexpensive access to and instruction in state-of-the-art synthesizer, MIDI, and audio equipment. Dedicated to the concept of true public access, CEM has been founded by the former staff of New York's Public Access Synthesizer Studio (PASS), in order to provide a wide variety of services to anyone requiring advice, instruction and access to such equipment. Future plans include the construction of a fully professional multi-track electronic music production studio and the implementation of an artist-in-residency program so as to provide free access and training to qualified individuals.
Center for Electronic Music, Inc.
15 Charles Street (7E)
New York, NY 10014
☎ 212 / 924-7316

▶**DATs Arrive:** Aiwa, Matsushita, Sharp, and Sony announced the first Digital Audio Tape recorders conforming to government guidelines designed to discourage duplication (i.e. digital-to-digital copying is not possible; the units have only an analog out). These will be sold initially in Japan. The Aiwa **Excelic XD001** lists for 188,000 yen (approximately $1,200 at current exchange rates) and samples at 32, 44.1, and 48 kHz (the 44.1 rate is playback only). Other specs are frequency response of 2 Hz to 22 kHz, signal-to-noise ratio 92 dB, dynamic range 90 dB, and distortion less than 0.005%. Wow and flutter are unmeasurable. The Matsushita **SV-D1000** ($1,320) features similar specs but also includes a high speed
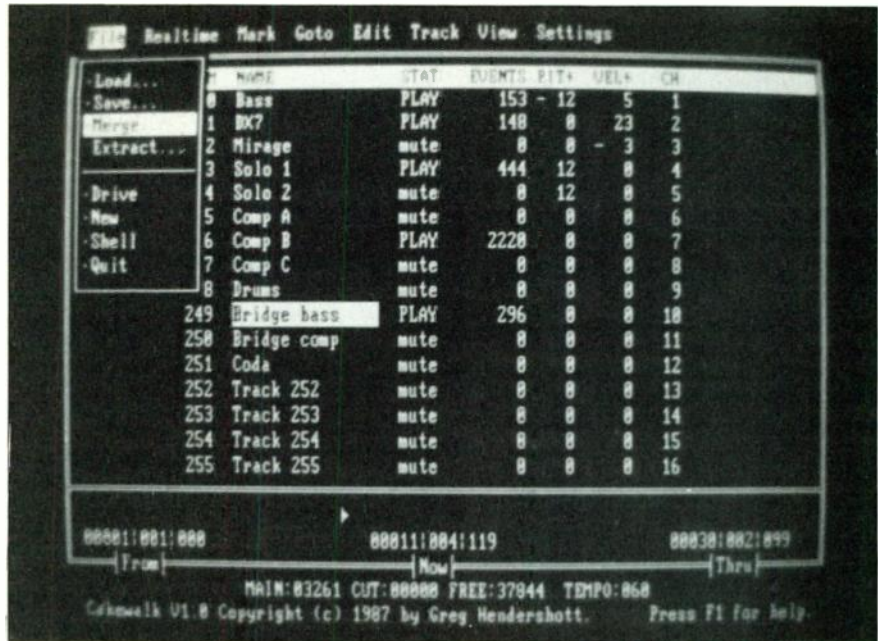
Cakewalk

search/rewind mode. Additionally, Matsushita announced DAT tapes ($13 for two hours, $9.33 for one hour). Sharp announced one standard size and one compact deck ($1,325 each); convenience features include 30-tune random selection, automatic recording of file name, and a 26-key solar-powered remote. Sony's DAT, the **DTC-1000ES**, lists for $1,333.

▶**High Quality VHS Format:** S-VHS is a high-quality member of the VHS family that uses half-inch tape, but is claimed to offer performance equivalent to one-inch tape. S-VHS tapes are not compatible with VHS machines, but VHS tapes can play back on S-VHS machines. Several Japanese manufacturers are gearing up to produce S-VHS machines.

CALENDAR

▶The second annual **San Francisco Music Fair** will be held May 29 through 31 at the Concourse at Showplace Square. Instituted last year by the San Francisco chapter of NARAS, the fair is a three-day event featuring live performances, exhibits and booths from instrument manufacturers and music support groups and services, as well as a wide variety of seminars featuring industry professionals. The SF Music Fair is *the* place in Northern California where state-of-the-art technology, the latest instruments, equipment, recordings, and the best in music industry services are all brought together under one

roof. For further information on exhibits, contact:
Events West
☎ 415 / 383-9378

▶The **Urban-15** group will host the 1987 Third Coast New Music Project at the Carver Community Cultural Center Auditorium in San Antonio, Texas, June 11 through 14. This is a four-day concert series of newly composed electronic and computer music by renowned sound artists. June 11 features Musimagery and Charles Ditto; June 12 is Robert Pearson and the Works on Disc competition winners; June 13 features Tina Marsh and the Creative Opportunity Orchestra, and Robert Pearson; June 14 is Urban-15, Musimagery and Gary Lee Nelson. Tickets are $5 per event, $9 for two events, $12 for three or $15 for all events. For reservations ☎ 512 / 225-6516; for information ☎ 512 / 226-2142.

▶The Institute of Audio/Video Engineering, located in Hollywood, CA, is sponsoring a summer recording seminar ($495) for music educators. The program lasts five days, July 27 through July 31, 1987. All classes are in a 24-track, state-of-the-art recording studio located at the Institute. Topics covered include recording techniques, outboard equipment, session procedures, mixing techniques, mastering, the use of sequencers, synthesizers and drum machines, MIDI, audio for video and film, and digital sampling.
☎ 800 / 551-8877 or 213 / 666-2380

Gateway Rhythm Machine
($25) ★★★

Hardware requirements: IBM PC or compatible computer; requires BASIC or BASICA, Roland MPU-401, and MIDI-capable drum machine.

And now for something completely different: the "Rhythm Machine" is *not* a drum machine sequencer or programmer, but is described as a program that creates rhythmic patterns. I consider it more like a controllable random drum-pattern generator, and feel its most useful application is to break a non-drummer out of drum pattern programming ruts.

This program is an independent effort—the software developer equivalent of an independent label musician—produced and written by Allen Conti of Gateway. The Rhythm Machine has some of the "rawness" of self-developed programs, but also its share of fresh, independent thought. To the benefit of both programmer and consumer, the program is priced very reasonably.

The program's premise is that two simple rhythms (referred to as "major" and "minor" generators) can generate many complex rhythm patterns through the use of an undisclosed algorithm. The major "generator" corresponds to the bass drum, and the minor "generator" to the snare. A third element can be introduced as a hi-hat component; this does not alter the rhythm, but adds a hi-hat hit with every percussive event.

The program requires that the drum machine be set to MIDI channel 16 (I wish this was variable). I recommend mapping the drum machine notes to the program; although the program's values can be changed once the program is running, the default values cannot. SCI drum machines work "as is," but the note settings do not correspond to Roland, Yamaha, and other devices. One setup caution: do not form a MIDI loop (i.e. use only the MIDI out from your MPU-401), otherwise the program might lock up.

Loading the program is exactly like running any other BASIC program. Upon loading, the program asks for the input of the two "generators" and a speed for the resultant rhythm to play. Also, the "duration" (total number of beats in a given measure) can be altered at this point. The screen then displays the resulting rhythm pattern in an easily understood format. You can switch generators if you like, even on a measure-to-measure basis, and specify the number of repetitions.

The output is but a few measures, with the only option being to "reverse" it. So forget about writing songs; to use this program for long compositions requires an external sequencer or a drum machine that can store user-definable patterns (many machines do this). The program patterns can be saved on disk, but not in any sequencer-compatible file format.

After spending several hours with the program and changing parameters through many

combinations, I found several rhythms that I felt were new, usable rhythms for my interests. By no means did I exhaust the possible combinations.

Some hints: Don't use "generator" values that are multiples of each other, or are whole numbers (fractional inputs seem to work best); "duration" values less than ten seem most interesting; and keep the number of repetitions very low, or you'll go nuts listening to some of the patterns you devise. Also, there is no obvious correlation between parameters and patterns, and the manual was quite oblique in this regard; sometimes a change of 0.1 can have a dramatic effect on the pattern, sometimes there's no change at all (this seems to be due to the rounding functions of the algorithm.)

For future versions, I'd like to be able to change parameters during playback—like most digital synths, you have to enter parameters and *then* discover what the sound is like. But despite any complaints, overall the Rhythm Machine is a unique program that

can spur your creativity. The fun factor alone easily makes it worth $25. Besides, it's always good to support independent efforts whenever possible.   —*Daniel Schroedinger*

Gateway
1700 Cleveland Ave.
San Jose, CA 95126
☎ 408 / 286-5490

JBL Control-1
($169.99) ★★★★★

A high-quality, small, two-way reference monitor speaker can be a thing of beauty. Many producers and engineers use small monitors as their reference (and preference) during tracking and mixdown sessions. If a mix sounds good and tight on them, then that mix will work on virtually any system. At the Gand Musitech show this past fall, I had the good fortune to hear the new JBL Control-1.

The loudspeaker system employs a 5¼-inch woofer and ¾-inch dome tweeter enclosed in a small, black, high-tech ported enclosure. This little speaker sounds great—uncolored and accurate. With a response of ± 3 dB from 120 Hz to 20 kHz, these monitors are tops in the price/performance category. If you are into a heavy bottom end, then the Control-1 is not for you, but if—like me—you appreciate a tight overall sound you will be pleased with the clean bass you *do* get. The midrange is smooth, and the highs sizzle without being harsh.

The Control-1 can also take lots of power. According to JBL, it can handle up to 150 watts; try hooking them up to a spare Crown DC300 with a CD of Mahler's First Symphony for an unforgettable experience. Yet they still work fine with low power amps as they are very efficient (90 dB at 1 watt).

With my own setup, I'm using a 50-watt Kenwood amp; the speakers are about 6-feet apart, with my mixing position as the third point of an equilateral triangle. This type of near-field monitoring works very well for me; recent mixes of mine have improved greatly

JBL Control One Monitors.

Coda

and sound good with a wider variety of playback systems.

Since speaker preference is so subjective it is hard to explain why I like these so much. To my ears they just sound exceptionally good! If you're looking for a small monitor speaker to complete or augment your setup, give the Control-1 a good, long listen.

—*Jeffrey P. Fisher*

JBL
Harman International
8500 Balboa Blvd.
Northridge, CA 91329
☎ 818 / 893-8411

Wenger Corporation Coda
($4, ppd.) ★★★★
Coda is an eclectic music software bibliography, with a twist: Wenger Corporation, its publisher, sells all the software described within, and after you've learned of its existence via *Coda,* they'd like for you to buy it from them, please, and they want your business so much that they've provided a toll-free number for ordering, and they'll even beat any documented lower price that you can find (though most of *Coda*'s prices are at suggested retail).

If that's all there was to it, it wouldn't be all that remarkable. But *Coda* just happens to be one of the most thoroughly artistic, visually appealing references you're likely to encounter on *any* subject. It's an enchanting mix of decoupage and art nouveau—in warm, muted colors throughout—that compel you to browse through its pages. (The black and white photo above is a mere shadow of its character.)

But it's more than that. *Coda* does a commendable job of the seemingly impossible: concisely communicating relevant information regarding the myriad music software available today. *Coda* has sections describing

hundreds of programs and accessories for the C-64/128, Apple II, Mac, IBM PC, Atari 130XE and ST, and Amiga—software for composition, ear-training, MIDI sequencing, music transcription, synthesis, and just plain fun.

Don't grab *Coda* from the shelf, hoping for a fast technical reference on MIDI Megabits 8.0 from Obscure Software, Inc. With entries that necessarily describe features more than theory, and only a brief glossary to demystify the jargon, it won't take you from a state of no knowledge to the stature of a computer/MIDI expert; *Coda* is not an encyclopedia. The index is cogent, but you have to know what you're looking for before you can use it; and *Coda*'s sections visually segue into one another—the headings seem microscopic among the text—yet, somehow, within the pages of *Coda* such piquancies seem almost charming.

Is it complete? *Coda* obviously doesn't list *everything,* or even everything that might subjectively be considered important (it didn't list any of my company's software!). Undaunted, Wenger intends to compensate for information omission and obsolescence with its monthly *Coda Report* (less art, more newsletter, *very* current—free to qualified music educators), and with an accelerating publication schedule that would eventually provide quarterly editions of *Coda.* But, for the present, is *Coda* timely enough to be really useful? Consider this unsolicited comment from a senior jazz composition major who happened to peruse my evaluation copy: "Wow. I need this. Are you selling these?"

As a software catalog, the *next* edition will always be the "best" one; but as *haute MIDI*— don't wait, buy it. (High school, music school, college, and university instructors would be absolutely remiss not to provide *Coda* for stu-

dents.) Odds are that *Coda* will introduce you to some really marvelous program that you would never have known of otherwise; and if it doesn't, you can always just sit there and *look* at it! —*Alan Gary Campbell*

Wenger Corp.
555 Park Drive, Dept. E2
Owatonna, MN 55060
☎ 800 / 843-1337

Roland MKS-50 Rack Mount Synthesizer ($695) ★★★
This modestly priced six-voice synth is basically an Alpha Juno in a rack. The MKS-50 stores 128 *tones*—pre-programmed sounds. Combining a tone with performance control functions creates a *patch*, and you can store 128 of them too. Patches and tones are stored as two groups of 64, in Roland Bank/Patch nomenclature.

Each tone has the following programmable modules: single DCO, high pass filter, voltage-controlled filter, voltage-controlled amplifier, chorus, envelope generator (feeds the DCO, LFO, VCF and VCA), and LFO (feeds the DCO, VCF and VCA). The DCO has some nifty extras—multiple waveforms that approach the kind of spectra you get with additive synthesis type stuff (although not as sophisticated), vibrato response to aftertouch, noise waveform, pulse width modulation, etc. The VCF and VCA do what you expect and thoughtfully respond to both aftertouch directly and velocity via velocity control of the envelope heights. The envelope generator is similar to an ADSR, but handles the AD part differently. The envelope amplitude rises for a specified time T1 to a level L1, then can either rise or fall over time T2 to level L2, then either rise or fall over time T3 to sustain point L3. Also, envelope times can get shorter as you move up the keyboard. These are just the basic options; all in all, the tone section is pretty versatile, especially at this price. Concerning sound quality, it's pretty much what you would expect from a single-oscillator-per-voice synth, with the significant exception that the sound has a bit of a brighter edge than usual and is very clean—no grit here.

The performance functions deal with MIDI parameters (whether bend, modulation, portamento, and other messages will be received), keyboard range, portamento time, volume, and so on.

A big plus is that the MKS-50 responds to MIDI poly or mono mode, which makes this a perfect synth for the beginning electronic guitarist on a budget—why pay for more voices than you need? And the manual is actually quite complete; it's very good on the MIDI documentation, although a little weak on explaining the big picture. Still, I had no problems figuring the unit out. Other nice extras include a stereo chorus (with stereo outs, although if the chorus is off, you only get a mono signal), 16 programmable chord memories for one-finger chords, nameable patches,
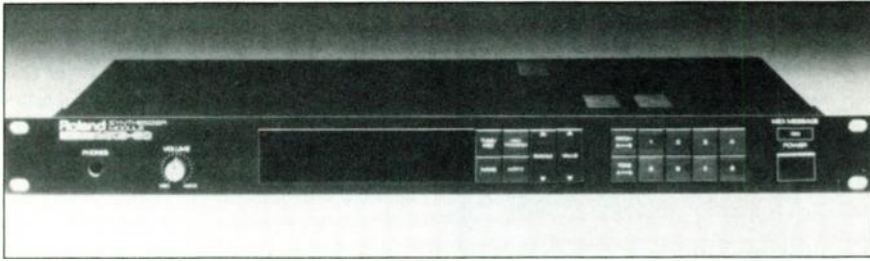
Roland MKS-50 rack mount synthesizer

a cassette interface, 16-character LCD display, and MIDI message indicator.

Programming is simplicity itself—and tedious. There are basically two switches: one steps through the available parameters, the other steps through the available values for those parameters. With 18 parameters in the Performance Control Functions and 36 tone parameters, that can be a lot of stepping! This approach works for tweaking patches, but for creating patches from scratch, Roland recommends their external PG-300 programmer—I can see why. Surely some voice editors will appear soon to also make life easier.

So we have a clean-sounding, inexpensive synth whose only real drawbacks are the lack of easy programming and the somewhat thin sound that results from a single DCO design. Aside from those two caveats, the MKS-50 provides an excellent opportunity to pick up a rackful of analog synth sounds at a budget price. —*Craig Anderton*

RolandCorp US
7200 Dominion Circle
Los Angeles, CA 90040
☎ 213 / 685-5141

Maniac Music Sustainiac™
($299.90) ★★★★
Since the dawn of rock and roll, listeners have said "turn it down!" To which guitarists say (or more likely, shout, having lost much of their hearing), "but I have to play this loud to get the tone I want."

While fuzztones, Heet E-Bows, and Scholz Power Soaks have helped reconcile these two viewpoints, until now no commercially available device could offer the kind of polyphonic, harmonic-enhancing physical feedback that comes only from blasting a guitar body with high-intensity sound waves at close range.

The Sustainiac's basic idea is simple: you tap a signal from your guitar, feed it to a little AC-powered floor box that processes and amplifies the signal, then send it to a transducer that attaches to the guitar headstock. The transducer turns the music you play into vibrations, which vibrate the body and cause feedback.

Several controls let you tailor the nature of the feedback, within the basic constraints of the concept. *Power level* determines the drive to the transducer, *auto-sense* provides variable compression to increase sustain, *harmonic enhance* cuts low frequencies to limit sustain to the higher notes (this control also provides an

"anti-squeal" position to minimize that ugly squeal that is most prominent when the strings are idle), and *threshold* sets the level at which the guitar signal goes to the transducer. Yes, we're talking *controlled* feedback here; the Sustainiac even works for clean sustain, and on bass. I can hardly wait to try it on a Stick™.

Two footswitches let you choose whether the feedback occurs mostly at the fundamental or at the higher harmonics. There are five jacks: in, out, two effects loop jacks, and a jack for the transducer cord.

The transducer requires installation. Sustainiac comes with a bar magnet, which the manual advises attaching to the headstock with silicone adhesive. They claim that if done carefully, this kind of adhesive can be removed without taking off the finish. The transducer, a block about the size of a typical AC wall adapter, grabs on to the magnet.

The system's biggest weakness is the cord that trails from the headstock to the Sustainiac box (I routed the cord between my body and the guitar). Having to deal with the cord reduces the system's effectiveness for live use, since you have to be careful to not trip over or get tangled in it. The box isn't all that sturdy for the rigors of the road, either. Another consideration is that the Sustainiac can't turn a dead guitar into a live one—guitars with reasonable sustain characteristics are the best candidates for the Sustainiac (graphite necks seem to work particularly well).

Sustainiac

I think the studio is probably Sustainiac's true home. Controlled sustain has always been the Holy Grail for guitarists, and Sustainiac does just that. I thought about giving a three-star rating due to the clumsiness of the cord; still, a product that truly lets a rock and roll player get massive sounds at any volume is worthy of five stars, and the price is reasonable too. . . so four stars it is, with extra credit for providing us guitarists with some sounds we've all wanted for a long, long time.

(As we go to press, Maniac Music is phasing out two lower-priced models in their line, which do not have a built-in amp but require power from your guitar amp. The Model T unit reviewed above is being replaced by an improved version, Model B, which is functionally the same but claimed to be "built like a tank" and offer even better control over the sustain. The Model B will also retail for $299.90.) —*Craig Anderton*
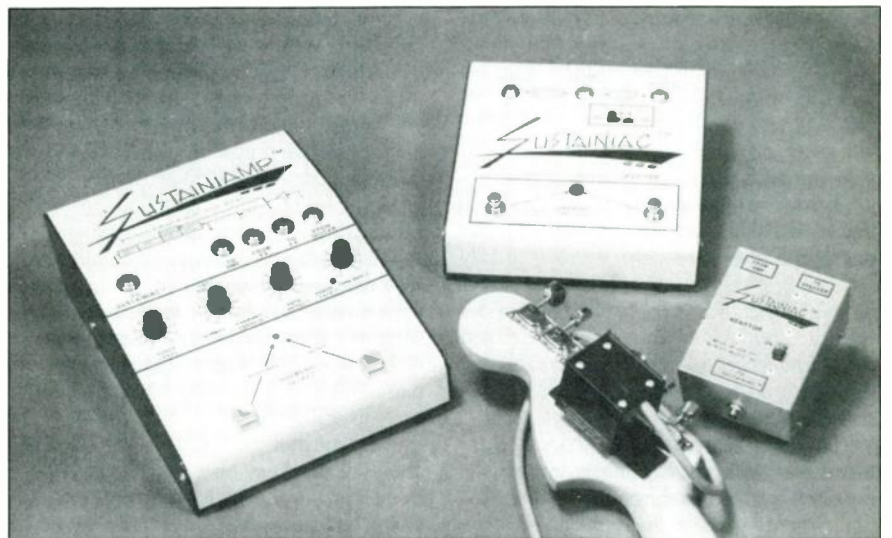
Maniac Music
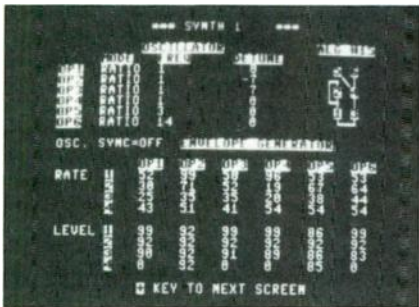6052 Guilford Ave.
Indianapolis, IN 46220
☎ 317 / 251-0470

Enharmonik Productions C-64 Patch Librarians
($35 ea.) ★★★
As the C-64 and Apple II become supplanted by newer personal computers, their roles are being relegated by many users to those of MIDI processors and patch data banks. In order to maintain a software market for these older machines, some companies are creating inexpensive, simplified programs, such as the Enharmonik Patch Librarians—available in Commodore 64/128-compatible versions for the Yamaha DX/TX (this version works with the DX1, 5, 7, 9, 21, 27, and 100, and the TX7 and TX2/8-16), and Oberheim Xpander/Matrix 12 synths; and also in a combined DX/TX and Xpander version ($45).

The Enharmonik Librarians work with Passport or Sequential interfaces. The Main Menu options (DX/TX version shown) are

Enharmonik C-64 Patch Librarians

screens to list a set of DX7 patch parameters); the printout is similarly clear.

The programs work well, though they will hang up if you instruct them to send or receive without having the appropriate instrument connected. All of the Enharmonik Librarians take some time to load; and although saving or loading banks of DX patches is pretty fast, saving or loading 100 Xpander patches takes *forever*.

The Librarians include patches—69 DX/TX and/or 48 Xpander—at no extra charge. (Note that the DX/TX patches are designed to work with six-operator synths, and aren't necessarily useable with the DX9, 21, 27, and 100.) The quality of the patches varies from good to not-

so-good; but they are free, and apparently original.

The performance/price ratio of the Enharmonik Librarians is high. Though the DX/TX version seems to be the best buy, given its compatibility with various DX/TX synths and its ability to display and print patch parameters. Still, the programs are a bit basic, and they won't read files created by other companies' librarians. They'll no doubt face some stiff competition in the budget-software market place.   —*Alan Gary Campbell*

Enharmonik Productions
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Sacramento, CA 95822
☎ 916 / 443-0373   EM

selected by entering the appropriate number and pressing "return." All librarian versions include Load from Disk, Save to Disk, Receive from Synth, Transmit to Synth, Print Bank, and Scratch Program on Disk functions; the DX/TX version also includes a Print Parameters function (which unfortunately is deleted in the combined DX/TX and Xpander version). You must exit the program to obtain a disk catalog—a major inconvenience.

The Receive and Transmit functions work only in banks, but the Load and Save functions can also handle individual patches. The current file name is displayed onscreen. You must type in the file name whenever you load or save banks, and you can't overwrite an existing file name (hence the Scratch function). Only one bank of patches may be loaded at a time. for the DX/TX version, this provides 32 patches; for the Xpander version, 100 patches (or multi-patches). There is no buffer area to allow patchswapping between banks. Disappointingly, the DX/TX and combined versions don't store or transmit TX function parameters.

The Print Bank function allows you to display or print out the patch names in a bank (the Xpander versions displays the 200 patches/multi-patches in groups of 50, but prints them all). With the DX/TX version (and the DX/TX part of the combined version), you can select and audition individual patches, without disturbing the synthesizer's memory; however, this function doesn't work with the DX21, 27, and 100. The Print Parameters function on the DX/TX version allows you to display or print out a patch: the display is understandable and very readable, which is a tremendous improvement over the cryptic alphanumeric gibberish provided by some C-64 librarians (though it does require three

It's one of the most useful signal processors for stage or studio—and you can easily build one yourself.

Five-Band Graphic Equalizer

BY BILL BERARDI

Octave band equalizers (i.e. equalizers that can boost or cut frequencies spaced one octave apart in the audio spectrum) are useful for compensating for room acoustics, notching out a nasty resonance, or fine tuning a specific sound. However, for adjusting the overall frequency contour, a five-band, two-octave equalizer is quite adequate and can be easier to use. Of course, five-band EQs are also less costly. So if you need more control over your sound than just bass and treble, but don't need quite the resolution of an octave equalizer, then a five-band EQ is for you. The EQ presented here allows about 22 dB total adjustment (cut and boost) at 40 Hz, 160 Hz, 640 Hz, 2.5 kHz, and 10 kHz. These bands are spaced evenly and symmetrically throughout the 20 Hz to 20 kHz audio band.

IC1a is a low noise buffer and pre-emphasis network. The highs are boosted in the first stage in order to achieve a good signal-to-noise ratio. The buffered output is broken up into the five frequency bands by five-bandpass filters built around IC1b, IC2a, IC2c, IC3a and IC3c. The two end bands, 40 Hz and 10 kHz, are wider than the others and are more like shelving-type equalizers. (With a low band shelving equalizer, also called a low pass filter, the high end response falls off as you go higher in frequency but the low end response remains constant; with a high band shelving equalizer, also called a high pass filter, the low end response falls off as you go lower but the high end response remains constant. This is different from a bandpass filter,

Bill Berardi is an electrical engineer who has been tinkering with electronic music circuits for eight years. Although his primary instrument is keyboard, he also enjoys playing guitar and designing guitar gadgets.

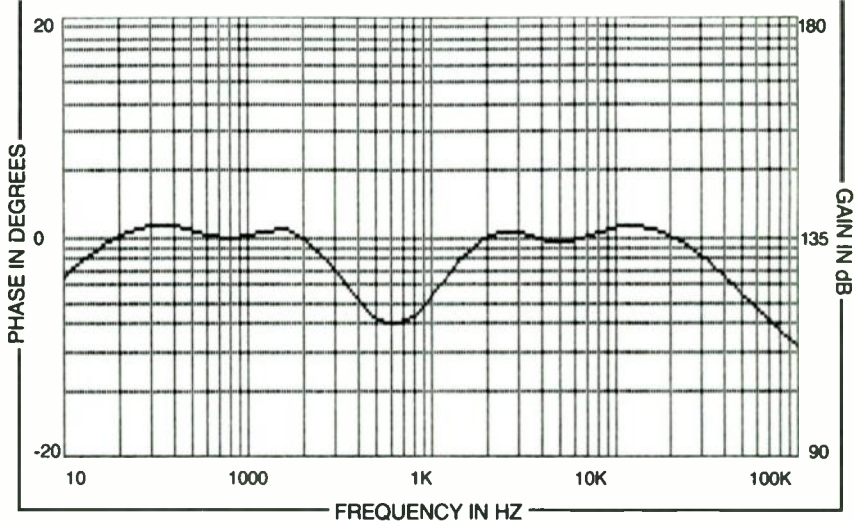
Fig. 1 *Schematic*

Fig. 1 *Set for flat response.*

FREQUENCY = 100,000
SLOPE = -3.802681 dB/Octave
GAIN = .3468934
PEAK GAIN = 1.117795
ANG = -137.82 9 -02
AT F = 160

ANALYSIS LIMITS
LOWEST FREQUENCY 10
HIGHEST FREQUENCY 100,000
LOWEST GAIN 0.1
HIGHEST GAIN 10
INPUT NODE NUMBER 3
OUTPUT NODE NUMBER 11

Fig. 2 *Center band, filter #3, 3dB down. Filters 1, 2, 4 and 5 are flat.*

FREQUENCY = 100,000
SLOPE = -4.583841 dB/Octave
GAIN = .3160901
PEAK GAIN = 1.135034
ANG = -128.87
AT F = 30 5

ANALYSIS LIMITS
LOWEST FREQUENCY 10
HIGHEST FREQUENCY 100,000
LOWEST GAIN 0.1
HIGHEST GAIN 10
INPUT NODE NUMBER 3
OUTPUT NODE NUMBER 11

where the response falls off on either side of the band peak.)

The output of each bandpass filter feeds a potentiometer (R24, R27, R30, R33 and R36 which are linear taper). Since a single pot simultaneously alters the input and feedback resistance, it provides a logarithmic-type response. Each stage can provide ± 12 dB of gain adjustment. Capacitors C14 to C18 restrict the op amp's high frequency response to pre-

Five-Band EQ Parts List

Resistors (¼ W, 5%)
R1	1M
R2	680 (680Ω)
R3	3k9 (3.9k)
R4, R8	16k
R5, R9	24k
R6, R10, R14,	
R18, R22, R25,	
R26, R28, R29,	
R31, R34, R35,	
R37, R38	3k3
R7, R23	75k
R12	22k
R13, R32	30k
R16	18k
R17	27k
R20	56k
R21	82k
R24, R27, R30,	
R33, R36	10k linear taper
R39	820
R40	4k7 (4.7k)
R41	1k

Capacitors (25 WVDC, 10%)
C1, C12	100n (0.1µF)
C2, C8, C9,	
C19, C22, C23	10n (0.01µF)
C3	1n (1000pF)
C4	470p
C5, C6	3n3 (0.0033µF)
C10, C11, C13	47n (0.047µF)
C14 - C18	10p
C20, C21	10µ

Semiconductors
IC1 - IC3	4136 Quad Op Amp
D1, D2	1N4001 or equivalent diode

For more information see "DataBank" in this issue.

EQ Basics

The range of human hearing typically spans 20 Hz to 20 kHz, although dogs and some humans can hear sounds above 20 kHz. One of the most basic ways a musician or engineer processes sound—whether an individual instrument or voice, or a fully orchestrated, multi-voice, multi-instrument piece of music—is with *equalization* or "EQ."

Simply, to equalize a sound is to select a specific band of frequencies within that 20 Hz to 20 kHz spectrum and boost or cut the amplitude of that band. The machine that does this is an equalizer; the name comes from the fact that equalizers were first used to even out, or make *equal*, the frequency response across the audio band.
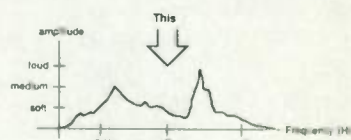
Typical waveform plot of a sound. When that sound is fed through an equalizer

added to this
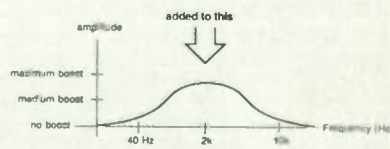
the response characteristics of the equalizer (shown here boosting a frequency band with a center frequency of 2 kHz)

equals this
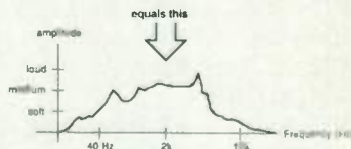
will change the waveform of the sound by boosting that particular band of frequencies

Equalizers come in two main flavors: graphic and parametric. A *graphic equalizer*, like the five-band EQ presented here, can cut and boost a number of frequency bands whose placement and width are pre-set and unalterable. A *parametric equalizer*, like a graphic EQ, can cut and boost frequency bands, but for *each* cut/boost control, the user can adjust not only the band *width* it affects, but the *center frequency* placement of that band.

The five-band graphic EQ has five cut/boost controls centered at 40 Hz, 160 Hz, 640 Hz, 2.5 kHz, and 10 kHz. Since the 20 Hz to 20 kHz spectrum

covers a little more than ten octaves, and the bands are evenly and symmetrically spaced, each of the five volume controls will cut or boost a band about two octaves wide. That's why this circuit is called a *five-band, two-octave equalizer.*

With parametric equalizers, each cut/boost control is linked to two other controls. (One machine will typically have from three to five of these sets of three controls.) With one of these other controls, the user can *sweep* the 20 Hz to 20 kHz spectrum to set the cut/boost at any frequency (referred to as a *center frequency*) in the spectrum. With the other control, the user sets the *width* (referred to as "Q") of the frequency band around that center frequency. This width can go from as narrow as 0.03 octave to as wide as two octaves. Since the width and placement of its frequency bands are so flexible, it is more accurate to think of the parametric as having *modules* rather than fixed *bands.*   **continued**
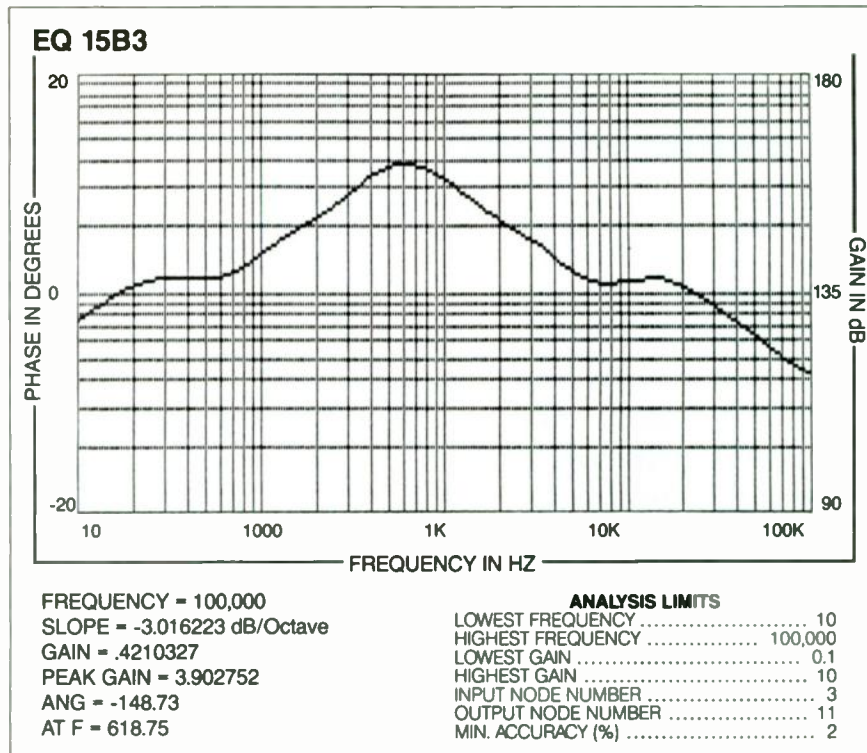

EQ 15B3

FREQUENCY = 100,000
SLOPE = -3.016223 dB/Octave
GAIN = .4210327
PEAK GAIN = 3.902752
ANG = -148.73
AT F = 618.75

ANALYSIS LIMITS

LOWEST FREQUENCY	10
HIGHEST FREQUENCY	100,000
LOWEST GAIN	0.1
HIGHEST GAIN	10
INPUT NODE NUMBER	3
OUTPUT NODE NUMBER	11
MIN. ACCURACY (%)	2

Fig. 3 *Center band, filter #3, with 3dB gain. Filters 1, 2, 4 and 5 are flat.*

5-Band EQ—from page 27

vent the possibility of oscillation. Keep these capacitors close to their respective chips. A portion of the original signal also feeds each gain adjustment stage via resistors R7, R11, R15, R19 and R23. Since each bandpass filter is built around an inverting amplifier, the original signal is subtracted from the filtered signal. This minimizes the effect of one frequency band spilling over into the next.

The five outputs are combined and de-emphasized by the last stage, IC1d. The circuit should be powered from a split supply in the range of ± 9V to ± 15V. Although each gain stage provides ± 12 dB of adjustment, the actual amount of cut is limited to about 11 dB by the resid-

Basics continued

The parametric is useful for extreme and/or unusual frequency response tailoring. For example, if a signal has some hum on it, the parametric can dial directly in to 60 Hz and put in a steep notch to cut the hum while leaving the rest of the sound virtually unaffected. And while the parametric can function as a graphic, the graphic cannot do all a parametric can. The parametric is clearly more flexible than the graphic EQ, but this also means it's more expensive and requires more time to adjust.

Quasi-parametrics, often found on the input modules of mixing boards, can cut, boost and sweep, but lack the Q control of the fully parametric equalizer. —*Tim Tully*
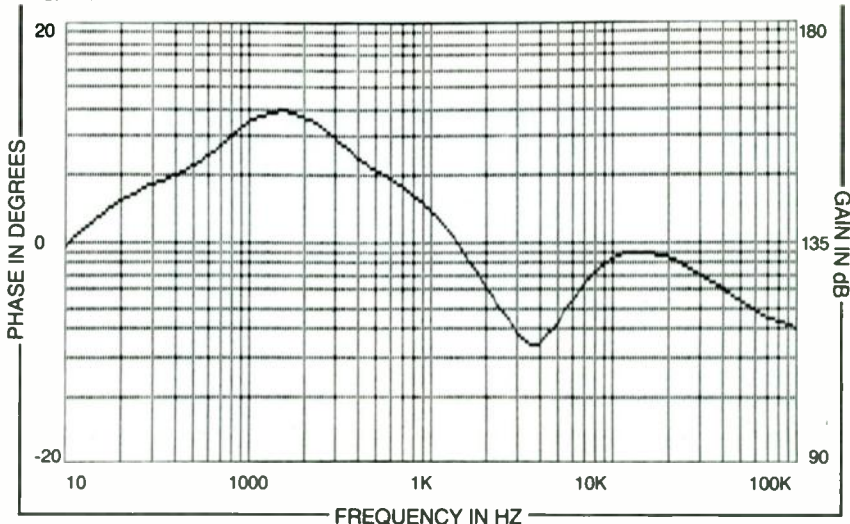

EQ 15B234
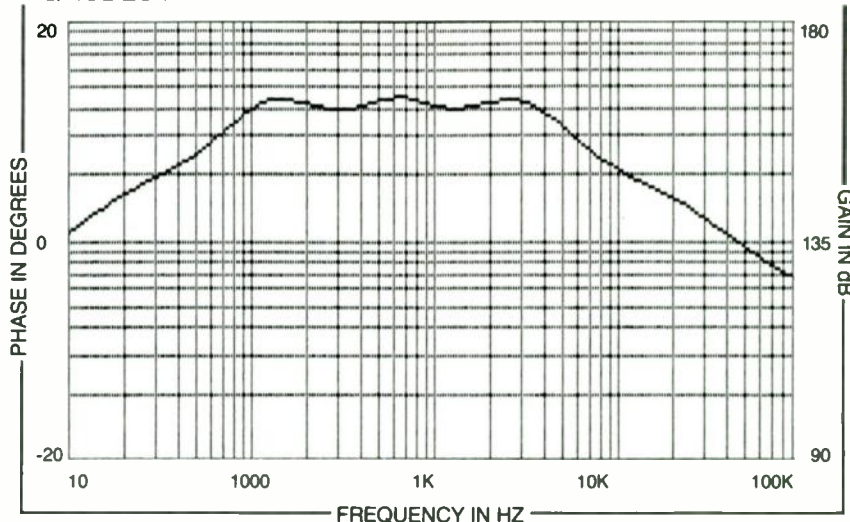
FREQUENCY = 100,000
SLOPE = -2.547367 dB/Octave
GAIN = .6590391
PEAK GAIN = 4.562054
ANG = -161.32 9
AT F = 672.5

ANALYSIS LIMITS
LOWEST FREQUENCY 10
HIGHEST FREQUENCY 100,000
LOWEST GAIN 0.1
HIGHEST GAIN 10

Fig. 4 *Filter bands 2, 3 and 4 with 12dB boost (gain). Filters 1 and 5 are flat.*

EQ 21

FREQUENCY = 100,000
SLOPE = -1.793678 dB/Octave
GAIN = .4036915
PEAK GAIN = 3.948085
ANG = -137.9
AT F = 153.75

ANALYSIS LIMITS
LOWEST FREQUENCY 10
HIGHEST FREQUENCY 100,000
LOWEST GAIN 0.1
HIGHEST GAIN 10
INPUT NODE NUMBER 1
OUTPUT NODE NUMBER 19

Fig. 5 *Filter band #2 with 12dB boost. Filter band #4 with 10dB cut. Filters 1, 3 and 5 are flat.*

EQ 15B24

FREQUENCY = 100,000
SLOPE = -2.974109 dB/Octave
GAIN = .5602037
PEAK GAIN = 3.871085
ANG = -153.77
AT F = 152.5

ANALYSIS LIMITS
LOWEST FREQUENCY 10
HIGHEST FREQUENCY 100,000
LOWEST GAIN 0.1
HIGHEST GAIN 10
INPUT NODE NUMBER 3
OUTPUT NODE NUMBER 11

Fig. 6 Filter bands 2 and 4 with 12dB boost. Filters 1, 3 and 5 are flat.

EQ 15B15

FREQUENCY = 100,000
SLOPE = -5.074709 dB/Octave
GAIN = 1.084725
PEAK GAIN = 4.129697
ANG = -128.2
AT F = 13210

ANALYSIS LIMITS
LOWEST FREQUENCY 10
HIGHEST FREQUENCY 100,000
LOWEST GAIN 0.1
HIGHEST GAIN 10
INPUT NODE NUMBER 3
OUTPUT NODE NUMBER 11

Fig. 7 Filter bands 1 and 5 with 12dB boost. Filters 2, 3 and 4 are flat.

"**W**hether for recording or on-stage use, a graphic equalizer can be a real 'workhorse' of a signal processor"

ual spillover from the adjacent bands. This could be improved by increasing the Q (resonance) of the bandpass filters or by subtracting more of the original signal, but this would cause more pass-band ripple. The circuit here was chosen as a "best compromise" design. Remember, equalizers cannot do outrageous frequency corrections—the actual frequency response will not be exactly the same as your control settings.

Fig. 1 shows the overall frequency response when all the controls are set flat. Fig. 2 shows the center band cut, Fig. 3 the center band boosted, Fig. 4 bands 2, 3 and 4 boosted, and Fig. 5 band 2 boosted and band 4 cut. These all agree quite well with what you would expect. Fig. 6 shows bands 2 and 4 boosted and Fig. 7 bands 1 and 5 boosted. Here we see some deviation from the ideal case. Construction technique is not particularly critical, but all the normal rules apply. Any circuit of this type is sensitive to component variations, so use the exact component values—5% resistors and 10% low temperature coefficient capacitors (polypropylene or polystyrene) for C3 to C13 are preferred.

Enough has already been written about how to use equalizers. Just remember, with only five bands you can't make microscopic changes in the texture of the sound. Use this EQ the same way that you use bass and treble controls—to adjust the overall balance of the sound. Whether for recording or on-stage use, a graphic equalizer can be a real "workhorse" of a signal processor. **EM**

Mod

> Disappointed because you wanted stereo outputs and didn't wait to buy a Mirage DSK? Don't worry—you can retrofit your existing Mirage for stereo outputs and enjoy the benefits of a fuller, richer sound.

Stereo Mirage—The Sequel

BY BOB YANNES
WITH TOM METCALF

We have received a number of inquiries in response to Don Slepian's January 1987 **EM** article on modifying the Ensoniq Mirage for stereo output. Ensoniq recently introduced the Mirage-DSK, which incorporates a *different* stereo output circuit that we at Ensoniq prefer as a modification to the Mirage. As a service to Mirage owners who want to modify their units, we would like to provide the information necessary to install our circuit. *Please note that this article is provided for informational purposes only.* The circuit does not represent a factory-authorized modification (such as the original noise reduction and bandwidth modification kit for the Mirage). Unauthorized modifications such as this, and any modifications performed by unauthorized service personnel *will* void your warranty!

POTENTIAL CIRCUIT PROBLEMS

We recommend this modification over Don's for a number of reasons. The circuit presented in his article uses passive mixing through resistors. This results in significant signal loss at the output, due to the voltage divider created by the resistors. In addition, the high output impedance ($10\ k\Omega$) will interact with the im-

***Bob Yannes**, who wrote the article, was a driving force in the design of the VIC 20 and Commodore 64 computers and designed the C-64's sound chip. After leaving Commodore, he helped found Ensoniq and develop the Q-chip, the heart of the Ensoniq Mirage.* ***Tom Metcalf**, who figured out the resistor values and wired the prototype, developed most of the Ensoniq sound library and is an expert on sampling with the Mirage.*
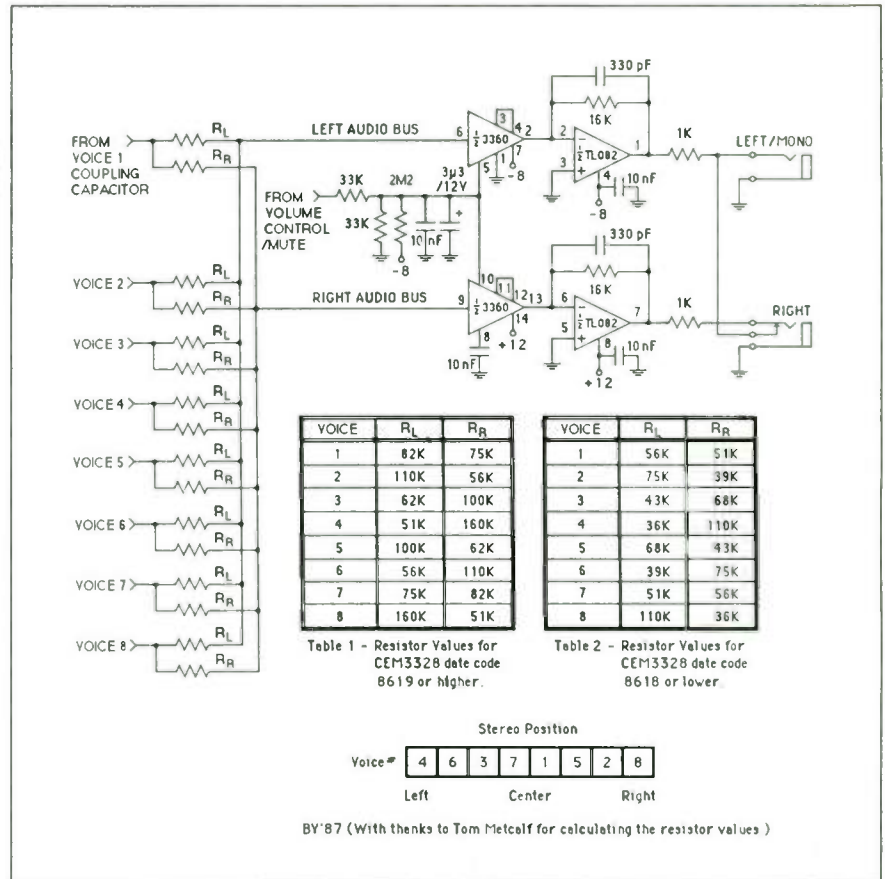
Fig. 1 *Mirage Stereo Output*

VOICE	R_L	R_R
1	82K	75K
2	110K	56K
3	62K	100K
4	51K	160K
5	100K	62K
6	56K	110K
7	75K	82K
8	160K	51K

Table 1 – Resistor Values for CEM3328 date code 8619 or higher.

VOICE	R_L	R_R
1	56K	51K
2	75K	39K
3	43K	68K
4	36K	110K
5	68K	43K
6	39K	75K
7	51K	56K
8	110K	36K

Table 2 – Resistor Values for CEM3328 date code 8618 or lower.

Stereo Position

Voice #	4	6	3	7	1	5	2	8

Left — Center — Right

BY'87 (With thanks to Tom Metcalf for calculating the resistor values.)

pedance of line inputs, producing different levels when plugged into different mixers and other inputs. This makes it difficult to achieve the best possible signal-to-noise ratio.

The circuit includes no output buffering. Ideally, a signal output should be a low impedance source with output protection circuitry to prevent damage to internal components in the event that improper voltages are applied. Don's circuit taps the signal directly from the Curtis Electromusic CEM3328 filters, leaving

the mixing resistors as the only output protection. The mixing resistor forms an output impedance higher than we feel is desirable, and also, the filters have no special output protection circuitry and potentially can be damaged. We have seen printed circuit boards on which the output traces were vaporized by high voltage occurring when musicians plugged their keyboards into improperly grounded audio systems. Think what that voltage could do to the filters—all eight of them!

Buffering also suppresses the CEM-3328 filter's tendency to oscillate under certain capacitive loads, which can produce birdies at the output, and there is a possibility that outside signals could affect the cutoff frequency of the filter (as the filter output forms one of the filter poles via local feedback).

Because the filters have a very sensitive differential input, it is not uncommon for a filter output to be offset to some DC level other than zero, which could easily cause problems with, or damage to, non-capacitively coupled audio systems. In the Mirage, each filter AC couples into the remaining circuitry to prevent any such DC offsets, but in Don's modification, the outputs are tapped directly off each filter *without* coupling capacitors.

Beyond these reservations, we also feel that the stereo outputs are of limited value. The circuit creates a 4/4 split, where the four voices of each output are assigned to the same place in the stereo field: either hard left or hard right. This does not produce a very natural stereo image and is useful primarily as an effect.

Neither can we recommend modifying the Mirage for eight independent outputs. It is a relatively simple modification, but because the Mirage voice assignment is random, there is no way to control or predict which sound will come out where. Since there is no way to assign a certain sound to a certain output, the independent outputs can't be used to process sounds individually and this is bound to disappoint many users. Finally, we feel that not only will the lack of output muting and volume control in Don's modification be annoying, but (as mentioned in the article) when the filters tune up, they produce full-level sine waves in the 1 to 2 kHz range. This is more than enough to blow out your average tweeter if you forget to turn down your amp when booting up.

If for some reason you prefer to use Don's circuit, we strongly recommend you use the active output summation he talks about towards the end of the article. The summing op amps will provide the desired output drive characteristics and buffering. If you don't add these op amps, you may regret it later if you run into any of the problems mentioned above.

AN ALTERNATIVE
The circuit we've developed here is designed to pan each voice to a different spot in the stereo field. This produces a

full stereo field for ensemble sounds such as strings, brass or vocals, and adds animation to percussion sounds by moving them around the stereo field. On some sounds, such as piano, it sounds unnatural to hear different notes coming from different directions, but this can be overcome simply by running the output in mono for these sounds or adjusting the pan pots on your mixer (note also that the ping-pong effect is not as exaggerated as it is in Don's circuit).

As shown in **Fig. 1**, each filter's audio output is tapped *after* the AC coupling capacitor, thus eliminating any DC offsets. This signal goes to two resistors, one feeding the left audio bus and the other feeding the right. Varying the value of these resistors with respect to each other can position the filter signal anywhere in the stereo field. If the resistor feeding the right bus, for example, is lower in value than the resistor feeding the left bus, more signal will go to the right, making it louder and subjectively positioning the signal further to the right in the stereo field.

The left and right buses feed into a Curtis Electromusic CEM3360 Dual Voltage-Controlled Amplifier. This chip provides two low-noise VCAs with current inputs and current outputs. One VCA sums the left bus resistors and provides the left output, while the other sums the right bus resistors and provides the right output. The current outputs of the VCAs are converted to voltages and buffered by the TL082 op amps. The low-impedance VCA input allows active summation of the signals, which eliminates any signal loss or undesirable interactions that can occur with passive mixing. The op amp outputs are very low impedance, with built-in output protection. The 1 kΩ series resistor at the output of each op amp provides additional protection in the event of severe output abuse.

FULLER STEREO

The VCAs provide stereo volume control from the single volume control pot on the front panel. In order to make this work properly, jumper across resistor R20 (3 kΩ) and resistor R21 (7.5 kΩ). Remove C84 (10 nF or 0.01 μF) and C86 (3μ3 or 3.3 μF). Both resistors are located on the far left of the Mirage main board near the ribbon cable connectors. The capacitors are on the far upper right corner, to the right of U48 (NE570). Disconnect pin 1 of U48 (NE570) from the board and con-

nect pin 1 to pin 5 of U48. This modification defeats the muting and volume control at this output and lets you use the original audio output jack as a monitor when sampling.

Since the muting circuit is incorporated into the volume control, the muting function occurs automatically on the new stereo outputs. The outputs will mute whenever the Mirage loads from disk, tunes its filters, or samples. Regarding the outputs, you have two options. Plugging into the left and right outputs gives a full stereo spread; plugging into only the left output picks up a mono mix. This mono mix is available because the switching jack used for the right output routes the right signal over to the left output when nothing is plugged in the right output jack.

The tables given in **Fig. 1** list the values for each voice's left and right resistors. These values place each voice at a different spot in the stereo field. The values are chosen carefully to produce equal apparent volume for each voice. Note that two tables are provided for use with different CEM3328 filters. Look carefully at the date code printed on the top of each filter chip to determine which values to use. If your filters have a date code of "8619" or higher, use the first table. If your filters have a date code of "8618" or lower, use the second table (in case you're wondering, "8619" means "the 19th week of 1986"). If you don't use the proper table, the stereo outputs will be distorted or weak.

WARRANTY VOIDING AND OTHER HAZARDS

Before you tear into your Mirage, remember: once you open the lid, you void your warranty!

Presumably that didn't scare you since you're still reading this and probably opening the lid anyhow. That being the case, please remember this: *there are lethal voltages present in the power supply area of the Mirage. Avoid killing yourself! If you have any doubts about your ability to perform this modification, don't do it.*

Never work inside the Mirage with the power cord plugged in. When checking your work with the Mirage under power, stay away from the power supply and/or cover that area with an insulator (see this month's article on safety in the workplace by Alan Campbell--Ed.).

HERE'S HOW

When assembling the circuit, we recom-

**Typical Filter PC layout
(1 of 8)**

Output Coupling Capacitor
(3µ3 Electrolytic)

U52,53,57,58,62,63,67,68   -8V

CEM3328
DATE CODE

Tap signal off of the right (–) side
of these capacitors. The capacitors
are: Voice 1-C191
      Voice 2-C182
      Voice 3-C159
      Voice 4-C151
      Voice 5-C131
      Voice 6-C118
      Voice 7-C124
      Voice 8-C111

Analog Ground

+12V   Pin 1

U48

NE570

Standard Mirage Output Circuit

Tap off Volume Control/Mute
from this feedthrough

Remove C84 and C86

Disconnect Pin 1 from the board and jumper Pin 1 to Pin 5

Fig. 2 *Mirage Board Layout*

mend the use of a printed circuit board if possible. Perf board is acceptable, although you may want to use the copper-clad variety and connect the copper to analog ground to act as a ground plane. This will minimize noise pickup. The output of each filter's coupling capacitor should be picked up at the right ("–") side of the capacitor in each filter section; the power supply connections (+12V, -8V, and Analog Ground) can be picked up at any of the filters (see **Fig. 2**). The volume control voltage can also be picked up as shown in **Fig. 2**. When installing the output jacks, be sure the jack ground does not connect with the metal case, as this will create a ground loop that could cause hum and noise in the output and sampling input. Concerning parts availability, the CEM3360 VCA is available from Curtis Electromusic Specialties, 110 Highland Ave., Los Gatos, CA 95030 (tel.

408 / 395-3350). The TL082 is available from Radio Shack or almost any electronics parts company.

After assembling the circuit, check your wiring. Using IC sockets for the CEM3360 and TL082 will minimize the chance of damaging these components if there is a wiring error. Use an Ohmmeter to verify that neither the +12 nor -8 Volt power supplies are shorted together or to ground.

THE BOOT

Turn down the volume on your amplifier, patch the old Mirage output jack into your audio system, and boot the Mirage. Do not turn up your amp until the Mirage is fully running and showing "21" in the display. The Mirage should operate normally (however, the volume slider will have no effect). Verify that all voices are working from the regular output jack.

If everything is working normally, measure the power supply pins of the CEM3360 and TL082 sockets. If the + and - supplies read properly on these pins, power down the Mirage and insert these ICs. Turn down the volume on your amplifier and plug your audio system into the left/mono output jack. Boot the Mirage and set the Mirage volume slider to max. While playing notes, slowly bring up your amplifier volume. If all is working properly, you should hear the normal mono output of the Mirage. Verify that each voice is working and all are approximately the same volume. Next, reduce your amplifier volume, plug a second cord into the right output jack, and plug the other end of this cord into the other channel of your amplifier. Carefully bring up the amplifier volume and play eight different keys slowly and separately. The sound from each key should come from a different spot in the stereo field. Don't hit the same key twice, or restrike a key before it has decayed, as the Mirage voice assignment algorithm will retrigger the same voice. Check that the Mirage volume slider alters the volume of the stereo outputs.

If you are using a mixing board between the Mirage and your amp, plug the left/mono output into one channel and the right output into another channel. Make sure all level, EQ, and phase settings are the same for both channels. Pan the left/mono input fully left and the right input fully right for maximum stereo effect. You can experiment later with the pan controls to constrain the stereo field on different sounds.

To get the full stereo effect, load in Strings or Vocals and play big chords. Stereo outputs greatly enhance the realism of sampled sounds by spreading them convincingly across acoustic space. Hearing a full string section coming from a single spot in space is a dead giveaway that you're hearing a recording (which is, after all, what a sampled sound is). Percussion sounds respond dramatically as different voices jump out at you from different directions. In fact, you'll find that the stereo outputs enhance most of the over 300 sounds in the Ensoniq library as well as the hundreds of Mirage sounds available from other sound designers such as KMUSE. If you've been looking to get more out of your Mirage, we feel this modification is one of the best places to start. **EM**

Fostex Presents a Series of Timely Investments:

4010 SMPTE Time Code Generator/Reader has all the functions you'll ever need because it's completely software based. As in all high tech digital electronics today, there's a switching power supply. And socketed EPROMs mean your hardware will never be obsolete. Plus, the 4010 has a serial data port on the rear panel which allows communication with (and control by) computers.

4030 Synchronizer with switching power supply and **4035** Controller for all major brands and models of tape recorders—both audio and video. Control up to one master and three slaves, or slave as many transports as you wish, just by adding a 4030 for each slave. A serial interface port allows communication with personal computers to run F.A.M.E., Fostex Automated Media Editing. SMPTE time code based, it works with 24, 25, 30 fps and drop frame, and features resolution to 1/100 of a frame, selectable pre-roll up to 99 seconds, 10 position auto-locate and **programmable, automatic punch in/out with rehearse function.**

4050 Auto Locator is a full function remote control unit for Fostex Models 80, 20 and all E-Series recorders. In addition to tempo control, you can locate to the measure bar and beat, thus the 4050 is the first autolocator to think musically. Up to ten cue points can be programmed and you can punch in and out automatically. The 4050 is also a MIDI Synchronizer, featuring a SMPTE/EBU Time Code generator/reader—all four formats. Any MIDI instrument can be synchronized to this most accurate timing reference. You can also simulate the running of the recorder in order to have the SMPTE/EBU code run MIDI instruments only. There's even a serial communication port which lets you use a personal computer.

Merging Technologies

FOSTEX

No matter what anybody says, you can take it with you...here's the low-down on a small, inexpensive, portable headphone amp.

Pocket Rockit, A Personal Amplifier

BY JIM FIORE

There I was, on the shore of a lovely Adirondack lake, bathed in the cool light of a mountain sunset...playing my bass guitar. Yes, thanks to a pocket-sized personal amplifier, designed to drive Walkman-style headphones from a single 9-volt battery, the full sound of my electric bass was right there with me in the sylvan mountain setting.

Since the "Pocket Rockit" (catchy name, eh?) seemed to work so well, I thought I'd share the design with other folks who like to play their instruments in odd surroundings, or practice, for example, at 3 a.m. but find that inconsiderate neighbors, policemen, or vigilante groups interrupt much too regularly. The Pocket Rockit is designed for guitar players, keyboardists, electronic drummers, and anyone who needs to monitor the output of

When not harassing his Electrical Engineering Tech. students at Mohawk Valley Community College in Utica, N.Y., Mr. Fiore spends vast amounts of time in his home studio communing with small pieces of silicon and lots of wire. Jim plays drums in deference to either Primal Scream therapy or Est. His hobbies include driving sideways down country roads, complaining about the high cost of yogurt, and Zelda.
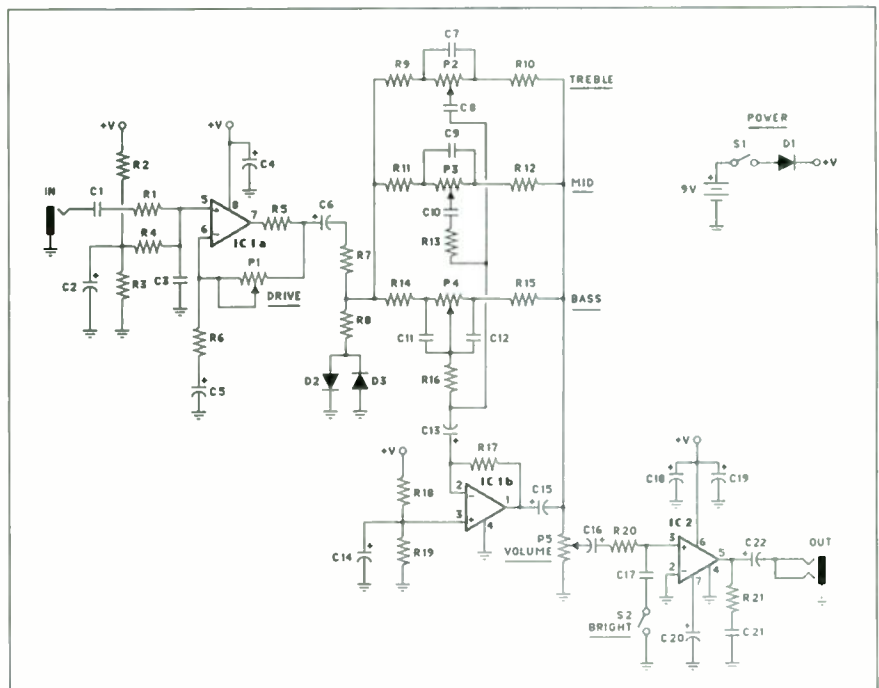
Fig. 1 *Schematic*

an electronic instrument via headphones or through small loudspeakers. Functionally, the unit has volume and overdrive (distortion) controls, a "bright" switch, and bass, mid, and treble EQ controls.

THE SCHEMATIC

Referring to the Rockit's schematic (**Fig. 1**), the first stage is a preamp/distortion block centered around op amp IC1a. An LF353 offers low battery drain, although an NE5532 could be used for lower noise. To eliminate the need for a bipolar power supply, R2, R3 and R4 bias the LF353 for single-ended operation. R4 and C2 decouple the input from the power supply noise while C1 and C6 are the input and output coupling capacitors, respectively.

C5 restricts frequency response at sub-audio frequencies. C3 and R1 minimize RFI (radio frequency interference). C4 bypasses the op amp power supply.

The gain of this stage equals (1+P1)/R6 and is adjustable from unity upwards toward 68 (about 36 dB). The load for this stage is non-linear and consists of R7, R8, D2 and D3. At low signal levels, the signal is passed on with no change in waveshape, but at higher levels, the two diodes kick in and clip the signal. However, the clipping will tend to be somewhat softer than that produced by normal diode clippers thanks to R5, R7, and R8. These resistors tend to "buffer" the drastic change in diode impedance once conduction begins. Heavy Metal

users fear not, as high overdrive settings will still produce the classic, full bore "transistor distortion."

Stage two, built around op amp IC1b, provides the bass, mid, and treble EQ (equalization) functions. This circuit is a typical inverting configuration with a frequency selective feedback network. R9, R10, P2, C7 and C8 produce high frequency shelving. R11, R12, R13, P3, C9 and C10 comprise a 2 kHz midrange peak/dip function, and R14, R15, R16, P4, C11, and C12 produce low frequency shelving. R17-R19 and C13-C15 provide

Fig. 4 *Wiring: How to avoid the counter clockwise blues*

biasing and coupling. P5 is the volume control. Like P1, P5 must be a log (audio) taper device for proper "feel."

The final stage consists of the LM386 power op amp. It is bypassed with C18 and C19 while C20 maximizes the PSRR (power supply rejection ratio). The "bright" switch connects in a single pole lag network (low pass 6 dB/octave filter) hinged at 3 kHz. The normal position is with the switch closed. This will echo the response of typical 12- and 15-inch loudspeakers, which start to roll off once they reach their mass controlled region (this usually occurs between 2 and 4 kHz). Opening the switch will produce true flat response, which is subjectively bright when compared to a

typical guitar or bass amplifier/loudspeaker setup. (Keyboardists, drummers and others who normally play through full range systems with a flat top end should consider this to be a "dull" switch instead— actually a more appropriate name, since it was designed with rock guitarists in mind. Har-de-har-har.)

Now that I've alienated several readers and the editor (just remember who signs your check, Jim—Ed.), let's continue. R21 and C21 stabilize the LM386 at high frequencies and C22 serves as the output coupling capacitor. Finally, at the output we connect the load. Normally this will be a pair of mini Walkman-style headphones. Note that a stereo output

Fig. 2 *Foil side of PC board*

Fig. 3 *Stuffing Guide*
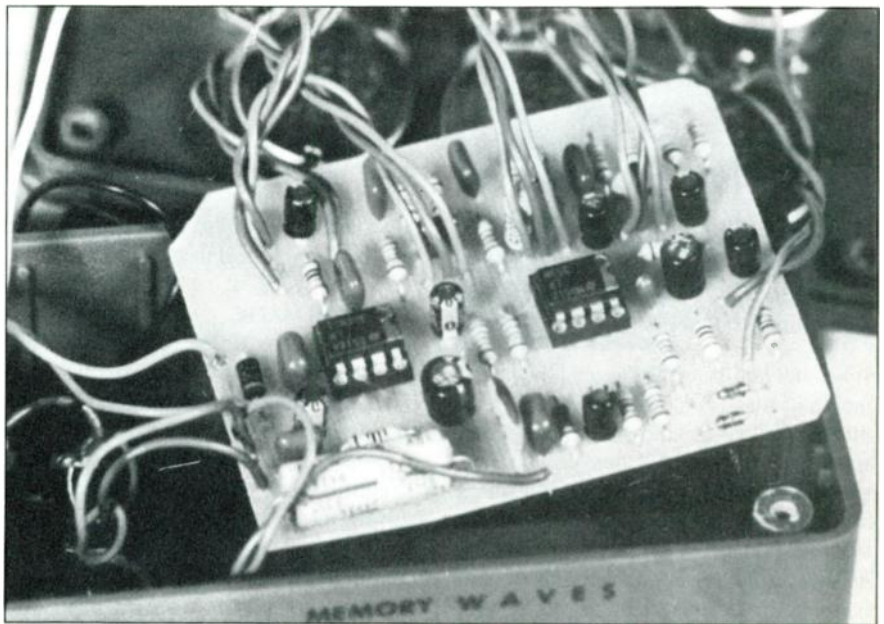
Fig. 5 *Completed PC Board*

plug is required to send the instrument signal (in mono, of course) to both ears. (It's also possible to connect a small loudspeaker here instead; for best performance make sure that the load does not drop below 8 ohms.) A pair of mini headphones connected in mono will usually produce around 25 ohms. (The load impedance is usually listed in the specifications for a particular loudspeaker; otherwise, those who have access to sophisticated electronics devices can measure load impedances with an impedance meter.)

The maximum output power is about ½ watt, which won't make a loudspeaker scream but can produce *very high* levels with headphones—so **be careful**. You can maximize battery life by keeping the volume low and using high impedance headphones (like the Sennheiser HD 414). Alternately, you can power the circuit from a 9-volt battery eliminator. If batteries are really important, you can get longer life by using six 1.5-volt AA batteries instead of the single 9-volt unit. The AA's will come out cheaper in the long run, particularly if you opt for the
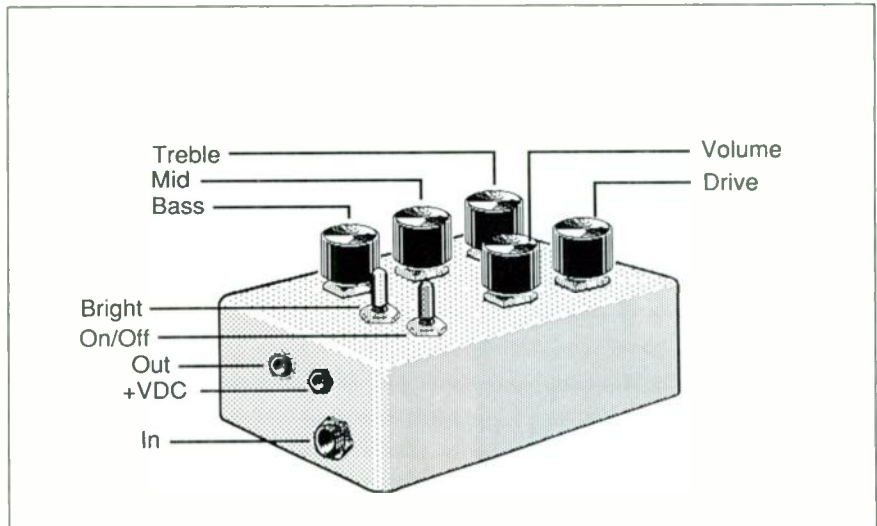
Fig. 6 *Suggested layout for complete project*

rechargeable Ni-Cad variety. They do, however, take up more space.

Fig. 2 shows the PC board (foil side) artwork; Fig. 3 shows the component layout. When assembling, make sure that you obey the polarity markings for the caps and diodes. Also make sure that the

ICs are oriented correctly. Miswiring the pots can produce some strange results; for all pots requiring three leads, the wiper arm should connect to the center pad on the PC board. The other wires are connected as shown in Fig. 4. Connecting them backwards will make your vol-

ume, drive, and other controls *decrease* when you rotate the pot clockwise, so pay attention. **Fig. 5** shows a completed board and pot assembly, and **Fig. 6** is a drawing of the completed unit.

FINDING PARTS

If you can't find parts locally, there are several good mail order sources including Digi-Key Electronics, PO Box 677, Thief River Falls, MN 56701; Jameco Electronics, 1355 Shoreway Rd., Belmont, CA 94002; and JDR Micro Devices, 1224 S. Bascom Ave., San Jose, CA 95128. If you use good, standard soldering techniques, IC sockets and the like, your Pocket Rockit should give you no trouble during assembly and test. I hope you find this circuit useful, and that it gives you a lifetime of service. **EM**

Parts List

Resistors (All resistors ⅛ W, 5% carbon film)

R1, R11, R12	4k7 (4.7K)
R2, R3, R18, R19	220k
R4	470k
R5	220
R6	1k
R7, R8	68 (68Ω)
R9, R10	3k3 (3.3K)
R13	6k8 (6.8K)
R14, R15	5k6 (5.6K)
R16	27k
R17	1 M
R20	10k
R21	10 (10μ)
P1	100k log potentiometer
P2, P3, P4	100k linear potentiometer
P5	1k log potentiometer

Capacitors (15 working volts DC or greater)

C1	100n mylar (0.1)
C2, C13-C16, C19	10μ electrolytic
C3	68p disc
C4	1μ5 tantalum
C5	22μ electrolytic
C6	33μ electrolytic
C7	560p disc
C8	1n5 mylar (0.0015)
C9	2n2 mylar (0.0022)
C10	4n7 mylar (0.0047)
C11, C12	22n mylar (0.022)
C17	6n8 mylar (0.0068)
C18	100n mylar (0.1)
C20	47μ electrolytic
C21	47n 50V mylar (0.047)
C22	470μ electrolytic

Semiconductors

IC1	LF353 dual op amp (National Semi)
IC2	LM386 power amp (National Semi)
D1	1N4001 rectifier
D2, D3	1N914 or equiv. diode

Mechanical Parts

S1, S2	SPST switch
Misc.	Knobs, case, hookup wire, IC sockets, ¼" and ⅛" jacks (mono and stereo), battery clip, etc.

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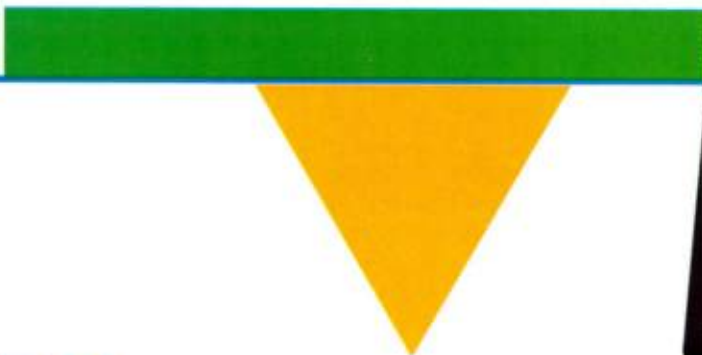
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How the ESQ-1 can sequence, display, control and sound better than any other synth in its class...

Let's start with sound. After all, sound is your first criteria for any musical instrument.

In his review of the ESQ-1, Peter Mengaziol of GUITAR WORLD wrote, "The ESQ-1's sound combines the flexibility and analog warmth of the Oberheim Matrix-6, the crisp ringing tones of a DX-7, the realism of a sampler, the lushness of a Korg DW-8000 and polytimbral capacity of the Casio CZ-1".

The ESQ-1 has 32 different waveforms — from analog and FM-type to multi-sampled and additive synthesis waves. And each of the ESQ-1's 8 voices is a wall of sound that's 3 oscillators thick.

Displays of intelligence
Next, there's simplicity. Synthesizer complexity has gotten out of hand recently. In fact, just try saying "linear arithmetic algorithmic operators" 3 times.

The ESQ-1 is a breeze to program. All the information you need is spelled out on the 80-character lighted display. And the clearly-written manual makes it easy to pick up the details.

Paul Wiffen of MUSIC TECHNOLOGY was pleasantly surprised with the ESQ-1's simplicity. "Unlike so many modern synths, the new Ensoniq has a programming layout that's so easy to get to grips with, it almost *invites* you to delve deeper."

Split ends
A split keyboard is a great performance feature. The ESQ-1 lets you split the keyboard anywhere you want and save the sounds and split point as one program.

Page by page the 80-character ESQ-1 display makes programming and sequencing a breeze.

You can also layer sounds across the entire keyboard or on either or both sides of a keyboard split with the Split/Layer function. Because the ESQ-1 has dynamic voice assignment, all 8 voices are available wherever and whenever you need them.

The multi-track, multi-timbral, multi-mode marvel
It's not easy to find a multi-track sequencer that's both powerful and easy-to-use. Seek no more. The ESQ-1 sequencer is loaded with features: 8 polyphonic tracks, multi-timbral, punch-in/punch-out, quantization, step-editing, auto-locate, mixdown levels — to mention just a few.

In summing up the sequencer, Peter Mengaziol suggests, "The sequencer alone compares favorably with stand-alone units that cost as much as, if not more than, the

Oberheim Matrix-6, Korg and Casio are trademarks of ECC Development Corp., Korg USA and Casio, Inc. respectively

"At $1395, it would be a bargain just for the sounds it makes, but when you consider the sequencer as well, this thing is a steal!"—Jim Johnson—KEYBOARDS, COMPUTERS & SOFTWARE Oct. '86

ESQ-1 itself. Go to the store and ask to hear the DEMO 1, 2 and 3 songs to appreciate what the sequencer section can do using the internal voices alone . . . it's scary." And if you're really brave, you can connect as many as 8 MIDI units to the ESQ sequencer and control them all independently.

With your ESQ, everything's under control
If you need a MIDI master keyboard, you'll be pleased to find how easily an ESQ-1 can get your entire MIDI setup under control.

For instance, you can send MIDI channel and patch changes to 8 different MIDI units one at a time or simultaneously at the push of one button.

The unique MIDI Overflow Mode gives your ESQ room to grow. Just add the new ESQ-M synthesizer module ($995 US) and you've got a powerful 16-voice synth. While you're thinking of expansion, there's the Mirage® Multi-Sampler which can add hundreds of sampled sounds to your setup or the new Ensoniq Sampled Piano Module ($895 US) with instant access to 12 sampled keyboard and bass sounds.

MIDI Modules for your growing ESQ-1 studio. The ESQ-M Digital Wave Synthesizer Module, the Mirage Digital Multi-Sampler, the Ensoniq Sampled Piano Module.

Unanimous decision
England's most respected music paper, MELODY MAKER, named the ESQ-1 as "Synth of the Year". In fact, all the major international music magazines who've reviewed the ESQ-1 have come to much the same conclusion, so let's give them the last word.

"Every once in a while, a product comes along that begs to be raved about. Ensoniq's ESQ-1 synth is just such a case"—Benjamin Russell—CANADIAN MUSICIAN Feb. '87

"Its voice has the kind of sonic potential that today's musicians demand, its 80-character display gives you plenty of information to help in programming, its keyboard splitting and layering are superior and the onboard sequencer is a killer . . . you simply can't afford not to check out the ESQ-1."—Jim Aikin, KEYBOARD Sept. '86

"I often suggest the ESQ-1 to those looking for a cost-effective mini MIDI studio."—Craig Anderton—Editor, ELECTRONIC MUSICIAN

For more information write: Department E.

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ensoniq®

He's worked with
the best—David Bowie,
Peter Gabriel, David
Sylvian, Talking Heads,
Brian Eno, and of course,
King Crimson—and
he's also worked away
from the public eye,
pursuing truths other
than the rock and roll
life. Transcendent, and
yet, down-to-earth,
Robert Fripp's
insights are of
interest to more than
just guitarists.

Zen & the Art of Fripp's Guitar

BY JOHN DILIBERTO

One doesn't talk with Robert Fripp; one engages in a philosophical treatise—a Zen foray into life, music, and the marketplace. Robert Fripp is an artist who has followed a distinctly personal path. This is a musician who dropped out of music in the mid-'70s, after several tumultuous years as the leader of King Crimson. When he emerged, instead of playing 20,000-seat auditoriums, he was playing in record stores, pizza parlors, and offices for audiences of one to 200.

He espoused a new philosophy and way of life born of his studies at the International Academy for Continuous Education, a descendant group of philosopher/mystic G.I. Gurdjieff and J.G. Bennett. He spoke of himself in the third person and put forth two three-year plans, the "Drive To 1981" and "Incline To 1984."

Robert Fripp has always been an enigma, a musician who lent an unusual intelligence to rock and roll. The first King Crimson album, In The Court Of The Crimson King stands as the highwater mark of the progressive rock movement, and "21st Century Schizoid Man" alone would assure Fripp and Crimson a place in rock's iconography. The version of King Crimson in the early '70s that put out Larks Tongues in Aspic, Starless and Bible Black, Red, and USA was marked by its freewheeling sense of adventure by playing against complex time signatures and arrangements. The most recent edition of King Crimson featured charged ethnic rhythms and the interlocking guitar cycles of Adrian Belew and Fripp.

Interestingly, more people know of Fripp through his work as player or producer with other artists—among them David Bowie, Peter Gabriel, The Roches, Blondie and Talking Heads. But through it all is Fripp's guitar, sometimes a violin-like wail of grace and sustain, at other times a jagged knife.

In 1972, Fripp began working with a tape loop system he learned from Brian Eno but which had also been used earlier by composers such as Terry Riley. His albums with Eno, No Pussyfooting and Evening Star are inner meditations, soul-searing inroads into the musical psyche of Fripp. The sound-generating system was dubbed "Frippertronics" and he took it on the road in the late '70s. He currently uses an Electro-Harmonix 16-second digital delay for a similar effect, played in conjunction with a Roland Space Echo, fuzz boxes, wah-wah and volume pedal and occasionally, an Ibanez digital delay.

Now Fripp is in semi-retirement again. He emerged to marry British pop singer Toyah Willcox, record an album with her called The Lady Or The Tiger, as well as Robert Fripp and the League of Crafty Guitarists—Live! He also contributed some beautiful, ethereal guitar to David Sylvian's Alchemy cassette and the Gone To Earth album.

When we talked, he was in the midst of a guitar craft seminar at Claymont Court, an elegantly decaying mansion in West Virginia. Here he teaches guitar classes that are as philosophical as they are technical. The students come to Fripp as if he were the Dali Lama of the guitar. And while they may learn something of the philosophy of Frippertronics, they'll learn it exclusively on acoustic guitars. The interview was conducted in front of about 30 students to whom Fripp directed many asides.

For an artist who has been embroiled with electronic music, be it the electric guitar, the guitar synthesizer, the Mellotron or Frippertronics and the Electro-Harmonix delay, Robert Fripp opened our interview with comments about the limits of electronics. Perhaps in clandestine support of his views, my tape recorder mangled those opening moments.

EM: *So you're not satisfied with the guitar synthesizer work with King Crimson?*

RF: If you ask me if I'm satisfied with any of the work I've done I have reservations, but my reservations would be towards the music rather than any of the instrumentation used. The Roland GR-300 guitar synthesizer was among the first synthesizers for the working guitar player—very limited in terms of sound but you could use it. We used it mainly to extend the timbre and range of the electric guitar. But with the penetration and development in guitar technology, none of it very much to my satisfaction, my approach—rather than plunge into the morass of nothing quite settled—was to go to the acoustic. That was my solution.

EM: *How do you view those first collaborations with Brian Eno, especially* No Pussyfooting?

RF: It's wonderful. It's wonderful. Side one particularly is just . . . it's true. Simply that.

EM: *And have you done anything since then that has equalled that feeling?*

RF: Yes. Yes. But you see in terms of feeling, when one does it in the moment . . . I have a little aphorism: "Distrust the musician." All they can say at the end of a performance is if they've liked it or not. They can't say if it's good or not. They can say "I had a great time thrashing around." They can say "I think it was a wonderful feel tonight." It's often irrelevant (to equate) whether the music's been good with what you feel about the evening. So sometimes it's very difficult to put your feelings on the side and simply play regardless of how you feel.

There were four performances by King Crimson, the *"Discipline"* King Crimson in 1981—it was at the Savoy in New York—which were true. The music was different. It wasn't as innocent as *No Pussyfooting*, but it had the same quality. It was true for those four performances. And there've been other occasions. It's very, very rare that you can actually, in some way, capture a moment like that.

Some of the work I did with Bowie was in the same kind of category of immediacy and honesty for me as a player. Eno, again, the solo on "Baby's On Fire" was there. I'd just gotten off a plane from America. I had the flu. I was exhausted. I was wretched, and yet the solo was burning. It doesn't matter how you feel.

The solo on Bowie's "Fashion" happened at 10:30 in the morning after a long drive back from Leeds gigging with The League of Gentlemen. There's nothing you feel less like in the world than turning out a burning solo—firey rock

THE *FRIPP PHILOSOPHY* OF *GUITAR CRAFT*

Over the last year or so, Robert Fripp has been conducting his guitar craft seminars at Claymont Court in West Virginia. He takes in students with no experience to 20-year classical guitar pros. They all come to Fripp to learn his plectrum style of playing, but perhaps more importantly, to tap into the spirit that has powered music as diverse as "Evening Star" and "Baby's On Fire."

The students play Ovation acoustic guitars almost exclusively. Fripp puts them through his new compositions, with intricate rhythms and torturous unison lines played by 12 to 30 guitarists seated in a circle. Sometimes they give concerts or record albums like *Robert Fripp & the League of Crafty Guitarists* and *The Lady or the Tiger*.

The mood at Claymont Court is somewhere between a summer camp and a monastery. The Claymont Mansion is austere and in disrepair. The only furniture I saw was folding chairs and pillows, except for the dining room which has picnic benches. Fripp's approach could be called Zen and the Art of Guitar Craft. His approach is as philosophical as it is practical. He describes it as coming into a relationship with the guitar, with music, and with the person.

EM: *I was wondering if you could talk about your teaching methods, because from what I've heard, they sound very non-western.*

RF: If you said what is it closest to, probably in terms of my cultural background—which is England primarily—it's what might happen if a carpenter in 13th century England took 208 apprentices into his home. And in England we had a dozen or so come to live and work and that's the closest to a craft apprenticeship. You know a craft is a mystery. You reach a point with your craft where the craft speaks directly to you, and at that point you enter into a kind of relationship with the craft where the craft is at least as alive and real as you are. That's the point at which you realize it's better to be anonymous than to be a star—because to be a star gets in the way of craft, and you begin

Robert Fripp (far right) conducts a Guitar Craft class.

and roll at 10:30 in the morning—just out of a truck. But it doesn't matter how you feel, you just get on with it.

EM: *You were on David Sylvian's record, Gone To Earth.*

RF: He asked me to play on his record. The actual message I got was—this was from EG, my office in London—"David Sylvian phoned. He has this piece of music and he says you're the only guitarist in the world who can play on it." Well I said "Yes!" I mean how could you say no to a line like that? So I went along and played. It's called "Wave" on the *Gone to Earth* album. The very long track, yeah. Sensational. Oh! Yes, that's true.

That music has something about it. That particular piece. The song was originally called "The Holy Blood of Saints and Sheep." Now I don't know why he changed the lyrics, but I loved the original

to value the anonymity of the craftsmen. Like who are the guys who built all those great cathedrals and so on? They didn't carve their names in the stones and leave testimonials to who they were because it would have gotten in the way. You can figure out who some of them were.

If you said what is guitar craft, I'd say one, it's a way of coming into a relationship with the guitar. Obviously. We're playing guitars.

Two, it's a way of coming into a relationship with music. So that implies that music is something of itself that you can come into a relationship with. And I'd say that's quite true. As we would express it here, music is a benevolent presence which is constantly available to us. It never goes away. Never. We do, but the music—never. So when we sit in there and we

vocal which I heard and worked through. The current one is fabulous too. He said "Go. Here you are. This is what we've got. Come up with something. Go." And I work well like that.

EM: *With the Frippertronics, were you familiar with the previous tape loop work that had been done in a similar vein?*

RF: No. I knew nothing of Terry Riley, I knew nothing of Steve Reich. I knew nothing of Lamonte Young. Yeah, I think Eno had. My understanding as Eno told me was that he came up with the idea for the tape loops system himself, although there is some thought that someone else came up with it in the '50s or something. But Eno told me he discovered it for himself. I had no prior knowledge or experience of any of the so-called minimalist or repetitive music schools. When we did *No Pussyfooting*, to me it was fairly

thrash our way through these different tunes sometimes the music is there despite what we're doing. That's remarkable when you know it's there despite all these clanging sounds, these bum notes.

And three, it's a way of practicing the *person* because to come into a relationship with one's instrument and music implies also that you're someone to come into relationship with. Guitar craft is a discipline; the discipline is the way of craft. There are other approaches. This is just one of them.

EM: *You've talked about music as being a state of mind and body of the whole being. You talked about working on four levels: automatic, sensitive, conscious, and creative.*

RF: Yes. Yes.

EM: *Please discuss this.*

RF: Yes, but in an adequate context. Otherwise they're just words. If you came on a guitar craft course, for example, we'd have three or four days common experience working together before we introduced these clever words so that we'd have some information to draw on. What is habit? Does habit have a use? What does it mean to be creative? What is creativity? These are very difficult concepts.

With habit, with functioning automatically, as soon as the little finger flies up and down off the fingerboard, well, why are you doing it? It's a waste of effort. Why not instead just keep the little finger low? This is the power of habit. Within two or three days everyone can see for themselves that their habits are more powerful than their relationship with

obvious, I used my ears. There it was, a way for one person to make an awful lot of noise. Wonderful!

EM: *And now you've replaced the tape loop system with the Electro-Harmonix box.*

RF: No! With 208 guitarists. Regarding the Electro-Harmonix, we read this advert for the Electro-Harmonix 16-second digital delay with this phrase in it, and the quote is "A Fripp in the box." So we got in the touch with them and said, you know, Fripp would like one for nothing. And they said no. So I bought one. You can't get them anymore.

Yeah I have a lot of fun with it. What I would do at the David Sylvian sessions, for example, is I'd set the equipment up and just for fun punch something in to the "Fripp in the box," and leave it playing in the studio. I did it on some Crimson sessions too, walk out and come back some three or four hours later and there it was still going except the sound had changed in the three or four hours in between. And with Sylvian, he really liked what was coming out so he recorded lots and lots and lots of these little soundscape pieces and they're all over his *Gone to Earth* album.

EM: *It seems like the Frippertronic format lends itself to long, droning patterns. Is that something that's inherent in the system or is that how you chose to use the system?*

RF: You don't have to use sustained notes but if you use impact notes without much sustain it gets very, very choppy. Very choppy! Sometimes to good effect, but I prefer the sustained approach. You can cut in and out in different ways but there's a number of ways of using the system. The way that I've used it is the way which I prefer. There are a number of other ways though.

EM: *What do you think of that music process as Eno uses it, because he uses it in a much more passive way than you?*

RF: Yes, yes. Brian doesn't really have a very strong musical background in terms

their left hand. If they had a relationship with their left hand, the hand would behave perfectly.

So is habit a bad thing? Well, obviously not. My playing is habitual. I don't have any time when I'm playing, or very, very little time to concern myself with how I'm playing and concerning myself with what I'm playing. So I've given habit over to playing the guitar. I call it a skill. It's a very, very efficient habit based on 26 years of developing a good habit. But if it's *only* habit well, there's going to be a load of dumb exercises. So I have to have a relationship with my habit so that I can direct it. Well, what part of me directs my hands? My thinking or what I feel? Very good question. Very good question. What on earth does it mean to be conscious? Well, it presumes we know what it means to be conscious. Have I been conscious? Yeah, now and again. Yeah! So I can say I know what it means to be conscious. Does it happen every day? No. Once a month? Oh, I doubt it. But I know what it means to be conscious.

Being creative? I can say whenever I've been creative it had nothing to do with me, and yet it must have had something to do with me or I wouldn't have known that the moment was a creative one. So this is when, for example, music turns over and breathes into the notes you play. One cannot fail to recognize it. It's not coming from you-the-player or me-the-player. It's something close but intangible because I can't walk into a room and say "Hey, I'm going to be creative." But what *can* I do to walk into the room so that I might stand a better chance of being in a creative moment? Well, using those four clever but mysterious words, if I walk into a room with my hands functioning automatically, superbly well, efficiently with no energy wasted, with me

in a relaxed and alert condition with my attention engaged, something becomes possible. So. That's what we can do. But until we actually experiment with that and work with it and build up information and experience, they're only bright words.

EM: *It seems like the music here and the atmosphere here is not meant to disturb; the music is very pretty. It certainly isn't "red."*

RF: No. No. The music is, there's a subtlety in there which probably wouldn't be very easy to capture with electric instruments. It's difficult to say that but there's an immediacy with the acoustic instrument. You see as soon as you plug in you have a state of "schizophonia." You are removed from the source of the sound. There is this distance. Now if you're a professional player it's something you must learn to work with. The sound that people will be listening to will be removed from you playing it and it can be at the very least, an exercise in attention. But it is schizophonic.

Working in an ensemble like this acoustically, one must be present. One must be listening all the time and be here. There's an immediacy and contact and it's quite different. The music is very carefully constructed without any solemnity and at best, it can be a construct that draws together a number of diverse people who could never otherwise work or be together in an intimate way. Now if one becomes a little skilled with musical form you can construct situations which will necessarily almost inevitably pull together the people playing them. At that point, you're beginning to be on to something. These bright ideas about the ideal society . . . if you wish to draw people together, get some of them playing in five and some of them playing in seven in a certain kind of way and it will inevitably

draw them together while they're playing it. If when they leave that room they have been together in a certain kind of way, if only for a moment on the outside meshing together, perhaps they go back in and perform it again, and maybe something can come together on the inside. Well that begins to be *very* interesting stuff. Now imagine, just as a possibility, an idea of a repertoire of music which will guarantee, by its performance, to unify the people playing it. Even as an idea that's worth shooting for. I've seen it happen here.

After being a guitar player for 26 years, working in a group context, in a studio in England I've found myself behind the hands of another member of that group looking out from behind his eyes, seeing things as he saw them, for just a fraction of a moment. We could probably accept the idea that a number of people together working as a real group would be able to experience the other members of that group in a close way. So let's say for 26 years I waited until I could know for myself that that was possible. Here I was behind the eyes, behind the hands of another member of the group, seeing things as they saw them.

Well, Bob Gerber here had this experience in a guitar craft ensemble after three days, four days? It doesn't take forever. It's always here, it's always available. And there in that moment Bob found himself behind the hands of another member of that small guitar craft group. So what would seem to be only a bright idea for us is quite real. Except Bob had it in three days and I waited for 26 years. Now there's the information: you can construct music in such a way on a purely structural and technical level that it pulls musicians together. ☐

Robert Fripp in the early 1970s on stage with King Crimson.

PHOTO: DAVIDEO KARR

of the craft of music. But what he does have is good taste. He has good taste and a perception of what's right that very, very few musicians have. So working with Eno, it's refreshing to hear the few notes but right, rather than the many, many, many that are wrong from most musicians of my acquaintance. So working with Eno was a cleaner experience and he's a very entertaining, captivating man. Lots of fun to be around.

EM: *You say that working with Eno was a "cleaner experience" and you talk about the spontaneity and how rewarding it was to do something like "The Heavenly Music Corporation." It seems like this music, Let the Power Fall, your later Frippertronics works, is a purer music and it comes through fewer filters before we hear it. What I'm wondering is, why would you go do this other music?*

RF: Which other music? With thrashing, crashing musicians? If you've ever been in the best performing live rock band in the world, it's an experience which is very difficult to put into words. But for reasons which are beyond knowing, at one particular point music bent over and took this band into its confidence. And it's very difficult not to have another shot at that.

It's very strange because it has nothing to do with you. On the other hand you must have some involvement in the process and it is an intensity of experi-

ence which leaves you never quite the same. When the power turns on, and not from you. This is the point. It's coming towards you from music in such an incredible way that the possibility of a musician living a life without that experience again is just too, too awful to contemplate. So I thought "Hey, let's have another shot at this." There you go. Easy as that.

But for having a second shot at that in 1981, for having maybe six months of something quite, quite remarkable for a second time in my life in a band like that, the next three years were utterly wretched. There you are.

EM: *On Discipline, you were getting involved in these very intricate double guitar pieces and that seemed to go away as the band became more involved in the process.*

RF: You noticed? Yes, I noticed that too.

EM: Discipline *was a very tightly arranged record. There were spontaneous improvisational moments, but the pieces were very tightly arranged, whereas* Three of a Perfect Pair, *the whole second side sounded like a fairly free improvisation. Were those reflecting changes in the band? Why did you go towards that improvisational. . . .*

RF: It wasn't a group anymore. It had ceased to be a group. It was a number of individuals. It began as a group. It began at the top and worked downwards. Yes. Very clear, very clear. It was a band of very, very fine players. The best. The best.

PCM 70 parameter settings,
Program Row 3
Concert Hall:

CONTROLS	REVERB DESIGN
	DIFFUSION 60
MIX 100 % WET	ATTACK 15
FX ADJ + 6 DB	DEFINITION 40
SOFT KNOB	REFLECTION LEVELS
SIZE 29B M	LVL MST
GATE OFF	L1 RFL/OFF
PDELAY 24 MS	L2 RFL OFF
HC 9.02 KHZ	R1 RFL/OFF
DECAY OPT ON	R2 RFL OFF
CHORUSING 51	REFLECTION DELAY TIMES
REVERB TIMES	DLY MST
RT LOW 2B S	L1 RFL 96 MS
RT MID 1.1 S	L2 RFL 000 MS
XOVER 350 HZ	R1 RFL 84 MS
RT HC 627 KHZ	R2 RFL 000 MS

"SPACE SHIFTER" *Dynamic MIDI* patches for use with the YAMAHA DX7:

MIDI CONTROLLER	PARAMETER	SCALING	LIMIT VALUES
DATA SLIDER	1.0 RT Low	+13	2.8/ 52
DATA SLIDER	1.1 RT Mid	+22	1.1/ 26
MOD WHEEL	1.2 Crossover	+17	350 Hz/ 4.85 kHz
MOD WHEEL	2.1 Attack	+81	15/ 95
FOOT PEDAL	0.8 Chorusing	+17	51/ 68
LAST VELOCITY	3.1 Lt. V1 Reflection Level	+33	Off/ −8.5
LAST VELOCITY	3.3 Rt. V3 Reflection Level	+30	Off/ −6.5
MIDI CLOCK	0.3 Size	−33	29.8 @ 100 BPM
MIDI CLOCK	4.0 Delay Master	−110	0 @ 100 BPM

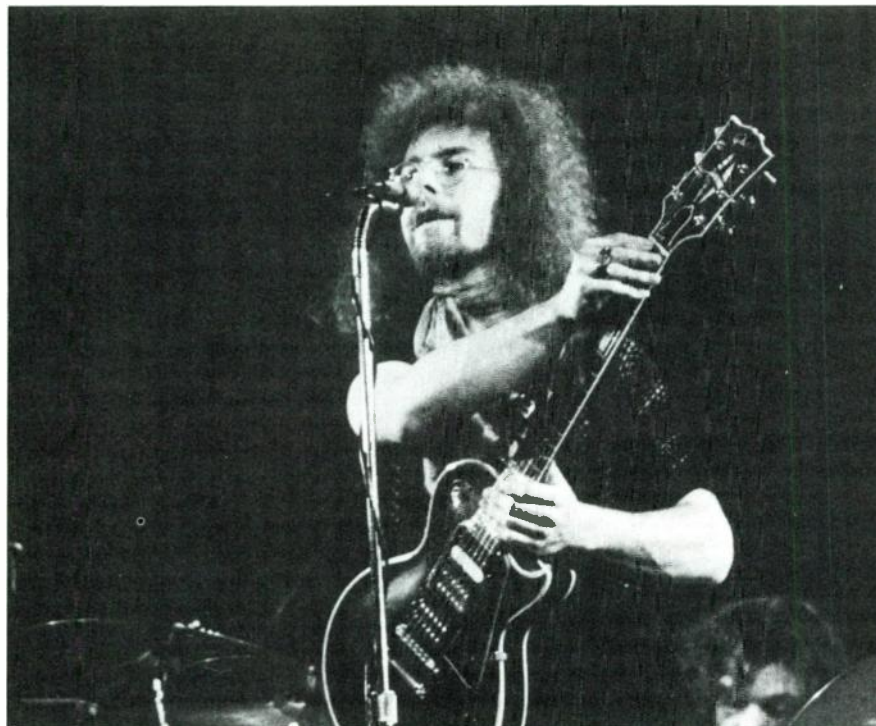"Space Shifter" was written by Lexicon Applications Engineer Will Eggleston.

WRH

We developed *Dynamic MIDI*® to take digital effects beyond static environments and flat textures. With it, you can use MIDI controllers to change your music's sonic space or texture as you're playing.

With the PCM 70 Digital Effects Processor, we pioneered the concept of parameter control through MIDI: it's still the most MIDI-responsive effects device available. Each program and register has 10 MIDI patches that assign any controller to any parameter. One controller can change two parameters, or two controllers can affect the same parameter. *Dynamic MIDI* makes possible dramatic unusual techniques that add impact to your music.

Why do we give you more MIDI patches than most effects have parameters? Because the PCM 70's algorithms (like the Concert Hall algorithm shown here) give you more parameters than some units have programs. Complex as it is, Concert Hall is only one of six (seven if you have optional Version 3.0 software) PCM 70 algorithms.

"Space Shifter" is based on Concert Hall: it's a good example of how *Dynamic MIDI* adds new dimensions to performance. The Data Entry slider lengthens the Mid and Low reverb times. The Mod Wheel moves your audience farther away by controlling

two parameters at once. As it raises the Crossover frequency, sending more reverb energy into the longer RT Low decay, it lengthens Attack time. The foot pedal choruses the reverb, while keyboard velocity controls the level of left and right reflections.

MIDI clock (included as a controller in our optional Version 3.0 software) adjusts Room Size and Delay Master for different tempos. The chart at bottom left shows that most controllers are scaled to vary the effect from a "minimal" space to one with a powerful impact. But the MIDI Clock patches are negatively scaled: as tempo goes up, size and reflection levels go down, to keep the reverb from "washing over" the beat.

For PCM 70 owners, "Space Shifter's" parameter settings and MIDI patches are in the charts on the left. The tenth MIDI patch is empty: it's yours to experiment with. In fact, the whole program is a starting point for exploration.

CONCERT HALL

Try adjusting the Scaling function to get the polarity and range of control you want. If you need a wider range than a controller offers, you can "double assign" it to the same parameter for up to twice the variation.

These are by no means the only controllers you can use. Pitch Wheels, Sustain pedals and over 120 others are included in *Dynamic MIDI*. Engineers often use Last Note when recording parameter changes to a MIDI sequencer for automated playback during mixdown.

Now a word of warning: If you don't yet own a PCM 70, don't drive yourself crazy trying to duplicate "Space Shifter" or other PCM 70 sounds using less advanced devices. With 30 to 45 parameters, every PCM 70 algorithm gives you multi-dimensional control that's simply not available in simpler units. Concert Hall alone generates many more sounds than we could discuss on one page.

Every PCM 70 has five more algorithms (six with optional Version 3.0 software, including Inverse Room). All of them are equally powerful, equally complex, equally ready to take you into new musical spaces. Which is why thousands of creative studios, innovative producers and productive musicians agree that the PCM 70 has no equal. If you think your music is something special, discover the PCM 70 at your Lexicon dealer.

PCM 70

lexicon

EM: *Bill Bruford said that when you formed that group that you wanted to do something very new and part of that newness was that you all had new instruments. He had Simmons drums. You and Adrian Belew had guitar synthesizers. Tony Levin played The Stick™.*

RF: Yes, yes. There was new technology involved. My interest in new technology is new music, but people will rarely thank you for interrupting their playing habits. They will maybe allow you to interrupt their playing habits for six to 12 months, and then there will be a change. And they might thank you years later but at the time they won't thank you.

EM: *When King Crimson started out it was involved with a general movement towards infusing rock music with classical elements.*

RF: Most of the so-called art rock music I heard, or progressive rock music, was a badly cobbled pastiche of a number of badly digested and ill-understood music forms. Yes.

EM: *And to a certain degree, King Crimson did a little bit of that. I don't mean that in a negative way but your music was infused with sort of early 20th century music. "Devil's Triangle" is from Holst's* The Planets, *obviously, but there was the Bartok sensibility that came in which was early 20th century. It seems like the late edition of King Crimson was much more contemporary classical, minimalist, and had a greater affinity with that whole movement.*

RF: All right. Go back to that night in early in 1967 when *Sgt. Pepper* was on the radio and I didn't know what it was, and my listening involved Bartok and Clapton and Hendrix and the Beatles and Stravinsky. To me they were all speaking with the same voice but with a different accent. Now for me it was if only the feel of Hendrix, if only the vocabulary was a little more sophisticated and if only Bartok was on guitar with a Marshall stack and the power turned up on 11, you know. There was a viscerality about standing in front of a wall of Marshalls and Les Pauls and thrashing Fender basses that didn't speak directly to the intellect. Why not? And yet in a chamber ensemble— quite wonderful, of course—but it hasn't got me by my nuts. Why not? I want that visceral (approach). So for me it was a question of how could you bring the two together. And Crimson in '69 was one approach, if you like to draw on the vocabulary of the western tonal harmonic tradition with the power of the Afro-Amer-

ican musical tradition known as rock and roll. Well that was '69.

In '81 we could draw on what would now be called ethnic music. In other words, the formal properties of different music systems were more readily available in 1981 than 1969 with better technology for playing them but may I say, if you're going to have a (melody) line (playing) in 14, one in 15, and a rhythm section in 17 all playing together, you've got to be pretty good. So it's no good pratting about ethnic music forms unless you've got the chops to deliver on them. If you want to know how long it takes to learn to tap your left foot in four, beat five on top and have a conversation at the same time, it's 12 to 14 years. I think everyone in the room knows that one.

EM: *When I think of Robert Fripp I think of Larks Tongues in Aspic, I think of Starless and Bible Black, and I think of you as an instrumental musician, not as a rock musician. It seems like when you are involved with rock songs, very often they tend to be satiric and kind of a parody of rock form. I was thinking particularly of "The League Of Gentlemen" and "You Burn Me Up, I'm a Cigarette" from Exposure.*

RF: I love "You Burn Me Up, I'm a Cigarette." Life with you is a loser's bet. You make me anxious. Strategic interaction, terminal inaction. Bitter hostile faction. Wonderful!

Hall was phenomenal, you know. Daryl Hall, you know. I said there's the words, go sing. He went out and sang, you know, one take. Fabulous. Oh, yes. Lovely, visceral. Yah-keh-keh-keh-keh-keh-keh-keh-keh. Very, very hard to play with that intensity because it requires an awful lot of effort for downstrokes at that speed, Yah-keh-keh-keh-keh-keh-keh, on those chords.

EM: *On his projects, what's your relationship with Eno? Do you come in as a sideman and he presents you with tracks that are already done or are you there—he seems to compose a lot in the studio—are you there during that composition process?*

RF: I generally turn up and he says go and puts it up and I respond and he gives me all the space and support to do that. Simply that.

EM: *I believe he was the one who created Sky Saw Guitar.*

RF: Yeah, Sky Saw was a name for a particular sound which had to do with feeding the guitar through a VCS Synthi synthesizer and digital feedback. It's a specific technical approach for getting

the sound. You can get the sound or a very close approximation in a number of different ways, but that was the name he came up with. He came up with that particular sound. Wonderful rrrr. So gripping. Great. Yeah. My response to that is visceral. Now you don't need lots of clever notes and theories about musical organization. You just hit the G and go rrrr. And that's it. It's all over.

You know, that used to really frustrate me. There I would be practicing and working hard and all the different things one could play, and yet you turn up the amplifier and hit the one note and it would go hhhhhhmmmmmzzzm and it would be all you needed. Wonderful appeal. Wonderful appeal.

EM: *Can you get that out of an Ovation Acoustic?*

RF: In a different kind of way actually. It's possible to have the same kind of visceral effect playing certain bass lines in a certain kind of way.

EM: *I think your electric guitar style is identified with a sustained sort of sound, but with acoustic guitar you can't really get that. Have you adjusted, or are you doing something on*

acoustic guitar completely and utterly different from what you do on electric?

RF: Yes is the correct answer. I began as an acoustic player and I was a good acoustic player. And it is a different instrument with only similarities to the electric guitar—

◯n tape loops: "There it was, a way for one person to make an awful lot of noise. Wonderful!"

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the frets and the strings basically, and two hands and a pick. But it's a different instrument with a different way of life involved in it. Different music. Different vocabulary. Quite different. I had to *learn* to be an electric guitar player, and it was only really after 12 years of being a player—about the time of the first Crimson

Les Paul and Marshall stack—that I could find my own voice with an electric instrument. Until then amplification hadn't quite been right. (Electric guitar) wasn't an instrument of itself somehow whereas Marshall stack and Les Paul, that was an electric guitar. You knew. Then you put a fuzz box in line and yynnnggg. You were away. Yes. So I became an electric player about 1969.

Now coming back to acoustic guitar about 1985, I suppose it's an irony that the one thing at which I excel as a guitar player, my specialty, is right-hand picking technique. Would Americans call it flat picking? I call it something different but it's a very, very specialized style which few players can play well—and I'm one of them. The irony is almost nowhere in my professional life will you find any reference to it. Now that's one of the ironies which I accept with a good grace as I accept most of the ironies in my life. But nevertheless it's true. Now the work we do here on some of the things we play, my guitar playing is exceptional and almost unique but very few people will ever have heard it. It is another of the ironies

of my life that my very finest playing very rarely gets heard, even the electric playing somehow mysteriously disappears from records. Yes it's interesting, isn't it.

EM: *Now much of your music has what I would call an almost demonic quality, and I don't mean that in a satanic sense but I mean a very dark quality. "Schizoid Man" is an obvious example. "Indiscipline" would be another one, "Requiem," and most of Red.*

RF: Oh, *Red* is something quite, quite different. *Red* has a viscerality but intelligence which combined is quite terrifying, or can be. The last Crimson played it well but it was never really played well. See, *Lark's Tongues* was very, very primal. But young guys of 23 to 25 trying to play that music in 1972, they weren't good enough to play the music for a number of reasons. One, chops. Two, egotism. You can't play music if you're imposing your bright ideas of how it should be played on it. You have to give the music more respect. And in '81 for a period, a few Crimson gigs got into the spirit of *Lark's Tongues*. But it still has never been played right which is very frustrating for me.

EM: *Was it a matter of chops, or was it a matter of understanding the spirit of the music?*

RF: The difficulty is that both are involved. Now you have good players, good chops people. How can you be a good player with chops but drop all your chops and just play one note? It means you have to be fully what a human being can be and at the same time plugging into this very, very primal situation where we are all animals and yet we're all animals. This is the creature we inhabit. Well, that may be part of what we are, but there's something more which is possible for us, and the demand is made in this music that that element must also be involved. So what you're asking for is four musicians in a rock and roll band that are enlightened. Well, let's face it. It's asking a bit much especially if you want to earn a living too. So maybe one could realistically say that the demand is a little high. Well, fair enough. So from my vantage point it's excruciatingly suffering, excruciatingly painful but there you are. I'm a hero. What more can I say? **EM**

John Diliberto is the producer of Totally Wired: Artists in Electronic Sound, *a weekly program on electronic music produced for Pennsylvania Public Radio Associates and broadcast on public radio stations across the United States.*

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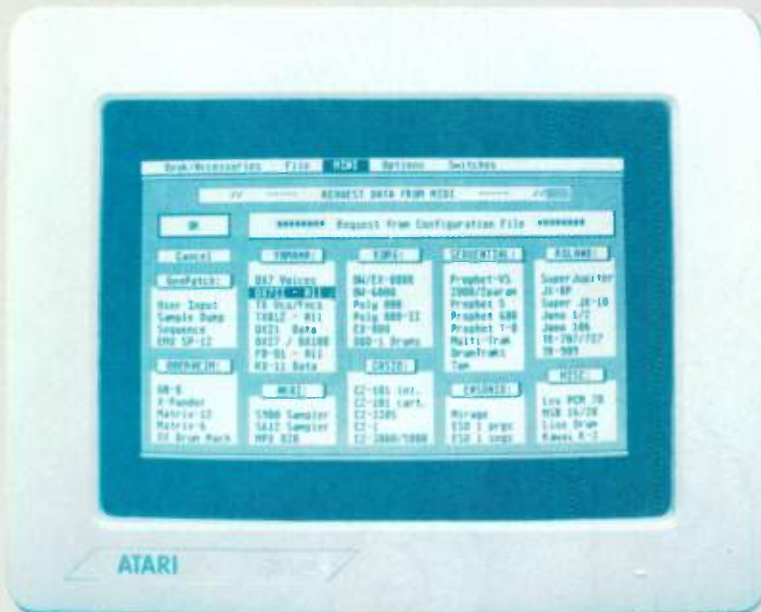
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WRH

Most of us would rather not dwell upon the unpleasant possibilities of accidents, so safety goes unconsidered. But safety must be practiced when it can do some good: before an accident happens.

Safety in the Electronic Music Workplace

BY ALAN GARY CAMPBELL

The electronic musician's skills are a symbiotic mix of methods from many fields. Yet, within this multi-disciplinary matrix, safety is often considered only after the fact. The electronic musician's tools are high-tech devices with high-tech hazards. Practicing safety is no less relevant than practicing musical technique or computer programming.

ELECTRICAL SAFETY

Like any powerful energy source, electricity can be dangerous if misused.

Never use equipment with damaged power cords or plugs; this can cause a shock hazard, or short out and ignite! Inspect cords and plugs regularly for signs of wear-and-tear, and replace any damaged components immediately. Never position line cords beneath a carpet or rug: the resulting friction can wear through the cord insulation, causing a short and a major fire hazard.

Do *not* overload extension cords, outlet strips, or wall receptacles; this too, can cause a fire hazard. Overloading often escapes consideration—most electronic music devices are low-current—but some power amplifiers and lighting devices can pull a *lot* of current.

AC current ratings can be confusing. A common misconception involves combined ratings: for example, many people infer that a 10-amp extension cord connected in series with a 10-amp outlet strip provides only a 5-amp capacity; but

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the actual capacity is 10 amps. However, when combining cords and strips of *different* ratings, the available capacity is the *lower* of the two ratings—e.g., a 10-amp extension cord in series with a 15-amp outlet strip provides 10 amps.

Check the current rating of extension cords and outlet strips, before installation. On cords, the rating is usually printed on the cable jacket, or embossed on the plug

(if these locations list different ratings, use the lower of the two). On strips, the rating is printed on the case, bottom panel, or ID plate. Add up the total current required by all the devices connected to the cord and/or strip (if any device lists, instead, a power rating in watts, divide this value by 110—the typical AC voltage value in the United States) to obtain the approximate average current, in amps.

Make sure that the total is *less* than the rated current. If any motor-driven devices (large fans, air compressors, Leslie™ speakers, etc.) are connected to the cord and/or strip, derate the rated current by about 20% (multiply the rated current by 0.8 to find the usable current).

Modern 3-wire, grounded wall receptacles usually feed from a 20-amp circuit breaker. Several receptacles may be serviced by the same breaker, so any one receptacle may provide considerably less than 20 amps, depending upon the loading of the others. Even when only a single outlet is fed from a breaker, the 20-amp maximum should not be exceeded (though many breakers will tolerate some excess current). Multiple-outlet "extenders," such as the Radio Shack Model 61-2622, are intended for low-current use, and should not be used with high-current equipment (motors, heaters, etc.).

Proper AC wiring is mandatory for the safe operation of electronic music equipment. Unfortunately, the working musician normally has little control over the condition of wiring at job locations; night club wiring, in particular, is all-too-often old or inadequate, or incorrectly installed. *Evaluate available wiring in advance of each gig, whenever possible.* Outlet wiring can be checked with an *outlet tester;* line voltage can be safely measured with a *line monitor* (see "Service Clinic: Preventive Maintenance, Shipping, and Storage," July '86 **EM**; back issues $3.50). Suspect wiring should be checked by a licensed electrician before use.

AC-powered electronic equipment often incorporates a 3-wire, grounded power plug: the grounded equipment case serves both as a shield against electromagnetic interference, and as a safety device to shunt hazardous internal AC voltages to ground in case of a malfunction. Nonetheless, to avoid ground loops in a system, it is often necessary to use a 3-wire to 2-wire adapter to lift the ground on one or more devices. When this is done, *the ground return of the affected device depends solely on the interconnecting audio cables.*

Therefore, to achieve proper safety grounding and adequate shielding, all audio cables must provide a low-impedance ground-return path (impedance can increase due to kinks, breaks and oxidation in the shielding braid; also, given two pieces of identical cable, the longer one will have greater impedance). *Test audio cables regularly* by measuring their resistance with an ohmmeter and checking for breaks or intermittents with a cable tester (such as the Tri-Test Cord Checker circuit presented in the December 1983 issue of *Guitar Player* magazine; also, a cord tester circuit is scheduled for publication in a future issue of **EM**). Repair or replace damaged, suspect, or intermittent cables. Do not use 3-wire to 2-wire adapters unnecessarily. In older buildings with 2-wire outlets, establish an earth ground at a water pipe, and connect all the adapter ground leads to it. If the available wiring appears to be damaged or inadequate, or you are simply not sure how to proceed, *always* consult a licensed electrician.

Never operate equipment that has been exposed to rain or condensation, or has had liquid spilled onto or into it. Pure water is essentially a nonconductor, but rain water, tap water, and beverage mixes are anything *but* pure. Minerals, sugar, salt, soda, etc., all form highly conductive ionic solutions with water. Thus, moisture can provide a conduction path for hazardous voltages normally present only *inside* equipment, and also provide a ground-return path such that these vol-

tages can conduct through the body of anyone coming in contact with the gear— a very dangerous situation.

Fuses and circuit breakers cannot protect personnel against these hazards; they are too slow and trip at loads that are too high. However, considerable protection is afforded by Ground Fault Interrupters (GFIs). These devices sense any difference in current flow between the "hot" and "neutral" AC conductors, and shut down the line; response time is on the order of milliseconds. *Install GFIs in any outlet that is to be used out-of-doors or near water, including stage wiring and power-distribution snakes.* Grounded outlets incorporating GFIs are available from most hardware and discount stores. These easily retrofit to replace existing grounded outlets; but, again, if you're not skilled in this type of work, refer the job to a licensed electrician.

Never replace a blown fuse with a fuse of a higher value, or with a fuse case wrapped in foil. Always carry spare fuses of the correct value—take the manufacturer's "For Continued Protection Against Fire Hazard, Replace Fuse With Same Type And Rating" sticker *very* seriously. If a device repeatedly blows fuses, it is probably defective—don't use it until you get it fixed.

Under normal conditions it is rare that a defective electronic music device will cause a fire. Usually, when equipment "smokes," a fault has occurred causing excess current to flow in a resistor or inductor, thus overheating the component and burning off its insulation. Generally, the excess current will blow the line fuse, and the event will be self-limiting—only the affected components will be damaged. But, if the line fuse is bypassed or replaced with one of a higher value, excess current may continue to flow, dangerously overheating the affected components, and increasing the risk of fire.

As I've mentioned before in "Service Clinic," many line-powered electronic music devices contain exposed AC contacts that are accessible when the device is disassembled for service. *Such contacts present a potential shock hazard.* The point here is not to put anyone off DIY service—technicians work safely with AC power every day—but to remind you to *be careful.*

Take extra precautions with AC-powered gear, especially when you're working near a power supply. Tape over exposed line-voltage contacts, or cover the entire power supply board with a towel (but don't run it this way for long—it

might overheat!). Never work on line-powered gear when you're tired, upset, ill, or distracted. Be safe.

CHEMICAL SAFETY

Service and maintenance procedures often involve hazardous chemicals, so proper attention to chemical safety is *very* important. Most chemical containers list specific safety precautions,

but these are often ignored in the mad rush to eject work from the shop or studio. Failure to follow chemical safety practices engenders unnecessary risk.

Many shop chemicals are somewhat toxic. *Read the labels. Follow the safety guidelines.* If the container is marked "Vapor Harmful," then use the chemical only out-of-doors or in a well-ventilated area. Persons with sensitive skin should wear gloves when working with chemicals. Wash your hands thoroughly afterwards— whether you wear gloves or not—but do not wash in a sink that is used for food

Soldering Safety

Hakko Workstation Smoke Absorber

Soldering is an important fabrication and repair skill that, despite the high temperatures involved (600 to 900 degrees Fahrenheit), is relatively safe. (Picking up the wrong end just *once* is usually enough safety training for an entire career!) Occasionally, rosin core solder (the *only* type of solder you should use to repair electronic music equipment) will splatter as the flux core boils, but this is generally not hazardous; however, solder dripped directly from the iron can cause a painful burn.

A soldering iron is hot enough to ignite paper and deform plastics. *Do not allow the soldering iron tip to inadvertently come in contact with combustible or heat-sensitive materials.* Return the iron to its holder between uses. Unplug your iron (or turn off your soldering station) if it is to be left unused for more than an hour. Keep children and pets away from soldering equipment.

Solder fumes contain traces of tin, lead, and organic compounds—all potentially harmful. Adequate ventilation is important, especially when you are soldering for extended periods. Local air filtering is advisable; but conventional room-air filters are not necessarily effective. Products such as the Hakko 491 Workstation Smoke Absorber are designed specifically to remove solder-vapor contaminants from the air. Inside the 491, a quiet fan draws air through the a replaceable urethane foam/activated carbon filter; up to 70% filtration efficiency is claimed. The 491 is available from Techni-Tool, 5 Apollo Road, PO Box 368, Plymouth Meeting, PA 19462-0368.

preparation or other day-to-day tasks. *Keep food and drink out of the work area— period.*

Spray chemicals require special safety precautions. *Wear eye protection* to guard against contact with the chemical, and against physical injury from the force of the spray. Normal eyeglasses are inadequate protection. "Wrap-around" type safety goggles are preferred; these are available in designs that can be worn over eyeglasses. *Never* discharge spray chemicals with the nozzle (or extension-tube orifice) near the skin.

Trichlorotrifluoroethane, or "TF," sold under the trade name Freon-113,™ is perhaps the most common shop chemical. It is usually packaged in spray cans (e.g., Radio Shack's Cleaner/Degreaser, Part No. 64-2322) and is used as a solvent to clean contacts, PC boards, etc. TF is nonflammable, will not attack plastics or most paints, and leaves no residue (it's often referred to in tech jargon as "no-residue cleaner"). It is, however, mildly toxic and somewhat irritating to the skin and mucous membranes. Use it with adequate ventilation, and wear eye protection when spraying.

TF is the base ingredient in TV-Tuner Cleaner (e.g., Radio Shack's TV Tuner & Control Cleaner & Lubricant, Part No. 64-2315) and similar products. These compounds contain silicon lubricant, and accidental overspray can cause slick spots on floors. Watch out for spills. Wipe up overspray with a paper towel moistened with pure TF.

Denatured ethyl alcohol is used as a solvent for defluxing circuit boards and other cleaning tasks. It is highly flammable; a spark from static discharge is sufficient to set it off. *Take extra precautions when working with flammable solvents.* Prohibit smoking materials, propane torches, resistive and quartz heaters, and other sources of ignition in the work area. Use optimum ventilation, or work outside. Discard solvent-soaked rags in a ventilated, exterior trash receptacle located safely away from any flammable materials.

Methylene chloride is also used to deflux PC boards; it is the active ingredient in most aerosol flux removers (e.g., Radio Shack's Flux Remover Spray, Part No. 16-1234). It, too, is highly flammable, and somewhat toxic—*all* of the previous cautions apply. Methylene chloride is a very active solvent; it will literally melt most plastic components, even after it has dried. Be careful.

Dichlorodifluoromethane, or Freon-12™ (e.g., Radio Shack Component Cooler, Part No. 64-2321), is used as a spray to cool suspect components during thermal-failure-mode analysis. It is nonflammable and nontoxic. However, the spray produces local temperatures as low as -70 degrees Fahrenheit! It can cause frostbite, so skin contact should be avoided.

Service-grade adhesives are considerably more powerful than their consumer counterparts. Many industrial-strength general-purpose cements will vigorously attack Formica™ and similar surfaces that are impervious to hardware-store glues; and commercial cyanoacrylate esters, unlike Super Glue™ and Crazy Glue,™ really do "bond skin on contact." Silicone sealer/cement releases acetic acid, while curing (hence the "vinegar" smell), which can discolor fabrics and paper, and irritate the skin and mucous membranes.

Uncap these high-tech adhesives on a protective surface (e.g., a sheet of aluminum foil). Apply them sparingly with a toothpick or dressmaker's pin. Avoid contact with the skin; wash your hands thoroughly afterwards. Keep a bottle of "Super Glue Remover" handy when working with cyanoacrylic adhesives.

Equipment modifications and prototype construction often involve priming and painting various materials. Paints

Protecting Your Equipment
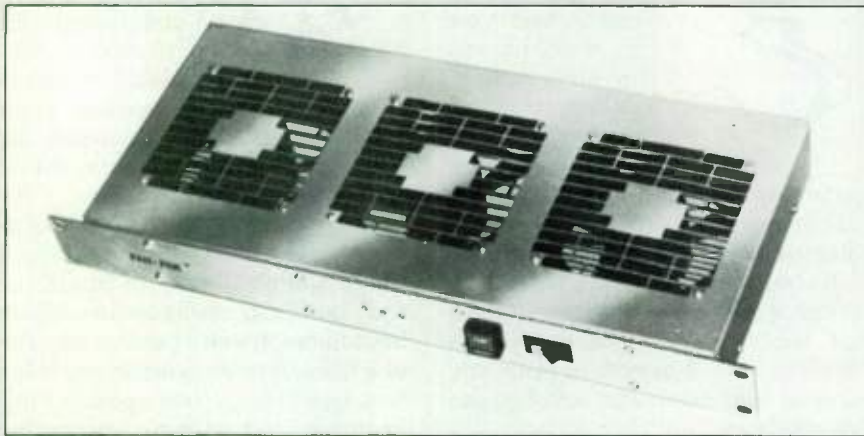
Unitrack Fan Pak

Your gear will perform best if it receives regularly scheduled maintenance, and is not subjected to extremes of temperature, humidity, dust and dirt. You should also protect it from possible damage due to improper AC power, unstable mounting, and rough handling during shipping and storage. These topics were covered in detail in a previous Service Clinic, "Preventive Maintenance, Shipping & Storage" (July '86 EM). Protection devices are a wise investment, and should be allotted a part of your equipment budget.

AC surge suppressors are becoming commonplace, and suppressor technology has recently improved markedly. (For those not familiar with such devices, surge suppressors incorporate Metal Oxide Varistors—MOVs—that act as voltage-variable resistors to shunt high-voltage transients, or "glitches," to ground. Transients can damage equipment and garble computer data.) For example, the RCA SK-403 boasts an improved transient response over previous designs; and comes in a three-outlet, cube-tap-style case, for about half the price of older single-outlet devices. As I've stressed before, surge suppressors are cheap insurance—use them.

Transients of a different kind can zap tweeters and midrange drivers should AC power fail and then restart while PA and instrument amps are in use. Speaker fuses and circuit breakers are not fast enough to stop these glitches. An automatic audio mute, such as the new MT-8 from Conquest Sound, can protect sound-reinforcement speakers. One channel of the device connects in series with each amplifier input; when a power failure occurs, the MT-8 immediately mutes all channels. Channels will not unmute until reset manually. *Conquest Sound, 7319 Duvan Drive, Box 757, Tinley Park, IL 60577.*

Power amps and accessories generate large amounts of heat; adequate heat dissipation must be provided to insure optimum performance and service life from the equipment. In rack mount enclosures, heat dissipation is often inadequate—even with cabinet-mounted and equip-

ment-mounted fans. One solution to this problem is to install a rack mount fan assembly, like the Fan-Pak™ from Unitrack.™ Multi-fan units (up to six fans) fit in a single rack space; optional replaceable air filters are available. No cutting or drilling is required for installation. (However, it might be necessary to leave an open rack space above and below a heat-generating piece of equipment in order to insure sufficient air flow.) *Unitrack, Calabro Industries, Inc., PO Box 1927, Goshen Corporate Park, West Chester, PA 19380.*

To protect your gear from dust and dirt, equipment covers are available from Cleer Covers,™ Gotcha Covered,™ Valley Designs, and others; or you can make your own—but avoid using plastics and synthetic fabrics that can produce potentially damaging static electricity. In a pinch, drape a clean towel over your synth. *Cleer Covers, Future Sounds, 932 East 5th Street, Oxnard, CA 93030; Gotcha Covered, PJL/A Music Products, Ltd., 22N159 Pepper Road, PO Box 242, Barrington, IL 60010; Valley Designs, PO Box 193, Shelburne Falls, MA 01370.*

RCA SK-403 Surge Suppressor

PHOTO BILL KINNAMAN
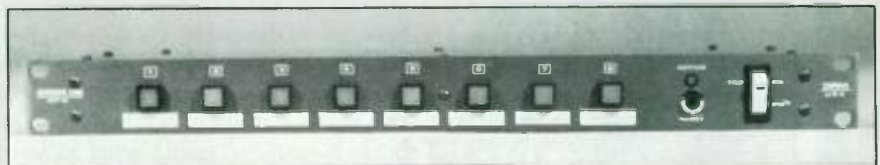
Conquest Sound MT-8 Automatic Audio Mute

should be treated as flammable, somewhat toxic chemicals. Use adequate ventilation. Wear a protective mask when spray-painting. Inexpensive, disposable masks are adequate for most applications; they should be used once, then discarded.

Also consider that any chemical, either liquid or aerosol, that contains trichlorotrifluoroethane, dichlorodifluoromethane, or similar compounds—collectively referred to as chlorofluorocarbons, or CFCs—will release these compounds into the atmosphere during use. Recent research in atmospheric chemistry strongly suggests that CFC contamination may deplete the earth's atmospheric ozone layer, causing an increased transmission of ultraviolet sunlight. This may, in turn, increase the risk of skin cancer for persons who work out-of-doors, and possibly affect certain biological functions in animal and plant life; it may also increase global temperatures, causing climatic changes with attendant agricultural and economic shifts. *Chemicals containing CFCs should not be used where non-CFC-based alternatives are available.* For example, "compressed air in a can" is actually a mixture of TF and air; in many applications filtered air from a compressor is a satisfactory alternative. Aerosol flux removers incorporate CFCs as propellant, but ethyl alcohol can often be used to deflux, instead. Components undergoing thermal-failure-mode testing can be

Flammable or Inflammable?

Inflammable means "capable of rapid combustion." It has the same word root as "flame"; but, because of the "in-" prefix, many people infer that it means "*not* flammable"—a dangerous mistake. To avoid potentially hazardous confusion, many safety signs and warning labels use the coined word *flammable* to identify combustible substances; and *nonflammable* to identify noncombustible substances. Most dictionaries and usage guides regard "flammable" as a functional non-word, but it seems that one must first survive contact with *flammable* materials in order to later learn the correct meaning of *inflammable*. Besides, what's in a name? Trichlorotrifluoroethane by any other name would smell...well, like Freon!

chilled using non-refrigerant-based devices. Unfortunately, CFC-based products are still necessary for certain service procedures, especially those performed in the field; but they should not be used indiscriminately.

For similar environmental reasons, always dispose of toxic chemicals safely. If in doubt about how to proceed, contact the manufacturer or your nearest Environmental Protection Agency office.

TOOLS
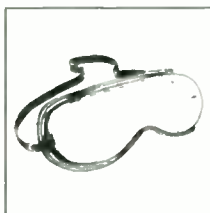
Electronics hand tools are relatively safe to use. Most hand tool hazards involve cutting, drilling, or forming operations. *Wear safety goggles when performing these tasks.* Most technicians do not wear goggles when using common diagonal cutters on soft copper wire. But, stiff wire and component leads can fly off at high speeds and in all directions when cut. This is clearly unsafe, and goggles *should* be worn. When cutting piano wire or other hard-drawn wire, safety goggles are *mandatory*.

Material to be sawed or worked should be held firmly in a vise. Never work material in such a manner that a blade, bit, or tool might slip and cause injury.

X-acto™-type razor knives are used for PC reworking and general cutting. Such blades are extremely sharp and should be used with the utmost care. Again, never cut material in such a way that the blade can slip and cause injury. Do not leave a razor knife in an unexpected place on the workbench. Never discard used blades haphazardly in a trash receptacle. Use the blade-disposal slot in a lavatory cabinet; or tape used blades between two thick pieces of cardboard, and then discard. X-acto blades can be purchased in a dispenser pack that incorporates a slot for safe disposal of used blades—a good idea.

Power tools are necessary for many fabricating processes, but they require skill and alertness to operate safely. Never operate power tools when you are tired, upset, ill, or distracted. Do not even turn on a power tool unless you have the requisite skill to use it properly. Many DIY books purport to teach such skills, but it is unquestionably better to learn them one-on-one from a craftsman or

teacher. Note that power tools need proper maintenance for safe operation. Inspect power tools before use. Never attempt to use a tool that is not in good working condition. Observe all safety procedures, and wear safety goggles. Do not wear loose clothing that can get caught in moving parts of the tool; tuck long hair under a cap or hair net.

STATIC ELECTRICITY
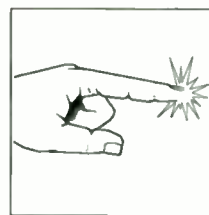
In equipment-service areas where CMOS ICs and other static-sensitive components are handled, or flammable chemicals are used, static electricity must be strictly controlled. This is accomplished by humidifying the air, and installing anti-static devices.

Large installations often incorporate humidifiers within the central heating and air conditioning system. Small shops and studios can usually obtain adequate humidification with a portable unit. Portable humidifiers are generally one of four basic types: the *cool mist* type uses a motor-driven mechanism to vaporize and disperse water contained in a reservoir. The *ultrasonic* type uses a piezoelectric transducer, driven by an ultrasonic signal source/amplifier, to atomize the water, which is then dispersed by a fan. The *fan-and-tank* type employs an AC fan motor to blow air across a reservoir of water, and disperse moisture—this construction looks rather like a rectangular trash can with air vents on two parallel sides. The *steam* type in essence consists of a large glass jar/reservoir with an attached AC heating element that boils the water and forces the resulting steam out through a dispersion valve.

Any of these will work, but the cool mist type is the most energy-efficient, and the least expensive; for many applications, it's all that's needed. This type does produce some water-droplet "fallout" near the unit, and should not be placed in close proximity to equipment and fixtures. The ultrasonic type discharges less fallout, and often provides a larger reservoir and longer running time, though in other respects it is probably unjustifiably expensive. The fan-and-tank type provides the largest reservoir and longest running time, but also produces the most fallout and the most noise; it is generally used to humidify large work

and storage spaces where these drawbacks are unimportant. The steam type, though common, is not recommended. It often exhibits a short service life: as the water in the reservoir boils, dissolved minerals precipitate out, eventually collecting on the element, causing it to fail. Also, the warm water and steam provide an attractive environment for bacteria, molds, etc.

All portable humidifiers—except the steam type—will deposit a layer of fine grayish-white dust on surfaces within the humidified area. This is caused by the dispersion of minerals dissolved in the water. Using distilled water in the humidifier eliminates this problem, though purchasing it in consumer quantities can be prohibitively expensive. Many shops distill their own.

In service areas, tile floor-covering is preferred to carpet; if carpet is used, it should be a low-pile type. Conductive tiles and anti-static carpets are available, but are *expensive*. Anti-static sprays can be used to reduce static in carpeted areas, but the effect is only temporary, and such sprays are costly and leave some residue on the carpet.

Better static protection is afforded by various anti-static devices: *Wear a conductive wrist strap when working with static-sensitive components:* **Fig.** 1 shows the Radio Shack Model 276-2399. This inexpensive device attaches to a local earth ground, and safely bleeds off static charge through an in-line 1 MΩ resistor. Anti-static mats and touch plates are available to protect computers and test equipment.

Store and transport static-sensitive ICs only in anti-static rails, foam, or bags—that are clearly marked as such. Place ICs on a conductive surface when they are out on the bench. A simple DIY conductive surface consists of a square of conductive foam (Radio Shack Part No. 12-3456, or equivalent) placed on a grounded metal plate. Use only 3-wire, grounded-tip soldering irons and stations for soldering static-sensitive ICs; the specified tip potential should be in the low millivolt range.

TF, component cooler, and "compressed air in a can" are available in anti-static versions (the egress of material from a standard spray can produces considerable static), but the anti-static ingredients in these can leave a residue, which rather negates the purpose of the chemicals. One solution to this problem is to use conductive, grounded spray nozzles, such as the Tech Spray models 1916-F and 1916-S that retrofit to the company's aerosol cans.

Commercial installations often employ elaborate static monitoring and control systems, at great expense; but the above procedures are adequate for most shops and studios—and, combined, they are probably less expensive than replacing a single blown LSI IC.

FIRE SAFETY

Observance of proper safety procedures will help to make a shop or studio fire-safe, but don't assume that a fire "can't happen to you." Commercial installations are required to install precautionary sprinkler systems, smoke

Fig. 1 *Conductive Wrist Strap*

PHOTO: BILL KINNAMAN

detectors, etc. Small shops and home studios are often exempt from such requirements, but should still install appropriate safety equipment.

Smoke detectors should be mounted in every hallway, and in every large room in a building—especially if the room is frequently closed off. Test detectors weekly. Rooms above the ground floor should be provided with chain ladders for emergency escape, unless multiple exits or fire escapes are present. Occupants should be familiar with available exits from the building.

Fire extinguishers should be located in main work areas, with their locations clearly visible and unobstructed.

Extinguishers are classified according to the type of fire they can effectively extinguish: *Class A* extinguishers are used for solid-material fires (wood, plastic, etc.), *Class B* are used for liquid fires (oil, gas, etc.), and *Class C* for electrical fires. Fire extinguishers may be classified for more than one type of fire. Additionally, extinguishers are *rated* with regard to the theoretical area of combustion that they can extinguish. For example, a 1A10BC extinguisher is rated for a one-square-foot Class A fire, or a 10-square-foot Class B or C fire.

Common extinguishers are one of three basic types: dry chemical, carbon

3M Model 6300 Disposable Hearing Protectors

Protecting Your Hearing

Repeated, prolonged exposure to loud sounds can permanently damage hearing. Musicians and sound reinforcement personnel in the rock and fusion genres are particularly vulnerable to such damage, and the results can be debilitating: loss of high-frequency hearing (not unlike the hearing loss that can accompany aging), and pain or discomfort upon later exposure to even moderately loud sounds.

Your ears can recover from brief exposure to loud sounds, but *not* from the kind of exposure that is often encountered with amplified music. Many performers disregard warning signs of hearing loss: tinnitus, or "ringing" in the ears; "noise hangover" after performance or rehearsal; and the inability to understand normal conversation in a noisy environment.

Some common-sense steps can help to preserve your hearing. In live performance, loud amplification should be directed away from the performer; and the audience should be prevented from moving too close to speaker arrays. Amplifiers that are run wide open to obtain speaker distortion should be located offstage and miked. In the studio, loud amps should be run in an isolation booth. Performers should cue from local monitors or headphones, at lower volumes.

Unfortunately, such ideals are often difficult to put into practice. Stage technicians may find it necessary to approach speaker arrays that are generating high sound pressure levels; and guitarists must be close to an amplifier to produce controlled feedback and other effects (unless they simulate these sounds via signal processors). Though headphones offer some protection from noise, and theoretically allow the wearer to hear a monitor mix at reduced levels, they are often run wide open, thus negating their principal advantage—and even wireless types restrict the performer's motion to some extent.

Hearing protectors offer the only truly effective safeguard against prolonged exposure to high sound levels; but they are severely underutilized. This stems largely from lack of availability: effective, disposable foam hearing protectors are rarely carried by music stores; they're often available only from safety-products wholesalers who deal in large quantities. (Foam hearing protectors can occasionally be found at sporting good stores.) Furthermore, many musicians have tried older, reusable plastic types—and abandoned them as uncomfortable, unsanitary, and ineffective. Thus they're reluctant to try more modern, improved devices.

Mix Bookshelf now offers the 3M Model 6300 disposable foam hearing protectors. These are inexpensive, and easy to use: the device is rolled longitudinally between the thumb and index finger to compress the material, then inserted into the ear canal, where the foam decompresses to provide a snug fit. Foam hearing protectors such as the 6300s are reasonably comfortable, and can provide up to 35 dB of attenuation. They're packaged in pairs in sealed, sanitary poly bags—to be used once, then discarded. 3M Model 6300s, 20 pairs for $10, postpaid. *Mix Bookshelf, 2608 Ninth Street, Berkeley, CA 94710.*

(Note: some persons experience lightheadedness when wearing hearing protectors. This can be caused by sinusitis and other conditions that affect the Eustachian tubes, making equalization of differential air pressure within the ear canal difficult. When this occurs, it may be necessary to use on-the-ear protectors or headphones instead.)

dioxide, or Halon. All three types are satisfactory for fighting Class A, B, or C fires. A 1A10BC dry-chemical extinguisher is probably adequate for studio or shop use, though a Halon extinguisher is advantageous for discharge in or near electronic equipment. Halon extinguishers leave no residue, unlike dry chemical types, and avoid subjecting equipment to the damaging thermal shock caused by carbon dioxide types; however, they are comparatively expensive.

The fire-hazard potential of portable electric heaters, both the resistance-element and quartz types, should be noted as such heaters are often used in small shops and home studios. Quite often, the cords and plugs provided with these heaters are of adequate current-carrying capacity only when new. As the components age, their contact surfaces become oxidized, causing excess resistance and overheating of the cord and plug, which in turn accelerates the contact oxidation— a vicious circle. Furthermore, heater fan motors require periodic lubrication, which is often overlooked. Lack of lubrication increases the load on the fan motor, and hence the required current, thus aggravating the degradation of the cord and plug. The combined factors eventually cause heater plugs and/or cords to melt through and short out. This problem is the cause of many wholly-avoidable fires.

Fan motors should be lubricated as required. Heater cords and plugs should be inspected regularly for discoloration, deformation, and other signs of high contact resistance and overheating. At least once yearly, heater plugs should be cut off, and a new plug installed. Only a heavy-duty commercial plug, with a current rating higher than the actual heater current, should be used; the cord leads should be tinned before attachment to the plug contacts. *Resistance heaters should never be allowed to run unattended.* Heaters should be UNPLUGGED WHEN OFF.

Soldering irons, of course, run at high temperatures and should be handled carefully (see sidebar).

It is normal for some electronic equipment to run rather hot. When servicing equipment, *avoid touching voltage regulator ICs and power transistors while they are in operation.* Work surfaces should be protected from excess heat. A suitable DIY protective surface can be constructed from a rectangle of sheet metal with some rubber feet on the bottom (for air circula-

tion). Even with such protection, devices under test should not be allowed to run until they "smoke"; the object of thermal-failure-mode-analysis is to identify suspect components, not to fry the gear!

Vacuum tubes can also become quite hot in operation. Hot tubes should be removed with a hot-tube-puller, such as the GC Model 5098, or 5099 (right-angle version).

A SAFE WORKING ENVIRONMENT
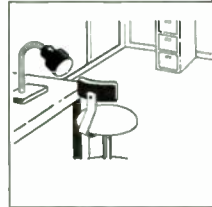
 A shop or studio should be laid out so that aisles between equipment and storage units are wide enough to allow unrestricted passage. Protruding fixtures that might snag loose clothing should be repositioned if possible. Electrical and electronic cables should be kept out of walkways. If this is not practical, then the cables should be protected under door-threshold molding, or within molding especially designed for wiring, such as the S L Waber Cordguard™ products.

Important safety procedures should be posted in a conspicuous place. A first aid kit should be available; it should be restocked periodically as the contents age or are depleted. Suitable kits are carried by hardware and discount stores.

Children and pets should not be allowed in the work area—especially in areas where soldering irons, power tools, and chemicals are present.

Safety training and first aid skills are important additions to the electronic musician's repertoire. Safety information is available from regional offices of the Occupational Safety & Health Administration. First aid courses and reference materials are available from local Red Cross chapters, and CPR and emergency medical training courses are offered by many community colleges. Some phone books also include first aid information in the first few pages, to which one can refer to in an emergency.

A concise first aid reference to put in the spare parts/tool box you take to gigs or your first aid kit is the *Reader's Digest Handbook of First Aid.* This little booklet was published in tear-out form in the March '87 issue of *Readers' Digest.* Reprints are available for $1, postpaid: Reprint Editor, *Reader's Digest,* Pleasantville, NY 10570. **EM**

The "MIDIPrint" MIDI data analyzer for the Commodore 64 was one of the most popular programs we've run in EM— and now it runs on the Atari ST.

MIDIPrint for the Atari ST

BY WALTER DANIEL

I recently found a need for Jim Johnson's useful program, MIDIPrint (Sept. '86 EM). Unfortunately, the only computer to which I had access was an Atari ST, and MIDIPrint was written for the Commodore 64. After some experimenting, I discovered that one could translate the C-64 version of MIDIPrint rather easily into Atari BASIC and run it on a 520 or 1040 ST. It also turns out that the ST program is shorter and simpler. Here are the changes to the original listing. An explanation of how it works, and why it is simpler, follows.

THE CHANGES

Boot your ST with a language disk and double click on the BASIC.PRG icon. Refer to the ST BASIC manual (comes with the computer) about program entry, execution, and so on. Dig out the September 1986 issue of EM (if you don't have that copy, back issues are available for $3.50 from Electronic Musician, 2608 Ninth Street, Berkeley, CA 94710), turn to Jim's Commodore code and do this:
Delete these lines:
150, 170-210, 230-240, and 1050-1130
Add these lines:

Walter Daniel is just another struggling writer in the Washington, DC area who should be out looking for a real job. His Compuserve ID is 75066,164.

> **"A**n analysis of a friend's electronic drum set, for example, verified that the instrument sends only eight different velocity levels instead of each integer available from one to 127"

```
170 CLEARW 2:FULLW 2
180 PRINT "MIDIPRINT ST":PRINT
190 PRINT "CONTROL-C TO QUIT"
230 S=INP(-3):IF S=0 THEN 230
235 BYTE=INP(3):BYTE=BYTE+256
240 IF BYTE=254 THEN 230
245 IF BYTE=248 THEN 230
```

Save the program to disk before running it so that you can recover your work in case the computer crashes or your neighborhood is subject to Electromagnetic Pulse.

HOW IT WORKS

This is what the changes do. Most of the program operates just like the original C-64 version. Lines 170 to 190 clear the output window and expand it to fill the screen, then print the opening message. Line 230 checks the status of the MIDI port. If a byte has been received, the program goes to line 235 to actually get the byte. Lines 240 and 245 filter active sensing and clock bytes, respectively. From there, control passes to the original statements for interpreting the message and displaying it on the screen.

Doesn't something seem too good to be true? Somehow the machine language routine that the C-64 required has been replaced with a few lines of BASIC. Isn't BASIC too slow to capture all this data coming into the computer? Ordinarily, yes, but ST BASIC seems to have hidden some of its capabilities behind an unassuming construction.

THE INP STATEMENT

By now, you've no doubt spotted the suspicious-looking statement in lines 230 and 235. INP(n) returns the number received by port number n or, if n is the negative of the port number, it returns the status of the port. The MIDI port number is 3. Still, this is only a BASIC command, so it theoretically should be too slow to keep up with data transmitted at the fast rate of the MIDI protocol—at least I thought so until I experimented with INP.

Type in this one-line program:
10 B=INP(3):PRINT B:GOTO 10
Connect a MIDI cable from the MIDI Out port of a MIDI-equipped instrument to the ST's MIDI In port. Run the program, then play a note on the instrument. Even though this simple program does not check the status of the MIDI port, it prints the number associated with note-on status, note number, and velocity value with one difference: these numbers are negative. Adding 256 to these numbers gives the proper output. I haven't the faintest idea why the numbers are negative, nor does the ST BASIC manual shed any light on the matter. But it does work.

> **"**Theoretically (a BASIC command) should be too slow to keep up with MIDI data...at least I thought so until I experimented with INP"

Another thing to notice about the experiment is that if you play several notes, the program keeps up with the data stream. Apparently, the INP instruction buffers the data until it can be handled by the program. As with the C-64 version, there seems to be some limit as to the size of the buffer, so don't overdo it when analyzing data streams. (Of course, someone will dump an 8 megabyte sample to MIDIPrint, the ST will melt, and I'll get blamed. Well, it's *not my fault!*)

The INP command checks its own status and buffers its own data. In fact, to exit the one-line program, a CONTROL-C alone will not work. After a CONTROL-C, the MIDI port must receive a byte before it will return control to the keyboard. I included a status check in the modifications above so that you don't have to bother playing another note to regain control of the computer.

HOW TO USE IT

I analyzed a DX7's output as Jim Johnson described in the original article and obtained the same result when I played the lowest C on the keyboard. As Jim suggested, the program gives you useful English versions of the MIDI data that normally hides inside your synths. An analysis of a friend's electronic drum set, for example, verified that the instrument sends only eight different velocity levels instead of each integer available from one to 127.

It's nice to find that a serviceable program can be translated from one machine to another. I just hope that one day I can unravel the mysteries of the INP instruction.

(Thanks to Larry Kolota for the ST used to develop this translation.) **EM**

Has the reflection of the LEDs flashing against the brushed aluminum blinded your muse? Don't lose sight of the basics of musicianship.

The MIDI Virtuoso

BY E.A. HOLLEY

The idea of virtuosity doesn't seem to sit well with the cold science of MIDI and electronic music, does it? The artistic excellence suggested by the one seems uncomfortable alongside the mechanical precision of the other.

As we know, every art has an element of the mechanical to it—an artist's paint formula; a pianist's finely crafted keyboard; a synthesist's silicon chip. But in recent years the scientific and technological part of music has zoomed past the ability of many people to know what to do with it all, leaving much of the artistic part undeveloped and minimized. Here are some tips that might help bring an element of the virtuoso to your MIDI music.

First Tip: Remember the haiku. No, it's not the sequel to *Shogun*. It's a three-line Japanese poem with just 17 syllables; five on the first and last lines, and seven sylla-

E.A. Holley is currently on staff at Delaware State College where he teaches English and Communications. The owner of Earth Studios, Mr. Holley is designing a class in electronic music to be offered at Delaware State College. He also does consulting with other schools to establish workshops and courses in electronic music.

> **"Think of your collection of presets sounds, not as a library, but as a band or orchestra, with no more than two or three musicians in each section"**

bles in the middle. That's all. No long epic here, buddy. But if you can write within those limits, what you *do* say is so clear and distilled that it still packs a big punch.

Limits. That's the haiku. What has this to do with MIDI music, you ask? Good question. In electronic music, where suddenly any sound that can be heard can be played musically, you should set certain limits. It'll bring out the virtuoso in you.

Second Tip: Limit yourself to a specific instrument or keyboard setup. Don't use an open-ended, constantly changing collection of fickle contraptions that come and go with the ads in the trade magazines. Great baseball players always use one favorite bat; great bowlers, one favorite ball; great lovers, one favorite—well, you know what I mean.

Take plenty of time to understand your needs, assemble a system that works for you, and then commit to it! This doesn't have to be just one synth, but certainly no larger a collection than could fit into the back seat of a Yugo. Don't even think about new instruments for awhile; choose wisely enough that you could use your system of choice for the next ten

years and still be satisfied.

Third Tip: Limit yourself to a specific number of instrument sounds. There's no need to re-program each instrument voice each time you play. Remember the Beatles? The Everly Brothers? How did they make all that great music with only two or four instruments and no 32-track, 16-bit digital microprocessor in sight? They focused on what was available to them, accentuating the *emotion* in the music while playing down the technical side.

Think of your collection of preset sounds, not as a library, but as a band or orchestra, with no more than two or three musicians in each section. In your string section, for example, you may want to use a solo string, plus an ensemble string voice, and maybe a string sweep that's not acoustic-sounding at all. For a bass section, develop a "realistic" acoustic upright sound, an electric bass for pop songs, and maybe a fat "Moog" bass for those dance tunes—no more. Use the same process in choosing the other "players" in your orchestra. Tweak a part if need be to fit into a particular rhythmic groove or sonic space, but there's no need to start from scratch each time.

Next, explore and develop the playing technique that each of your "players" might use with his individual instrument voice. With a bass sound, for example, explore the best way to bend with the pitch wheel. How can you use the glissando or portamento to be more expressive? What keyboard velocity setting provides the best cross-fade or slap-bass attack? How can you make the bass sound other-worldly and not like a traditional bass at all? These and all the other considerations involved in developing a *good* technique on an instrument require that most of us work with only a few instrument voices. After all, a Steinway is pretty much a preset instrument and astound-

ing music has been made with it for generations.

Fourth Tip: If you *do* incorporate synthesizer programming into your musicianship, try analog. I recommend starting your programming odyssey with analog instruments (the ones that you can program in real time with buttons and knobs) before getting into the more confusing digital synths (where everything is done with numbers and charts). Why? Because programming in real time is more intuitive and far easier to learn and understand. Each parameter has a corresponding knob, switch, or button that can be varied in real time—you can learn simply by doodling with the buttons. You might find that turning the second knob

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"Digital synths offer much less chance for serendipity"

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in the third row makes the sound brighter. Who cares what it's called? The older Sequential series (Prophet 5, 600 and T-8) are hard to beat as programming synths. Roland has always provided real time programming on their Jupiter, Juno, and JX model synths as well.

Not so with programming many digital machines. To enter program parameters one at a time in step mode, you must first have some idea of what each parameter does before you can punch it up and alter it. Even if you do stumble upon a sound that works, it's difficult, especially for the beginner, to understand exactly how it was created. Digital synths offer much less chance for serendipity and happy accidents.

In any case, whether you're strictly a preset player, or whether you play analog or digital, some knowledge of analog programming is essential not only to take full advantage of any electronic instrument you might use, but to help you understand digital and sampling techniques, because it's the granddaddy of both.

Many musicians believe that new music requires new sounds, but apply some of these techniques to your programming and playing style and you'll probably be making better music with less hardware—and that's adding the element of the virtuoso. **EM**

Caught between drum machines in the fast-channel world of MIDI sequencing? Don't suffer needless embarrassment, just fire up old faithful . . .

Casio's New Drum Machine—The CZ-101?

BY LACHLAN (LUCKY) WESTFALL

There was a time when a musician's greatest worry in a recording session was running out of tracks. In most cases the solution was some strategic bouncing or even better, a little forethought. In the age of MIDI, this problem has undergone a significant transformation. Although—or more accurately, because—the number of available tracks on a MIDI sequencer these days is often unlimited, we find ourselves running out of *instruments* more often than tracks. Unlike tape recording, there is usually no solution to this problem that costs less than $500, and because of this constant need for more instruments, one synthesizer has become indispensable in all but the most aloof of MIDI setups. The Casio CZ-101, with its four-voice multi-timbral capabilities, has proven itself to be the workhorse (or "volks-synth" as it has been called) of many a MIDI studio. Want to play a bass line and melody? No problem. The CZ-101 can do that with two voices to spare. I use three CZ voices to cover the kick, snare and miscellaneous percussion, and leave the fourth open to play a bass line, a second snare drum sound or a melody line. To accomplish this, you must set the CZ-101 to receive four monophonic parts simultaneously on four MIDI channels.

THE WONDERS OF MULTI-TIMBRAL MODE

To set the CZ-101 to multi-timbral mode, first press the "MIDI" button. The synth will display its basic channel like this:

Lachlan Westfall is a freelance writer, avid Macintosh fanatic, and the director of the International MIDI Association.

What's Mono Mode All About?

Mode 4, also called Omni Off/Mono mode or sometimes simply Mono mode, is a very powerful MIDI mode—but many people find it confusing, so pay close attention.

With Mode 4 selected, *each* synth voice is assigned to its own MIDI channel number. For example, with a six-voice synth tuned to basic channel 1, the first voice responds to the data coming in over channel 1, the second voice responds to the data coming in over channel 2, and so on until the sixth voice responds to the data coming in over channel 6. Synthesizers with "multi-timbral" Mono capability also allow each voice to be set to its own sound program. Therefore, each channel can access a particular voice via MIDI, and this voice can be programmed for its own timbre.

Within certain limitations (such as whether pitch bend affects all voices or just one), multi-timbral synthesizers and MIDI sequencers make a great team, since a single instrument can produce several independent melody lines that respond to information coming in over different channels. Therefore, if the sequencer has a bass line programmed in sequencer track 1, a horn line programmed in track 2, and a violin solo programmed in track 3, you could send the data from track 1 to the multi-timbral synth's first voice (which would be programmed for a bass sound, of course), the data from track 2 to the multi-timbral synth's second voice (programmed for a horn sound), and the data from track 3 to the multi-timbral synth's third voice (programmed for a violin sound). Since a single multi-timbral instrument can provide multiple single-line melodies, each with its own timbre, it offers more flexibility than standard synthesizers. Chords can also be created by using more than one sequencer track and more than one instrument voice. In the example given above, sequencer tracks 4, 5, and 6 could be programmed for triads; assigning these to voices 4, 5, and 6 of the multi-timbral instrument—and setting voices 4, 5, and 6 for the same timbre—would produce chords where each note has the same timbre.

Note that some synthesizers provide *bi-timbral* operation by letting you split the keyboard into two (or more) sections, whereupon you can assign each half of the split to a different MIDI channel. For example, if the lower half of the split is set for a bass sound and the upper half of the split for a trumpet, each could be driven polyphonically from a separate sequencer track over separate MIDI channels. Some bi-timbral synths can handle separate pitch bends for each side of the split, but others cannot.

—*Craig Anderton*

(Excerpted with permission from *MIDI For Musicians*, published by Music Sales—available from Mix Bookshelf, 2608 Ninth St., Berkeley, CA 94710 ☎ 415 / 843-7901; write or call for free catalog.)

MIDI BASIC CH=01

Then press the "solo" button. In addition to the basic channel (CH=01), the display now shows the VO (VOice) parameter indicating to which MIDI channel the *currently active voice* will respond, like this:
MIDI CH=01, VO=01
In solo (multi-timbral) mode, the four voices on the Casio respond to data on the basic channel (N) as well as N+1, N+2, and N+3. So when the basic channel is 1, as in this case, the instrument will respond, monophonically, to data received on channels 1, 2, 3, and 4. At this point, VO displays the same number as the CH parameter—the basic channel, N. Select a voice as usual using programmer buttons 1 to 16, and that particular voice will respond to MIDI data on the channel that corresponds to the VO number.

To select voices to respond to channels N+1, N+2, and N+3, you will need to increment VO respectively. To make these assignments, move the cursor to the VO display by pressing the right arrow cursor key. Press the up arrow "value" key to change the VO parameter to a number indicating N+1; the display shows:
MIDI CH=01, VO=02
Select a voice on the synth and it will respond to data sent over the MIDI channel indicated by the VO number, in this case, channel 2. The third and fourth voices are assigned in the same fashion, by pressing the up arrow to the next VO number and selecting the corresponding voice.

When all four voices have been assigned, press the "solo" button twice (don't ask me why but it eliminates some problems) and you are ready to go.

Note that the VO parameter shows only MIDI channel numbers N, N+1, N+2 and N+3, not actual "voice numbers." The "VO" label sometimes causes confusion here; what the display is trying to get across is that a particular voice number (not displayed) corresponds to a particular MIDI channel number (i.e. the VO number).

MIDI MACROS, TRANSPORTABILITY, AND SOUNDS

Unfortunately, these setups, along with the instrument's basic channel, are forgotten when the machine is turned off. To get around this, I set up a kind of "MIDI Macro" in my sequencing program. To set the CZ-101 up via MIDI, all you have to do is select the basic channel, press "solo," and then have the sequencer send MIDI program change data on the four desired channels.

Drum patterns created in this manner are very transportable. Because the drum sounds are triggered by standard MIDI note data, they can later be used to trigger any MIDI drum machine or sampling keyboard set up with percussion sounds. Just put all the percussion tracks on a single MIDI channel, transpose (or remap) the notes so that they trigger the correct drum sounds on the drum machine, and there you have it.

Programming drum sounds on the CZ can be fun. For instance, the white noise generator is useful for creating snare sounds. In addition, on the CZ you can simultaneously play white noise and a tone with which you can synthesize the resonance of the drum shell. Kick drums and toms are fairly easy to simulate and the ring modulation effect is good for cowbells and other metallic types of percussion. The eight-stage envelope generator can create interesting attacks such as a quick downward pitch shift at the beginning of the sound which is characteristic of a tom and other percussive instruments. So use your imagination and your ears to create other unusual percussive sounds.

This "mock drum machine" technique is not limited to the CZ-101. All the other synths in Casio's CZ line can receive data on multiple MIDI channels, with the CZ-3000, CZ-5000 and CZ-1 offering eight voices instead of four. Further, with a sequencer capable of sending in MIDI Mode 4 (Omni Off/Mono), these synthesizers can play various combinations of monophonic *and* polyphonic MIDI data up to the instrument's voice limit. Thus three monophonic percussion voices, a melody line, a bass line, and a three-voice polyphonic string voice can all be played simultaneously on a single eight-voice instrument. More and more synthesizers are implementing MIDI Mode 4, including Sequential's Prophet 2000/2002 and Prophet VS, Ensoniq's ESQ1 and Oberheim's Xpander and Matrix-12.

I hope you like these sounds and if you come up with a good pseudo hi-hat, let me know! **EM**

Envelopes used to be simple...now they're complex. But that complexity yields new opportunities, once you realize what these sweeties can do.

Envelope Intimacy and the Art of Shaping Sound

BY CLARK SALISBURY

You may not know it, but the envelope is your friend. It can help smooth out the dynamics of a balky string patch, put the bite into a too-tame clavinet simulation, or give your slap bass patch just the right pitch bend to kick "Feel The Heat" right through the roof.

But it seems that a lot of us don't get everything *out* of envelopes that's been put into them, and it's simply that they seem so darn complicated. Four stage envelopes, eight stage envelopes, positive modulation, negative modulation—you'd think that envelopes were designed by the same guys who design tax codes. But they aren't all that tricky. Really.

ENVELOPE BASICS
Probably most of you are familiar with a general-issue, four-stage envelope generator (EG). This type of EG is called "four stage" because it has four basic functions: Attack, Decay, Sustain, and Release (ADSR). *Attack* controls attack time (how long it takes for the envelope to go from full off to maximum level), *decay* controls decay time (how long it takes for the envelope to go from maximum level down to a specified sustain level), *sustain* controls the level while a note is being held (key down), and *release* controls the final

Clark Salisbury is a studio composer and programmer, a partner in The MIDI Connection, a Portland, Oregon-based consulting firm, and this year's president of the Portland Computer Arts Resource Center. His favorite color is chrome.

> **"You'd think that envelopes were designed by the same guys who design tax codes. But they aren't all that tricky. Really."**

decay rate of the envelope once a note has been released.

"But Clark," I hear you saying. "Attack time of what? Sustain level of what?!" And that, dear reader, is the beauty of envelope generators. They can control almost anything.

In the old days (four, maybe five years ago) most synthesizers were supplied with one or two envelope generators. If you were fortunate enough to find *two* whole ADSRs on your Jupiter Four or Prophet Five, you'd discover that one was hard-wired to the voltage controlled filter (VCF) and the other was hard-wired to the voltage controlled amplifier (VCA). You found that whenever you twisted the attack knob of the EG connected to the VCA, the time it took for the *volume* of a note to reach a peak would change. Likewise, the attack knob of the Filter's ADSR controlled the time it took for the sound to get *brighter* and *darker* (never mind about the EG connections to the oscilla-

tor's pitch or pulse width for now. I'm making a point). The simplicity of this situation may have eliminated a lot of time-consuming guesswork about what your EGs controlled, but limited your flexibility a lot.

MODERN MULTI-STAGE MEGA ENVELOPES
Now consider the present crop of electro-marvels. We have EGs that will control nearly anything in a synthesizer. In many cases they're no longer hard-wired to a specific component, which forces you to decide what the EG should control in any given program. Why, even the low-priced Casio CZ-101 can have, in the Tone Mix mode, as many as 12 envelope generators going at once, each of which can have as many as eight stages.

To further confuse matters, on most machines you don't get a nice tidy knob to fiddle with anymore. At best, you'll probably just get a single slider or worse, the dreaded pair of *parameter buttons*. With either one you have to enter a set of numbers describing envelope values in a decidedly non-intuitive way. Let's just hope nobody comes up with a way to reduce all of the functions of a synthesizer to a single button.

Faced with all this, how does one become intimate with the modern envelope generator? Let's take a quick look at a couple of applications for these multi-purpose wonders.

If you happen to have a Casio CZ series synthesizer, you're in envelope heaven (or hell, depending on your point of view). If you own something else, no problem...an envelope is an envelope

is an envelope, more or less, and the following examples will work on almost any synthesizer equipped with multistage envelope generators.

The CZ synthesizers give you six envelope generators with which to work. Each DCO (digitally controlled oscillators) is controlled by its own EG. Each DCW (digitally controlled waveforms, a

> **"**Let's just hope nobody comes up with a way to reduce all of the functions of a synthesizer to a single button"

feature of Casio's phase distortion synthesis, and roughly analagous to the filters in analog instruments) also has its own EG, as do each of the two DCAs (digitally controlled amplifiers). The CZ envelopes consist of eight individual stages. You are not required to use all eight stages if you don't need to (merciful God), but they are available if you can think of something to do with them.

You can describe each stage of the envelope with a pair of numbers labelled "Level," (representing *how far* that stage changes a parameter) and "Rate," (*how fast* the change from the previous level to the current one happens). In other words, if you want to set a fairly long attack time, set the level for stage one to something greater than zero (i. e. however high you want the parameter to change) and the rate to a fairly low number (0 is the slowest rate). This arrangement is becoming increasingly common on a number of instruments. It gets even more interesting when you discover there's no law that says the value for any particular level must be greater or less than that of the previous level; the envelope can go either way.

APPLICATIONS
What these options mean is that you can design all manner of bizarre envelopes: ones with multiple attacks, multiple decays, manic roller coaster dips and climbs—

all set with a couple of silly data entry buttons. One of my favorite settings yields what I like to think of as "duration sensitive echo." With this, you can create a pseudo-echo effect that becomes apparent only if you hold the note long enough. Set up the envelope you see in **Fig. 1a** for the EG that feeds the Casio's DCAs (or the equivalent amplifier circuitry in your instrument).

This envelope exhibits a number of percussive attacks, each rising to a successively lower level, and each decaying at a moderate rate. The reason I call this a "duration sensitive" envelope, other than

making me sound like I know something you don't, is that if you strike and release the keys quickly, the envelope never has a chance to reach its second stage; it immediately jumps to its release stage and decays normally. But hold the keys for a longer duration and the envelope will repeatedly fire the DCA, which creates the pseudo-echo effect.

Of course there are variations. What would life be without variations? Spaghetti without meatballs.

Start with a fairly low level in stage one and successively increase the level of each stage (**Fig. 1b**). The effect will be of
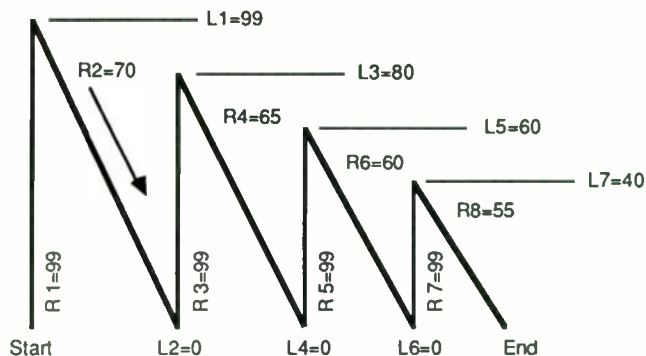
Figure 1a:
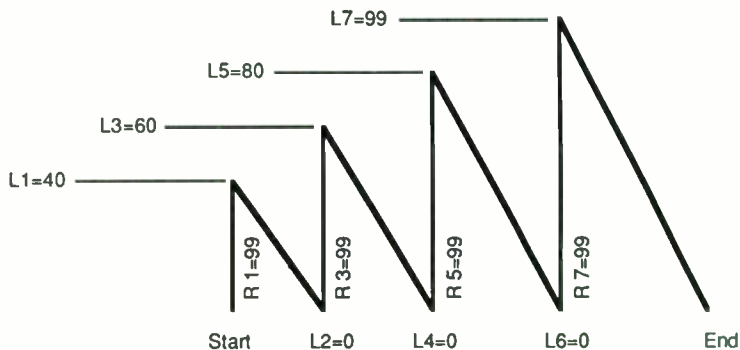
Figure 1b: Figures for rate and level are for Casio CZ. Values given may vary for other machines.

Figs. 1a and 1b *"Duration sensitive" envelopes*

When is a Rate not a Time?

One of the confusing aspects of modern-day envelopes is the relationship between envelope *rate* and *time*. Generally, you want to describe how much *time* it should take for an envelope to go from one level to the next. However, with most multi-stage envelopes you have to specify the *rate of change* between two levels, which means that the time it takes to get from a starting level to one destination level may be quite different than the time it would take to get to a *different* destination level.

Consider **Fig. 3**, which shows three envelopes. In envelopes 1 and 2, the rate of change from level 1 to level 2 is constant. Note, however, that with the second envelope it takes much longer to change from level 1 to level 2, even though the *rate* of change between levels is identical. Therefore, as you change levels in an envelope, it is usually necessary to change the rate between those levels as well in order to maintain a constant time change between levels (envelope 3). Note that some voice editing programs for instruments with multi-stage envelopes actually recalculate the rate between levels to maintain a fairly constant time when levels are changed.
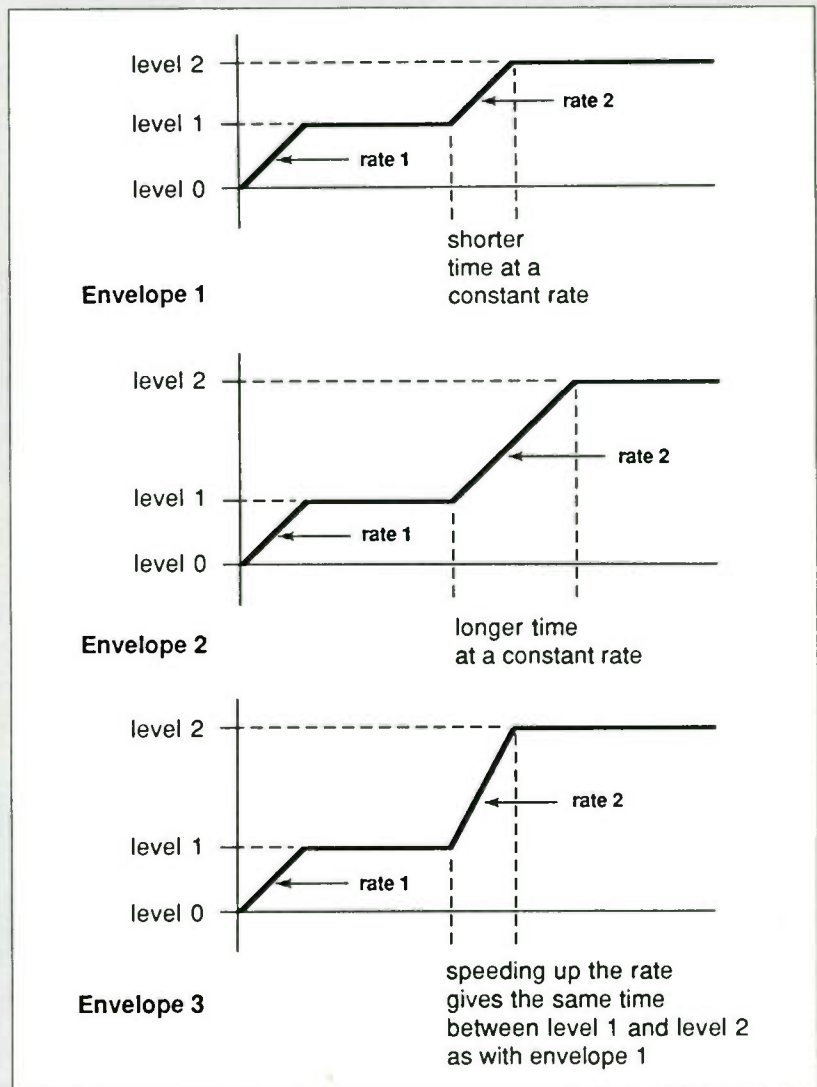
Fig. 3 *Keeping rate 2 constant, but increasing level 2, means that it takes more time to go from level 1 to level 2. Speeding up the rate shortens the time required to go from one level to another.*
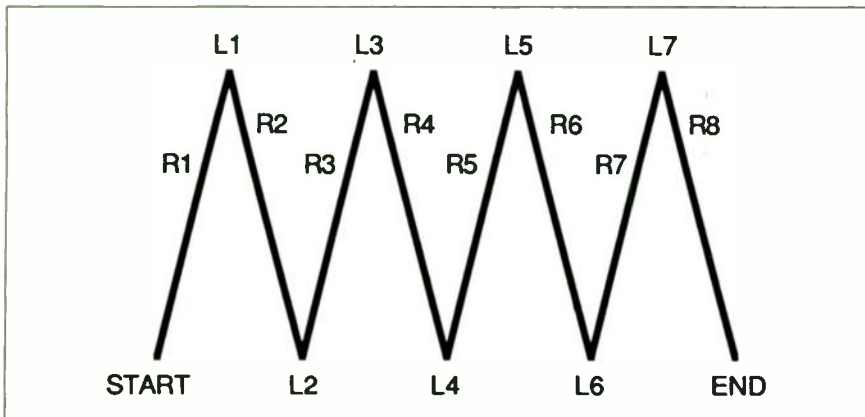
Fig. 2 *Envelope for creating a brass "buzz"*

the echos getting louder as time passes. Apply similar envelopes to the DCW or to whatever corresponds to the filter on your synthesizer.

For instant Space Invaders, set up an identical envelope to control the DCOs. The pitch of the oscillators will fall or rise, as the case may be, in time with the echos.

When a brass player plays a note, the first thing you hear is the initial buzz of the player's lips getting up to speed. Create this effect for solo brass sounds (as well as some nice stab brass effects) by using an envelope to modulate the pitch of the DCOs (or VCOs) rapidly for an instant at the beginning of the sound. Try something like the envelope in **Fig. 2** applied to your synthesizer's DCOs. If your synthesizer doesn't allow you to modulate the oscillator's pitch from an envelope, don't despair. Modulating the filter with this type of envelope can be just as effective.

A fun variation on this theme is to use a pitch envelope to create automated trills. This type of effect is only possible with envelopes that use *time* rather than *rate* parameters, so you can't do this one

on a Casio. The problem is that you must be able to specify how long the envelope will stay at one level before going to the next, rather than just set the rate of change between two levels.

Anyway, what I've set up here is a pitch envelope that, at the beginning of each note, repeatedly pushes the pitch of your oscillator(s) up maybe a whole step, holds it there for a moment, then brings it back down again—what a trill. You may

need to do a little tweaking on your machine to come up with the appropriate pitch values. (Maybe this is not an exceptionally useful patch, but then old Ben Franklin probably never envisioned microwave ovens and electric socks when he took up kite flying.)

You can vary this envelope by applying it to the filters or amplifiers in your instrument. By working with the "time one" values you should be able to create sounds with a variety of delayed attacks. And by working with the "time two" values, you should be able to control how long a sound sustains, regardless of how long you hold a note. Try blending some of these sounds with other sounds having more "normal" types of envelopes—either through MIDI or simply by using another channel of your synthesizer.

These aren't the only tricks to be done with envelope generators, just some of the more obvious things I've discovered. Experimentation, as always, is the key to understanding. So get up, grab those parameter buttons, and do the envelope generator boogaloo.   **EM**

They say you no longer have to be a "slave to the rhythm" and now it really is possible for sequencers and drum machines to follow human timekeepers rather than the other way around.

Kahler's Human Clock

BY JIM JOHNSON

Synchronized music has come a long way in the last few years, but a lot of musicians see the growing presence of programmed drum parts and sequenced accompaniment as a step in the wrong direction. While most drummers have learned to at least put up with drum machines—if not love them—many feel slighted when asked to "just play fills over the beat, okay?" In addition, rock musicians often have difficulty following the rigid tempo of a computer-generated clock, even when they *can* hear it over the music. (In fact, a mean and ugly bass player once threatened my life, insisting my drum machine's "sloppy timing" was the cause of his inability to follow the beat.)

Kahler's new Human Clock is designed to solve all these problems. Instead of having a machine dictate the now-proverbial "cold, computerized, machine-like" tempo, the Clock lets your sequencers and drum machines *follow* the tempo set by your drummer or any other electrified musician in the band. Unlike the usual "tap tempo" clocks that require the drummer to provide a more or less steady stream of quarter notes, the Human Clock can derive a tempo from just about any

In between writing, programming, and writing software for Dr. T, **Jim Johnson** *occasionally finds time to eat and sleep. His original musical goal was to become a renowned Dixieland trumpet player; he modified his aspirations when his high school bought a PAiA modular synthesizer.*

Product Summary

Product: Human Clock
Type: Machine-to-live playing MIDI synchronizer
Retail price: $650
Size: One standard rack height
Input impedance: 100k
Input level: -50 dB to 0 dB
Manufacturer: Kahler, PO Box 9305, Anaheim, CA 92802 ☎ 714 / 632-5280

bass or snare rhythm, within reason, that a drummer might play—and output MIDI timing clock data to your MIDI-compatible devices.

The Clock is housed in a single-height, 19-inch rack package. Six knobs on the front panel—*Sensitivity, Smooth, Advance, Feel, Level* and *Mask*—tune the instrument to the drummer's style. Two push buttons labeled *Restart* and *Reset* are the only controls the drummer needs to touch in a performance: footswitch jacks on the rear panel duplicate these functions. The back also has a single instrument input (parallel wired to two jacks), a switch labeled *Perc/Sust*, and of course, a MIDI output jack. Power and trigger indicator LEDs appear on the front panel.

Operation is fairly simple once the Clock is tuned in to the drummer's style. Reset clears and prepares the Clock for a new tempo. On the drum(s) that will drive the Clock, two beats played four quarter notes apart set the initial tempo, and the clock starts on the third hit.

Obviously, this means that the drummer will have to be a bit more careful about starting a song, but no more so than synthesists who have to start up a bunch of sequenced instruments. Restart starts the Clock without changing the tempo from its previous setting.

THE CONTROLS

You will never get optimum results from the Human Clock unless you spend some time reading the owner's manual and fiddling with the knobs. The Level control sets the level of the signal going into the Clock's trigger extraction circuitry. Adjusting it to the lowest value that produces reliable triggering can be tricky, since too high an input level can cause double triggering—a disaster when setting the initial tempo. The Mask knob sets a time delay after each trigger in which the Clock will ignore new triggers; this helps reduce retriggering. The Perc/Sust switch on the rear panel allows use with sustaining instruments like bass guitar.

Smooth and Sensitivity control the way the Human Clock tracks the drummer's tempo. Because they provide functions that are new to electronic musicians, they require more effort to comprehend than the other controls. Sensitivity affects how much of a perceived tempo change is actually fed to the Clock's microprocessor. For example, at a low sensitivity setting the Clock will tell itself that any large tempo changes it sees are really only small tempo changes played by a sloppy musician.

Smooth consists of the two parameters, *Window* and *Speed*. Window sets the

amount of time around each eighth note in which the Clock looks for a valid beat. Wider windows follow more drastic tempo changes, but are more apt to go crazy on triplet patterns, while narrower windows work better with strange rhythms but won't follow drastic tempo changes as well. Speed controls how quickly the Clock responds to tempo changes, and can compensate for uneven playing. Three different windows, and two speed options, give a total of six Smooth settings. Advance pushes the clock forward by a number of MIDI clock pulses, while Feel delays or advances the MIDI clock output relative to the drummer's beat. These controls can help correct for any timing delays you encounter, or fine-tune the "feel" of a song.

HOW DOES IT KNOW?

Technophiles like myself are always curious about how gadgets like this work, so I called Kahler up and asked them a few questions about what makes the Clock tick. (Sorry about that one.) Understandably, they were reluctant to release too much information on the Clock's inner workings, but I did manage to get a few details. Rather than simply measuring the time between drum hits and directly updating the tempo, the Clock uses its tempo to predict when the next eighth note should occur, and analyzes the difference between the *predicted* time and the *actual* time to adjust the tempo. What this means is that the Clock bases its tempo not on *instantaneous* changes in tempo, but on the overall *trend* of the last few notes. This is why the Clock is able to track rhythms other than straight quarter notes.

Since the Human Clock is Kahler's first foray into electronics manufacturing (they are known principally for their vibrato tailpieces for guitar), I wanted to check out the quality of construction. The all-metal case opens easily by removing two screws that secure the top. Inside are four circuit boards (power supply, front panel controls, rear panel input/output connections and the 8618 microprocessor) plus some jumpers and a lot of empty space. Everything is easily accessible, so service should be no problem. The components and connectors are high quality, but one of the circuit boards in my unit was badly bent due to an improperly soldered LED. This is the type of problem that can lead to reliability problems further down the road, so I hope that Kahler tightens up their inspection procedures as they become more experi-

enced with electronic instruments. Other than that, the workmanship displayed in the Human Clock is as good as in any other piece of musical gear I've popped open lately.

SO DOES IT REALLY WORK?

Now that the technobabble is out of the way, let's talk about what matters—like, how does it play? For the most part, very well. Probably the best way to describe the Clock's performance would be to compare it to a technically proficient, but musically naive musician. If you're willing

to work with it for a while to help it become familiar with your music and don't throw it too many curve balls, it will be a definite asset in many situations. The drummers to whom I've spoken had varying opinions about the value of the Clock, but there seemed to be a definite relationship between positive response and the patience of the player. One dismissed it after about 15 minutes, saying, "it's a good idea; too bad it doesn't work yet," while those who were willing to spend more time with the Clock got very good results. One drawback is that the

The Complete Guide to MIDI Software

by Howard Massey and the staff of PASS (Public Access Synthesizer Studio)

If you need to make a decision on which software package is best for you, you need this book!

■ *THE COMPLETE GUIDE TO MIDI SOFTWARE presents an unbiased and in-depth look at 60 of the most popular software packages available today. The book features fact sheets on all of the packages, but was written after hands-on comparison-testing of each and every program. The book actually provides step-by-step guided tours to these programs with over 200 photographs of actual screens and detailed operator analyses of program ease of use, power and applications. Each "tour" also has a very useful summary which points out each program's special features and limitations.*

■ *Programs tested run on Apple II series, Apple Macintosh series, IBM PC and compatibles, Atari ST and Commodore computers.*
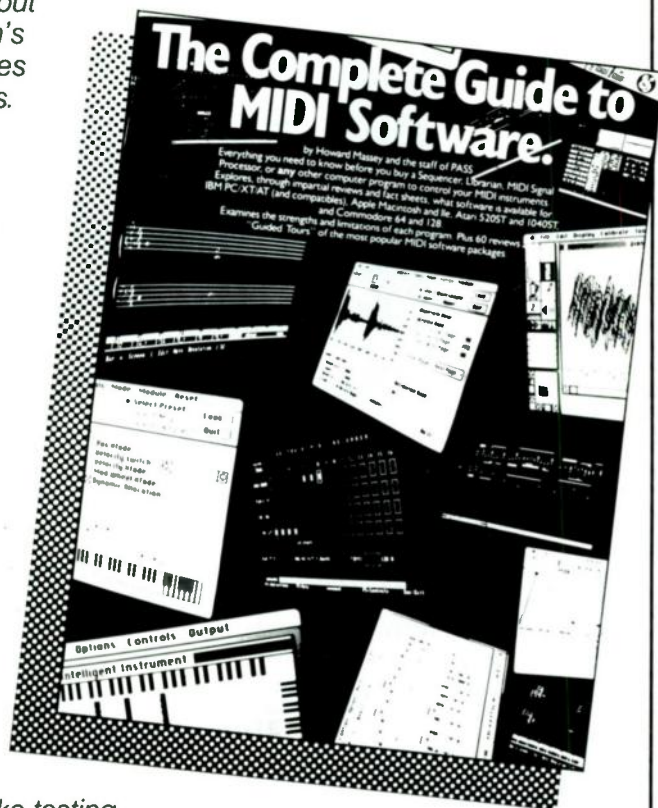
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Smooth control is only read when the Restart or Reset button is pushed, so in order to adjust this critical parameter one has to scream at the drummer to stop playing for a second, adjust the controls, reset the sequencer, and start over again—definitely a drag when working with impatient musicians. For drummers who approach the Clock as a musical instrument rather than as an effect box, this should be less of a problem.

Controlling the tempo from a bass synth was a little tougher for me, but for different reasons. One is that the Clock has trouble deriving a trigger from a sustained instrument if there's not a brief period of silence between notes; adding a noise gate, as the manual suggests, helps. The other is more subtle: I found it tough to overcome my tendency to follow the drum machine and make it follow me. But, with practice....All this is not to say that the Human Clock is infallible and any problems lie with the musician; there are also some quirks that should be taken into account for proper operation.

GLITCHES

One condition *guaranteed* to cause trouble is the drummer sitting out for more than one measure, then coming back in on a 16th note, rather than an eighth note, offbeat. Apparently the Clock needs to hear at least one hit per measure to keep track of the beat, though it continues to run steadily if the drummer stops playing. *Practicing* with the Clock before taking it into a band or studio situation will help identify any possible incompatibilities with your playing, whereupon you can make any necessary adjustments.

I have experienced a few software glitches. One is that the Clock seems to put out some kind of non-MIDI garbage occasionally on power-up; this caused my ESQ1 to display a SYSTEM ERROR message and somehow interfered with the screen display routines (!) in my MIDIed Commodore 64. Another problem was that if I pressed Restart immediately after pressing Reset (not the correct method of operation, but easy enough to do accidentally), then the next hit on the drum would start the sequencer running at maximum warp speed. This is the type of thing that makes drummers want to go back to logs covered with dead animal skins. The designer of the Clock, Michael Stewart, was unaware of these bugs when I informed him of them and promised to look into them right away. Kahler also

said that ROM updates will be made available for anyone who has a problem with these bugs. (Editor's note: As we go to press, the new ROM has just been released. It fixes the "coming back in on a 16th note after sitting out" problem, the restart/reset problem, and some of the other minor bugs mentioned in the review.)

The manual is pretty well-written, but the nature of the instrument makes it bound to be confusing the first time through. There are plenty of examples of how to set the Clock's controls for various playing conditions. One area I found not covered adequately is the device's MIDI implementation. By connecting my C-64 running MIDIPrint to the output, I discovered that when setting the initial tempo, the second drum strike sends a MIDI stop command, and the third strike sends a start command followed by the clock stream. Holding down the Restart button also sends stop commands continuously. This kind of trivia is important in some situations and should be included in the manual, but musicians who are less persnickety than I am about the technical details probably won't miss it.

OTHER NIFTIES

Some other nifty applications of the Clock include deriving a sync tone from a recorded drum part. If, for example, the bass drum is recorded on a separate track, or if some creative processing can derive a not-too-complex rhythmic pulse from another track, the Clock can read these and generate a sync to replace a missing or damaged sync track.

If you've ever come up with a spectacular part that you wanted to record into your sequencer, only to forget the part while trying to find the right tempo setting, then you'll appreciate how handy it is to have a machine that can take its tempo from your playing. Another novel application was used during the Clock's beta testing on Neil Young's most recent tour. A piezo pickup was attached to Neil's foot and a tempo extracted from that. This let him run his sequencers while playing solo acoustic guitar.

I only have one major complaint about the Clock: its price! At $650 list, it probably won't find its way into too many bar bands in the near future. But for a large studio or a touring band that uses a lot of sequenced instruments, the Human Clock could well become indispensable.

EM

Simmons goes MIDI in a big way.

Simmons MTM, TMI, and SDE

BY JIM FIORE

I t's about time. All that MIDI power; the MIDI flexibility that's been in the hands of keyboardists, computer composers and the like—it's finally coming to us drummers.

Thanks to MIDI, the role of the electronic drum kit has evolved from the ornamental, "space drum" effect generator into the drummer's mainstay. Simmons has just introduced three products aimed specifically at the MIDI drum market. The *Trigger MIDI Interface* (TMI) and *MIDI-Trigger-MIDI* (MTM) are interface devices that translate drum pad whacks to MIDI messages. The third, the *Simmons Drum Expander* (SDE), is a sound module. All three can be effective tools for the MIDI drummer.

THE TMI

Housed in a sturdy, single-height 19-inch rack chassis, the TMI accepts signals from drum pads and converts them to note-on messages for driving MIDI drum machines or synthesizers. The TMI has a single MIDI Out jack, but neither a MIDI In nor MIDI Thru. It has inputs for up to eight pads (or up to four Simmons stereo pads), each of which has its own front panel sensitivity control and LED that lights whenever a pad is struck hard enough to produce a MIDI message. Each pad input can be programmed to send a note-on message over any of the 16 MIDI

Jim Fiore, known to his friends as Jim, spends a portion of his time as an associate professor in Electrical Engineering Technology at MVCC in Utica, NY. In his off-hours he can be found sprawled out, semi-conscious, near his Amiga with old King Crimson tunes (particularly "Larks Tongues in Aspic") playing loudly on the stereo.

Product Summary

Product Name: TMI
Type: Eight-input drum pad-to-MIDI device
List Price: $455

Product Name: MTM
Type: Eight-input pad/drum-to-MIDI trigger device with effects
List Price: $1,150

Product Name: SDE
Type: Six-voice FM synth/expander
List Price: $895
Manufacturer: Simmons, 23917 Craftsman Road, Calabasas, CA 91302 ☎ 818 / 884-2653

channels, either by itself or grouped with other pads. A special global mode makes grouping of drums particularly easy.

A combination of eight programmed pads is called a "patch," and up to 50 different patches can be stored in the TMI and strung together in sequences containing up to 99 patches. The TMI stores up to eight sequences. You can use a footswitch to step through the sequences bi-directionally, and patch data can be off-loaded onto cassette tape. There is also an input for a Simmons Suitcase Kit, a multiple-pad setup in a single case.

In operation the TMI is quite straightforward and effective. Programming is relatively easy, although you have to cycle through all eight channels in order to change or update parameters. The dynamic response is not adjustable but you should be able to compensate for this with the velocity scaling on the unit the TMI drives. The footswitch/sequence operation could be very useful in a live situation, with each sequence representing, say, a preplanned set and each patch within the sequence a song or song segment. Parallel trigger output jacks make it a snap to trigger a MIDI device simultaneously with a non-MIDI drum kit (like a Simmons SDS8). The omission of a MIDI In jack is unfortunate. It would be very nice to cascade a couple of TMI for 16 or more pads, but this is impossible without an external MIDI blender.
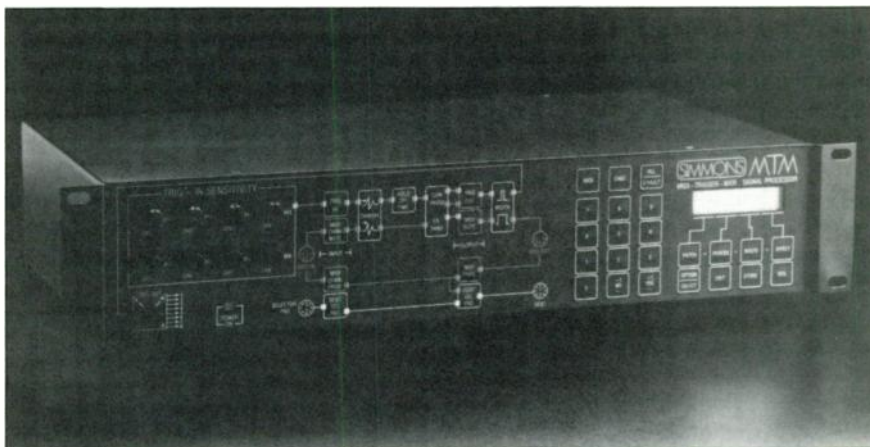
Simmons MTM

BIG BROTHER
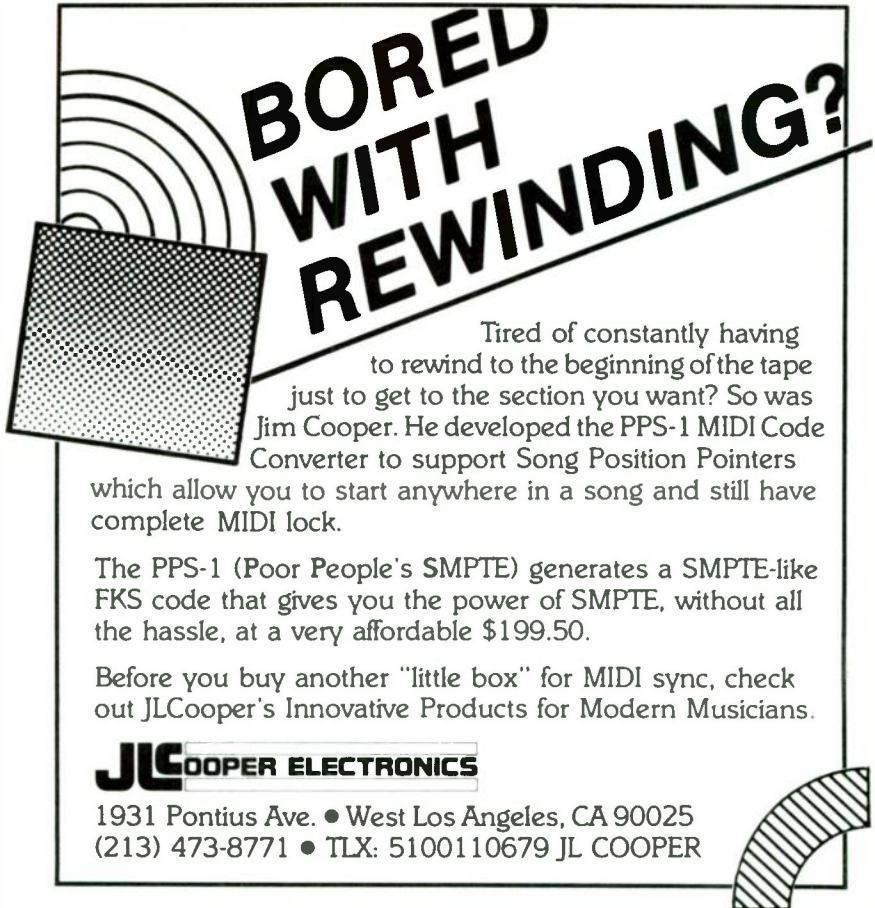
The MTM is like the TMI's big brother. It too can operate as a pad-to-MIDI converter, but includes many additional features. The MTM is a double-height 19-inch rack unit that sports a functional layout of its operation on its front panel. Besides pad-to-MIDI conversion, the MTM accepts triggers from acoustic sources (including pre-recorded drum tracks) and sends output-programmable triggers to electronic drum kits. Input trigger jacks are both ¼-inch and XLR type and accept Simmons stereo pads and the Simmons hi-hat pedal. A special connector is included for the SDS7 selector pad, and jacks are provided for cassette in/out, dual footswitch, and MIDI In, Out, and Thru.

To accommodate a variety of non-MIDI trigger sources, the MTM has an extensive input processing section. Input Gain can be set to either mic or line level. Absolute Threshold sets the level at which trigger signals become effective and works in conjunction with Percent Above Previous Threshold, which allows the MTM to ignore signals as they die away, thus minimizing false triggers. Channel Compare Threshold minimizes false triggers due to acoustic or mechanical bleed. Hold Off and Dynamic Hold Off determine the invalid trigger window. A number of different Dynamic Response Curves range from linear to exponential to downright bizarre (some get softer as you strike harder—great for velocity-controlled sample layering). Triggers received via MIDI do not require such extensive processing and can be altered via Absolute Threshold and Dynamic Curve.

Trigger outputs have a number of programmable parameters as well. Minimum Output sets the lower limit of a trigger amplitude, so no matter how soft you play, the output will not fall below this value. Pulse Width is programmable as is Dynamic Pulse Width, which makes the pulse width a function of how hard you play. MIDI Out triggers are programmed with the same three parameters. In this case the pulse width may be thought of as gate time. The set of parameters defining a particular trigger in/trigger out setup is called a Process. There are five factory and 15 user Processes.

Once processed, a trigger takes a "Route" to appropriate output jacks and is assigned MIDI note numbers. Twenty factory and 79 user Routes are available.

The MTM's Effects Section has ten factory and 30 user programmable Effects,

including simple echo (repeat); echo up and echo down (where each successive echo changes by a semi-tone); various forms of sequence (where each echo is a different note); and a number of chord effects, allowing chords (including major, minor, 6ths, 7ths, suspended and more) to be played in variations of Split (where each note has its own threshold—a soft strike produces low notes, a hard strike, higher) and variations of Layer (which is similar, except that the successive notes of the chord do not replace the earlier note, but sound together).

Combine a Process, a Route, an Effect, MIDI program change and SDS7 selector pad info, and you've got a Patch. The MTM can store 20 factory and 99 user Patches. The MTM has ten Sequences (a string of up to 100 Patches) that may be stepped through by footswitch.

REACTIONS
Programming something as powerful and complex as the MTM could be frustrating for the beginner, though the factory settings make for a good starting point, and Processes (perhaps the trickiest part) are provided for both Simmons pads and acoustic drum sources.

Triggering from pads was more reliable, had more dynamic range and faster response, and was more effective overall than triggering from live drums. Considering the complex waveforms involved this is not surprising. Triggering from acoustic sources can sometimes get rather nasty and there is a definite trade off between speed of response and false triggers. I tried to replace the snare from my drum machine with a sound on my sampler. When the MTM was adjusted to eliminate false triggers, it skipped a few notes in a roll; adjusted to get all the notes, it produced some double triggers. You'll need to experiment with settings and maybe do some equalization, compression or other processing of the source signal for best results. If you intend to mic the acoustic sound and blend it with the electronic sound, consider using pickups for the drum signal instead of making the mic do double duty. This is more expensive and yes, there will be more wires hanging about, but the pickup's signal should give superior performance.

The Routing and Effects sections of the MTM are very nice. The variety of echo and chord effects brings a new level to drumming. At last drummers can play real melodies, provide backing chords

and do things that would have been impossible just a few years ago, although a number of us are going to have to go back to school! One of the really nice things about this unit is that you won't have to tie up your DDL in order to get a slapback echo on your snare since the MIDI echo (a genuine echo, *not* just the "MIDI Thru-type" effect you may know from some sequencers) specifies both the number of repeats and the time between repeats.

There are many possibilities for triggering different drum sounds and patterns. You can set the unit so soft strikes will produce a snare while harder strikes produce an explosion; layer sounds or use three or four sounds instead of just two. The sequence/footswitch operation is very useful, and the programmable trigger outputs on the MTM—*light years* ahead of the TMI's simple parallel jacks—finally allow us to trigger a non-MIDI drum kit like the SDS7. All parameters can also be offloaded to cassette.

MTM MIDI
The MTM's MIDI implementation is good. The unit can send and receive over all channels, can ignore channel data and inhibit the send/receive of MIDI effects. A special controller change allows the SDS7 hi-hat pedal to function via MIDI. Program change is supported, and the MTM will function along with the SDS7 selector pad. Like the TMI, the MTM cannot be daisy-chained to increase the number of notes accessed. Overall, the MTM is a very versatile performer.

THE SDE
The SDE is like Simmons' answer to the DX7: a six-voice, FM synthesis expander unit with an easy, boiled-down programming technique. It uses familiar drum terms like "attack," "bite," and "brightness" instead of your typical "operator" business. The unit is a fairly compact, single-rack space high, with MIDI In and MIDI Out jacks, a cassette jack for off-loading to tape, a footswitch jack for stepping through programs, and a pair of audio outputs (one for voices one through three, and the other for voices four through six). The SDE also has provision for a front panel expansion cartridge.

Designed as a MIDI expansion device, the SDE has several different operation modes. For starters, it will respond in either omni-on or omni-off mode; program change toggles on and off, and

Roland has been in the guitar synthesis game for just about longer than anyone else. Has their research paid off with a reliable guitar-to-MIDI system?

GK-1 and GM-70 Guitar-to-MIDI System

BY CRAIG ANDERTON

I know most of you don't read reviews from the beginning. You jump to the conclusion, and see if that gives you an indication whether the rest of the review is worth perusing. So I'll spare you the suspense: the GK-1 and GM-70 are the best pitch-to-voltage MIDI guitar gadgets I've played. There are still occasional glitches, and occasional moments of sluggishness. But finally, here's an interface with which you don't have to pause and

Product Summaries

Product: GK-1 Synthesizer Driver
Type: Hex pickup-based guitar retrofit to drive Roland MIDI guitar products
Price: $225
Controls: Volume, balance, two assignable control knobs
Manufacturer: Roland, 7200 Dominion Circle, Los Angeles, CA 90040; ☎ 213 / 685-5141.

Product: GM-70
Type: Guitar-to-MIDI converter
Price: $895
Hardware requirements: Roland G-series guitar or guitar retrofitted with GK-1 Synth Driver
MIDI implementation: In, out, assignable controllers, mono or poly mode
Manufacturer: Roland, 7200 Dominion Circle, Los Angeles, CA 90040; ☎ 213 / 685-5141.

Craig Anderton enjoys music, non-poisonous animals, hot springs, traveling, pre-12 volt Volkswagens, the Disneyland Star Tours ride and staring at the ocean.

Fig. 1 *The GK-1 installed on a guitar. Note the hex pickup between the treble and bridge.*

contemplate each note before playing it. Perhaps the most telling comment is that during some times while playing this setup, *I forgot I was playing a guitar synthesizer.*

SYSTEM DESCRIPTION
Roland's latest MIDI guitar system consists of three main elements.

The **GK-1 Synthesizer Driver** includes a hex pickup (improved over previous Roland offerings) that mounts between the guitar's treble pickup and the bridge, along with a small control unit (**Fig. 1**). The control unit (with MIDI volume control, MIDI balance control to mix straight and synth sounds, and two assignable MIDI controller knobs) slides into a mounting bracket that sits behind

the bridge, secured by the end strap post. A multi-conductor cable connects the control unit to the GM-70 (described below); another short cable plugs into your guitar's audio output jack to feed straight audio to the GM-70.

The **GM-70** (see **Fig. 2**) analyzes the signal coming from the GK-1 and outputs MIDI. It has a bunch of useful MIDI-related features, which we'll get into shortly.

And of course, you'll need a synthesizer (or more, if you really want to use the GM-70 to the fullest). A multi-timbral (mono mode) synth is preferred, so each string can have its own voice and react independently to articulation such as pitch bending. Roland recommends their

MKS-50 and MKS-70 as good choices; I did most of my initial testing on the MKS-50, an Oberheim Xpander, and a Kawai K3m (which doesn't do mono mode, but let me test out the poly mode response).

IN SEARCH OF THE RIGHT GUITAR

I had played the GM-70 at NAMM (National Association of Music Merchants trade show) and was very pleased by how well it responded and tracked. Since the GM-70 is compatible with Roland G-series hex guitars, I first tried using a Roland G-808 that I use with their older GR-300 synthesizer. The tracking was not as good as I had remembered. Since the NAMM unit used the GK-1 driver, not a standard G-series guitar, I thought that using the GK-1 might help.

The GK-1 hex pickup mounts with two small screws (forget about using double-backed tape or other makeshift measures in real world applications, although these can come in handy when searching for the proper pickup placement). Unfortunately, my Peavey T-60 and Tele-caster weren't compatible with the retrofit, and I wasn't about to stick two screws into my main axe, a Paul Reed Smith. A Les Paul or Strat would work, but I don't own either one. So, I tried adding the GK-1 to both the G-808 and to an old custom guitar. The results were good, but still not as good as what I had expected.

After much trial-and-error, a little guitar scarring, and a few cuss words, I finally found the secret: pickup positioning is *critical*, and unfortunately, the manual doesn't really describe what makes for a truly successful installation. First, the pickup must go as close to the bridge as possible. Once mounted, the pickup height must be adjusted (via the two mounting screws) as must the sensitivity (via six trimpots in the control unit or the body of a G-series guitar). Note that these two groups of settings interact.

The manual recommends a 0.5 mm gap between the strings and pickup, which might work for someone who picks softly; I hit the strings pretty hard, and encountered lots of glitching. What worked best for me was using the G-808,

GK-1 Pickup Installation

The following quick and dirty installation procedure does not have Roland's endorsement, but it worked for me and is the result of hours and hours of experimentation. At least give it a try.
1. Mount the pickup as close to the bridge as possible.
2. Set the pickup so that its top surface is located about 1 mm from the strings. Due to differences between bridge construction and the hex pickup, with all the guitars I tested the first and sixth strings were furthest away from the pickup, and the third string was the closest.
3. Adjust the sensitivity for the furthest-away string (either first or sixth; let's say sixth) to maximum, and adjust the pickup height at the sixth string for the proper indication on the GM-70 LED VU meter (i.e. it goes into the red on the very loudest peaks of your playing).
4. Adjust the pickup height so that it is as equidistant as possible from the other strings. The first string will probably be a little closer to the pickup than the sixth string, and the third string will be the closest.
5. Carefully set each sensitivity control for uniform response from the various strings. This setting is critical, as it not only affects "glitchability" but also the velocity response.
6. You are now set up for (hopefully) optimum tracking if you're a string-bashing kind of player. If you play softly, start moving the pickup closer to the strings, and readjust the sensitivity each time you make a pickup adjustment. Move the pickup in very small increments, and readjust the sensitivity each time (yeah, it's tedious). As you keep moving the pickup closer, at some point glitching will get worse even with proper adjustment of the sensitivity controls. At this point, move the pickup a little further away from the strings, adjust the sensitivity controls, and all should be well.
7. After you've played with the system for a while and are familiar with its personality, try the above steps once more and you'll be able to tweak things even closer to the ideal.

with its stock hex pickup, set for a distance of 1.25 mm at the first string sloping to about 2 mm at the sixth string. The sensitivity had to be cranked up to compensate for the extra distance from the strings. Using the GK-1 mounted on the G-808 seemed to track about 10% better than the stock G-808 pickup; still, I prefer using the G-808 as its controls are easier to manipulate than the ones on the GK-1 control unit. Expect to spend some time optimizing the pickup placement for your particular playing style—it took me about

four full evenings, working with four different guitars, to get this whole procedure figured out. The payoff, though, is that once you find the sweet spot, you don't have to find it again.

I mention all this because it would be a real shame if someone got a GK-1, installed it without fully knowing its requirements, and when it didn't work, figure that guitar synths still glitch like crazy. Proper pickup installation makes the difference between agony and ecstasy; the sidebar should help those of you

taking the plunge. (One other GK-1 glitch worth noting is that the pickup area under the G string was closer to its string than the areas under the other strings. I *very carefully* filed away a little bit of the plastic cover to solve this problem.)

Hopefully, someone will come out with a controller optimized for the GK-1. The ideal controller would have a somewhat deader response (a live, vibrating body promotes spurious responses and glitching), built-in controls (the knobs on the control unit are uncomfortably small
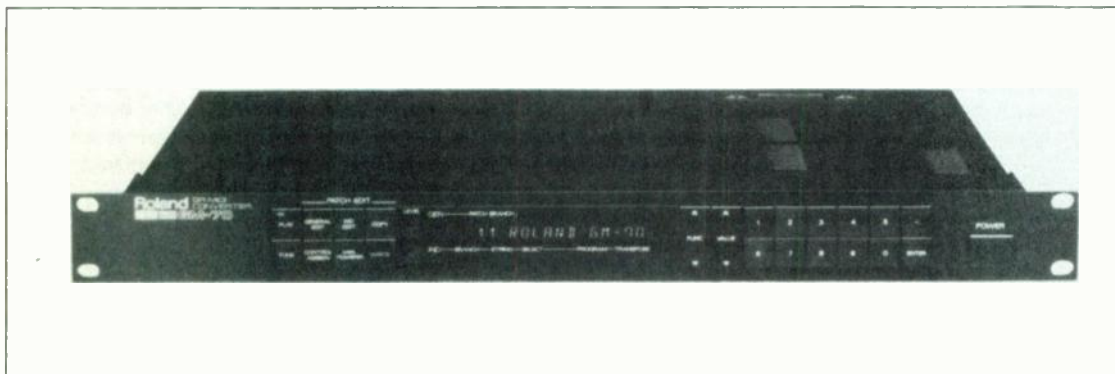
Fig. 2 *The GM-70 rack mount guitar-to-MIDI converter.*

and the whole unit, in fact, seems kind of fragile—presumably a necessary compromise to maintain reasonable size), and a removable felt mute on the bridge to damp string vibrations somewhat. I'd also appreciate a pitch bend wheel "whammy bar," a la Ibanez. Although the GK-1 can track a vibrato tailpiece, an electronic pitch bend mechanism is more accurate and does Real Live Transposing (just like a Steinberger TransTrem).

GM-70 BASICS

The GM-70 is a single-space rack mount device that does the actual MIDI conversion. Let's look at the rear panel jacks first to get an idea of how the GM-70 patches into a system.

MIDI In and Out: *Out* sends the MIDI-fied guitar data and bulk data dump info; *In* accepts program change and bulk data load info.

RRC In: This accepts program change data from the Roland FC-100 foot controller unit (which does program change selection and has two function-assignable pedals). A switch selects between RRC In and MIDI In, so you can't use both.

Switch/Pedal Inputs: There are two jacks for footswitches and one for a pedal. A "control assign" function assigns these to various MIDI control functions.

Guitar In: This accepts the Roland 24-pin GR-series guitar cord.

Guitar Out: This sends the straight guitar audio.

Stereo In and Out jacks: These are an unusual, but most welcome, feature. The Roland G-series guitars, and the GK-1 driver, have volume controls that are mapped to control MIDI volume (controller #7). The balance control also sends volume data over controller #7. However, not all synths respond to MIDI volume data; in this case, you can patch the synth *audio outputs* through the stereo in/out jacks, and these provide attenuation that responds to controller #7 info. If the "guitar out" jack is not being used, the straight guitar sound appears at these jacks, mixed in with the synth sounds.

Before getting all those cool synth sounds, you of course want to tune up. Standard pitch can be set from A=430.0 to A=450.0, and a tuner display shows whether each string is in tune with the reference. Proper tuning is important for best results.

The GM-70 supports poly or mono modes. Mono mode is preferred, but I was surprised how well poly worked with most synths.

There are 128 nameable presets, one for each memory slot of a synthesizer that implements all 128 MIDI programs. The presets memorize the following parameters: MIDI channel (in mono mode, this sets the lowest-numbered of the six channels); mono/poly mode; bend range; patch volume (this sets the overall level and, like the guitar's volume control, is sent over controller #7); velocity curve (one of five options); and the program number to be called up by the preset (thus, GM-70 preset 21 could call up, say, preset 107 on your synth).

Bend range is rather interesting. It can select chromatic (no bending; all notes quantized to the nearest half-step) or bend response, with lower bend numbers giving less stable operation but increased bend range. With a bend range of 1, for example, you can turn your normal half-step bend into an octave jump. Generally this is too unstable for anything except special effects; I found 7 to 12 to be a good bend range value.

One annoyance is that the presets are numbered in Roland's Bank/Program numbering system—16 banks of eight programs. Instead of simply counting from one to 128, the first digit gives the bank and the second digit gives the program number. For example, the count goes 11, 12, 13...18, 21, 22...28, 31, 32...38, etc. up to 88. Ah, but that's only 64 patches total. The remaining eight banks are indicated with a minus sign, so patch -88 is actually MIDI patch number 128. Got that? Mercifully, the program change numbers specified to call up programs on an external synth go from one to 128; I wish the presets themselves were numbered similarly.

BRANCHING OUT

The GM-70 has four "branches" that can drive up to four MIDI synths, each on its own channel (or channels, for mono mode). With only 16 MIDI channels, you can't drive all four synths in mono mode (since each synth would require six channels), but you can set up, say, two synths in mono mode and two synths in poly mode. (Note, however, that the GM-70 spits out *lots* of MIDI data when using all four branches—there is a definite delay by the time the data hits the last branch. The manual recommends placing the most percussive voices in the first branch.) Each

branch can be programmed for mode and channel, bend range, volume, velocity curve and program number.

Each string in a branch can also be programmed for string on/off, program number for a particular string (with a multi-timbral synth, each string can drive its own sound) and transpose—just the thing for dropping the sixth string down an octave and getting *monster* bass sounds with full chords.

Regarding controllers, the GK-1 has two knob controllers; the G-series guitars have three knob controllers and an assignable switch. The pedal that plugs into the GM-70 can also be assigned to any of several controllers, as can the two optional footswitches; the FC-100 foot controller includes two assignable pedals. Controller numbers one to 95 are supported, and for several of these, the display shows the actual function (mod, breath, foot, etc.)—a helpful touch reminiscent of the PCM70. Rotary controllers can also be assigned to "pressure" and "bend up."

All this gives a lot of latitude for fun 'n' games . . . set up the footswitches to do octave jumps or patch change, while the controllers vary balance, or panning, or modulation, or whatever. (Of course, the limits of your synth's MIDI implementation will determine just how much fun 'n' games you're going to have, which is one reason why Roland's MKS-50 and -70 are well-suited to the Roland MIDI guitar system). In the details category, switches can be set to latch (press on/press off) or unlatch (press for on, release for off), and if you're using a controller whose vibrato arm sends out controller data, you can choose from "absolute" (the rest point of the arm is value 0; raising or lowering the arm increases or decreases the controller value respectively) or "center" (the arm's rest point falls in the middle of the controller value range).

One request: I wish one of the switches could be assigned to choose between chromatic and a pre-programmed bend range. This would in essence be a rhythm/lead switch, since I generally use chromatic mode for rhythm and bending for leads.

But Is It Musical?
When I talked about the GM-70 with a friend of mine, he wanted to know if MIDI guitar is really worth it. Well, yes and no. MIDI guitar does not replace the beautiful sound of pick-on-steel through

a stack of amps strained to their limits. And frankly, prior to the GM-70 I found most guitar-to-MIDI stuff to be so glitchy it took all the fun out of playing anyway. But the GM-70 is good enough that, for the first time, I am able to *appreciate* what MIDI guitar can do. First, when sequencing, recording guitar parts gives a different feel from the same part played on a keyboard—there's something about a strum that adds a very human quality to a MIDI sequence. Second, it's inspirational to play a guitar and have *all those different sounds come out*. This is the kind of thing that gives a real creative boost when composing, and can be somewhat addictive. Third, between the four branches, the multi-timbral effects, the straight audio output, and the assignable controllers, you can get totally out of control. It's nice to know that in the digital age such things are still possible. About the only drawback was that I had to program a bunch of custom synth patches; what sounds good on a keyboard does not always sound good when played on a guitar.

Is the GM-70 fast? I can play mando-

lin-like pickings and still have it track. Based on my playing at NAMM shows the Photon guitar is a tad faster, but until I get to check one out in depth that's a qualified opinion. Although the GM-70 is not so fast when you're jumping from string to string, the limits seem to be how fast you can play *cleanly*—as long as what I played was clean, the GM-70 followed right along. Does it track? Quite well, and especially once you learn some of the playing techniques that work best with MIDI guitar. Most of my tracking problems come when I forget I'm playing a guitar synth and try to do mutes, harmonics, and other idiosyncratic playing styles.

I've always felt it was brave for Roland to keep pushing guitar synthesis after everyone else had given up, and now they really have something to show for their efforts. The GM-70 isn't perfect. But it's musical, and as close as you're going to get to perfection for the $1,000 price point. I like it a lot—in fact, I think I'm going to stop typing and go play with it some more. **EM**

Send albums, tapes, CDs or music videos to Robert Carlberg, PO Box 16211, Seattle, WA 98116. The opinions expressed herein are not necessarily shared by the management or any other intelligent lifeform. When ordering, tell them you saw it in Electronic Musician!

JEEM, *Neuro Pond* (cassette); *Star Gait* (cassette). It's entirely possible to come up with important, worthwhile, even surprising music without ever challenging the UREM (Unwritten Rules of Electronic Music). James Seeley proves this in his two tapes which take a familiar "4-track-tunes-on-analog-synthesizers" approach but do it with such imagination, variety, skill and enthusiasm that it all seems fresh again. He claims to also use a computer and MIDI software, but there isn't a trace of mechanical feel anywhere, indicating he's either very good at it or too modest to admit he played it all himself. *Neuro Pond* received *Soundboard*'s 1985 "Cassette of the Year" award; *Star Gait* is just as good. Both prove conclusively that the cutting edge isn't the only source of good music. $8.98 ($9.98 for *Star Gait*) postpaid from JimSyn Music, Suite 167, PO Box C-900, Scottsdale, AZ 85252.

The Muffins, *Open City* (Cuneiform Rune 5). Eight or nine years ago there was a thriving progressive rock scene in Maryland, populated by the Genesis/Yes-like Happy The Man; the Soft Machine-like Mars Everywhere; and the Henry Cow-like Muffins (I realize that makes *no* sense unless you're at least aware of the names...). Unfortunately, for them and for the world at large, the scene "died a horrible death" of economic strangulation according to Steven Feigenbaum, one of the founders and now a record exec with Cuneiform. As his good deed he's resurrected some unreleased tracks, demos and live cuts from one of the best, making nostalgia for the Golden Age all the more acute. Cuneiform Records, PO Box 6517, Wheaton, MD 20906.

Robert Carlberg is the national service manager for Audio Environments Inc., a nationwide supplier of original-artist music for restaurants and fashion stores. His hobbies are electronics and music, particularly electronic music.

Various, *Meadowlark Sampler 1986* (Meadowlark 7014). Meadowlark's second wave of releases includes this sampler, nicely paced to provide a coherent tour through one of the best catalogs in contemporary instrumentals. Premium vinyl, first-class recordings and tastefully understated covers help them permeate markets beyond their Christian music inception. Individual reviews follow.

Richard Souther, *InnerMission* (Meadowlark 7012). Souther's *Heirborne* was one of the highlights of 1986, so his second album for Meadowlark has been eagerly anticipated. Using a full complement of MIDI keyboards triggered by a Macintosh and Southworth software, *InnerMission* turns out to be noticeably more sedate than the manually-played ("dig-it"ally triggered) *Heirborne*. Still, it has some lovely and expressive synthesis, and does nothing to diminish Souther's status as a luminary of genre.

Amy Shreve, *Peace in the Puzzle* (Meadowlark 7011). The harp is a beautiful instrument, but a bitch to play. Shreve is an accomplished performer, playing with confidence and sureness throughout, except her compositions leave almost as much silence as sound. This makes *Peace in the Puzzle* an excellent background but not really "listening music."

Various, *Incarnation* (Meadowlark 7013). Interpretations of the Christmas season from the Meadowlark stable. Next time you're cringing from the well-meaning assassination of yet another tired carol, listen to these original offerings from true believers in the spirit. Actually, nothing here would prevent you from enjoying these new acoustic instrumentals year 'round.

Billy Smiley, *New Night* (Meadowlark 7009). Flugelhornist Smiley performs with a Fairlight CMI and a percussionist. If the instrumentation sounds like Mark Isham, well so does the music somewhat. I can imagine no higher praise.

John Michael Talbot, *Empty Canvas* (Meadowlark 7015). Sometimes peacefulness in mu-

sic is achieved through minimalism: no melody, no development, no variety of tone colors. A more satisfying peacefulness, however, is achieved when accomplished composers turn their craftsmanship to the task; Talbot falls squarely into the latter. Playing nylon-stringed acoustic guitar with the interweaving accompaniment of a cello, oboe, flute, recorder, French horn or harp, Talbot creates quite meditational music because he wants to, not because he has to.

Hadley Hockensmith, *Heartsounds* (Meadowlark 7010). Koinonia's guitarist and co-founder cops a few licks from Earl Klugh, Wes Montgomery and even Bob James to make an album of soothingly smooth guitar/synth jazz. Koinonia and the keyboardist from the Yellowjackets stand behind him.

Koinonia, *Frontline* (Sparrow 1116). Koinonia was Sparrow's (Meadowlark's parent corporation) first experience with the secular cross-over potential of instrumental music. Their earlier albums, *More Than a Feelin'* (1983) and *Celebration* (1984), were top notch examples of fusion jazz featuring Hockensmith, Weather Report percussionist Alex Acuna, and famous bassist Abraham Laboriel. The new album (1986) is a little less spontaneous sounding, but still ample evidence that jazz can be divine.

Daughter Judy, *Daughter Judy* (cassette). Entry-level girl-group pop music with cheesy Casio organ, bass, drums, guitar and the hard-workin' vocals of Julie Schmittinger. They have a lot of fun and don't take themselves too seriously, which is good. This was a big, big hit with the 13-year-old daughter of a friend of mine, if that's any indication. $4 from 322 Fourth Street Box 3, Jersey City, NJ 07302.

Dennis Andrew, *Reflections* (Daylight 02; cassette). New-agey minimal keyboard meditations. Ear balm for burned-out neurons. PO Box 284, Metuchen, NJ 08840.

McMike, *A Gift of Christmas Music* (McM 86121; cassette). Tacky lounge jazz versions of familiar Xmas carols done on DX7, CZ-101 and Juno-1 synthesizers. There's nothing wrong with the execution; it's the attitude that sends me screaming. Mike McRoberts, PO Box 26230, Colorado Springs, CO 80936.

McMike, *Music by McMike* (cassette). Mike McRoberts's own compositions are perfectly comfortable being just what they are, which is—for lack of a better term—tacky lounge jazz. Again, they're extremely well executed, gangland style. Same address as above.

Paul Simon, *Graceland* (Warner Bros. 25447-2). Much has been made of Simon's recording in South Africa with local musicians, some praising the exposure this gives the

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Oberheim, A Division of ECC Development Corporation   May/June 1987

The DPX-1 Manuever

Captain's log, Stardate 2341.57. While on routine patrol in sector Gamma Zeta 5 the U.S.S. Oberheim has encountered a strange satellite orbiting the planet Musicon IV.

"Report, Helm."

"Kiptin, the helm is not respondink to controls and our orbit is beginnink to decay. Phasers are inoperative – so are the rest of our signal processink gear. It must have somethink to do with that satellite. At this rate, we'll enter the atmosphere in less than 2 hours!"

"Science Officer?"

"The ship will enter the planet's atmosphere in 1.68796 hours, to be precise. The satellite seems to be transmitting an old Earth-style binary code, but the ship`s computer doesn't identify it as any known form of communication, Captain."

"The Federation`s last contact with the Musiconians 400 years ago reported the inhabitants were a very musical people. The records also show that the Non-Interference Directive was compromised and the Musiconians were inadvertently exposed to earth's MIDI technology. Communications Officer, is it possible that the transmission is some sort of musical code?"

"I've tried all hailing frequencies, sir, but it doesn't check out like any sequence data we've ever encountered before. Perhaps if we run it through the ship's Oberheim DPX-1, it might show up as sample data."

"Well, Science Officer, what do you think?"

"Yes, Captain, we had it modified with the latest software and hardware modifications just before we left port. The Oberheim DPX-1 can not only read disks created on the E-mu Emulator II, Sequential Prophet 2000 and Ensoniq Mirage, but it can send and receive samples in the form of MIDI Universal Data Dumps. It works much in the same way as our Universal Language Translator. The DPX-1 may just be the answer. The odds are..."

"Then try patching in the DPX-1. We don't have much time."

"Affirmative, Captain. I'm getting something."

"Put it on audio."

Wwwwrrraannngggoowweeejjjaaa allliirruuu... (Sound emanates from all directions.)

"That sound... it seems to be coming from everywhere!"

"Quite correct, Captain. The new DPX-1 options include eight individual outputs to take advantage of the sample-to-voice mapping from the original instrument. The new software even routes different voices from Mirage and Prophet 2000 disks to the eight outputs for individual amplification and processing."

"That's all well and good, but we've got to communicate with that satellite before it's too late. We've got to start coming up with sounds to transmit to it fast. Any ideas, Doctor?"

"Damn it, I'm a doctor, not a programmer! Doesn't he have any music in that green blood of yours or are those damned ears tone-deaf?!"

"I assure you there's no need to panic, Doctor. The DPX-1 was developed so that anybody can access a large library of professional quality samples. I've patched it into the ship's communications bay and begun transmission, Captain. But even with the improved sample loading time, I don't know if we can access enough sounds fast enough to..."

(Crew is thrown around the bridge as the ship's orbit begins to decay.)

"Bridge to engineering."

"'Aye, sir."

"Engineering, we need more speed."

"Cap'n, I don't think she can take any more!"

"Engineer, if we don't pull out of this decaying orbit, you're fired! Patch in the CD-ROM System from Optical Media International. Not only do samples load faster, there are thousands of sounds on each CD. There's got to be something there that will do the job. Bridge out."

"Kiptin!"

"What is it, Helm?"

"Kiptin, the DPX-1 and CD-ROM are workink! The satellite is movink away and the helm is beginnink to respond!"

"Communications, open a hailing frequency."

"Yes, sir. There's a message coming in now. Sir... the translation is... SEND MORE PATCHES!"

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Matrix-6 Patches From Key Clique

Besides the 1000 patches Oberheim has assembled for the Matrix-6, the programming experts at Key Clique are releasing their own great set of 100 sounds. Key Clique has built its reputation on offering thematic groups of professional-quality programs for the Yamaha DX7 - one of the most popular has been their Top Forty set. Their Matrix-6 set features 64 sounds modeled from that Top Forty set which are designed to emulate and compliment the original DX7 set. The other 36 patches concentrate on taking full advantage of the Matrix-6's sophisticated programming options and feature a wide range of analog sounds.

The Key Clique Matrix-6 Vol. 1 collection is available on cassette and on floppy disk in the SYS/EX format for Atari ST, Commadore 64, IBM PC and Apple II+/IIe (Mac users can get it in Opcode format). Each package actually contains three variations on the 100-patch set. While one variation is designed for Matrix-6 alone, the other two versions are programmed for use with a DX7/Matrix-6 combination and heavy emphasis is placed use of the DX7 pedal to control effects like modulation, filter changes, timbre variations and volume. The two versions compensate for the differences between older and newer DX7 pedals.

The Key Clique Matrix-6 collection is available from Oberheim or Key Clique dealers and retails for $50. To locate your nearest Key Clique dealer contact Key Clique at 3960 Laurel Canyon Blvd. Suite #374, Studio City, CA 91604 (818) 905-9136.

Recipe of the Month –
EPCH+BRZ (Electric Piano Plus Brass)

Thanks for joining us again at the Digital Deli. This month's recipe comes from the kitchens of our friends at Key Clique. The basic sound is an initial chime-like electric piano with a delayed brass effect coming in if the key is held more than about a second. Playing the keyboard with a lot of velocity results not only in a brighter sound but a shorter delay before the brass comes in. Lighter playing makes the sound softer and provides a longer brass delay.

Once again Matrix Modulation makes this patch irresistibly mouth-watering. Since the piano and brass parts of the sound require both different timbres and pitches for DCO2, ENV3 is used to change the waveshape from an initial mellow sawtooth to the bright saw needed for the brass. ENV3 simultaneously removes DCO2 from the MIX while sweeping DCO2 FREQ from the high tine pitch to a detuned level for the brass. ENV1 then forces DCO2 back into the MIX as the brass comes in, while the decay stage of ENV3 reduces the amount of DCO2 detuning. Keyboard velocity is also routed to ENV3 DELAY to control the time it takes for the brass to come in. Enjoy!

Patch EPCH+BRZ

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	0	1	2	3	4	5	6	7	8	9
00 DCO1	Freq 12	Fr/Lf1 -10	Sync 0	Pw 31	PW/Lf2 0	Wave 63	Wsel WAVE	Levers BEND	Keybd KYBD	Click OFF
10 DCO2	Freq 43	Fr/Lf1 +23	Detune +24	Pw 31	PW/Lf2 0	Wave 0	Wsel WAVE	Levers BEND	Keybd KYBD	Click OFF
20 VCF/VCA	Mix 50	Freq 40	Fr/E1 0	Fr/Prs 0	Res 10	Levers OFF	Keybd KYBD	VA1 63	VA/VI 0	VA/E2 +63
30 FM/TRCK	FM 7	FM/E3 0	FM/Prs 0	TrckIn KYBD	Track1 5	Track2 3	Track3 0	Track4 0	Track5 0	
40 RMP/PRT	R1 Spd 0	Trig STRIG	R2 Spd 46	Trig STRIG	Spd/Vl 0	Mode LIN	Legato OFF	Keymd RSGN		
50 ENV1	Delay 0	Attack 32	Decay 8	Sustn 0	Rel 01	Amp 63	Amp/Vl 0	Trig SRSET	Mode NORM	Lf1 Trig NORM
60 ENV2	Delay 0	Attack 0	Decay 44	Sustn 0	Rel 13	Amp 5	Amp/Vl +63	Trggr SRSET	Mode NORM	Lf1 Trig NORM
70 ENV3	Delay 45	Attack 12	Decay 12	Sustn 62	Rel 0	Amp 63	Amp/Vl 0	Trig SRSET	Mode NORM	Lf1 Trig NORM
80 LFO1	Speed 59	Sp/Prs 0	Wave TRI	Retrig 30	Amp 0	Ap/R1 +63	Trig STRIG	Lag ON	Smpl KYBD	
90 LFO2	Speed 38	Sp/Kbd +22	Wave TRI	Retrig 0	Amp 55	Ap/R2 0	Trig OFF	Lag ON	Smpl KYBD	

Matrix Modulation

	Source	Amount	Destination
0	ENV3	-63	DCO2F
1	EVN3	-63	DCO2F
2	VEL	-33	E3DEL
3	PED2	+63	E3REL
4	PED2	+54	E2REL
5	PED2	+64	E1REL
6	ENV3	-52	MIX
7	ENV3	-63	DCO1W
8	ENV2	+55	VCFFQ
9	ENV1	+45	MIX

Performance Notes

Play this voice staccato with the sustain pedal down to get a chimey electric piano.

Keep your fingers on the keys for about a second and a brass sound is added.

Velocity controls the speed of the brass entrance as well as timbre.

OberView

WRH

OberView.

Billy Ocean Makes Waves with **Oberheim.**

In a few short years Billy Ocean has skyrocketed to fame after the release of his "Suddenly" LP, now double-platinum. But years of dues-paying preceded his success, emigrating from his native Trinidad at an early age to London's east side where he worked as a tailor. Seeing talent in the making, his boss even gave him 30 pounds to buy a piano. And talent it is. The tides changed with a sea of Top 10 hits include "Loverboy", "Caribbean Queen", "There'll Be Sad Songs", "When The Going Gets Tough" and "Love Zone". He's also won a 1985 Grammy Award for Best Male R&B Vocal Performance and 1986 American Music Awards for Top Single and Top Male Vocalist Video.

Many of Billy's hits have had that tell-tale Oberheim sound on the tracks. "I had always heard good things about Oberheim. Then I got to hear the gear first-hand because my keyboard player, Godfrey Wang, uses a Matrix-12. Since then, Oberheim has played a great role on my records and in concert."

Since Billy likes to write his own material, it wasn't long before he jumped on-board with Oberheim himself. "I'm a singer and a songwriter, not a programmer. I like to use the Matrix-6 when I write because I've got Oberheim's library of 1000 patches at my fingertips. New sounds mean new inspiration - and inspiration is the name of the game when it comes to writing songs!" No problem. With Oberheim at his side, Billy's cruising toward another tsunami of hits.

Turbo-Charge Your DPX-1 With The CD-ROM System From Optical Media

If you've read "The DPX-1 Maneuver" on Page 1, you know that the new DPX-1 improvements allow it to be interfaced directly to the CD-ROM System from Optical Media. Their CDS3 player and 'Universe of Sounds' CD-ROM discs give you thousands of digital sounds to choose from. No more searching through piles of floppy disks — you merely select the number of the sound you want on the hand-held remote and voila — you've got your sound!

The OMI CD-ROM System consists of the CDS3 CD-ROM Drive, a hand-held remote control selection unit, and your choice of two 'Universe of Sounds' CD-ROM discs. Each 'Universe of Sounds' CD-ROM disc contains over 4,000 sounds — individually sampled, tuned, looped and blended with Sound Designer by DigiDesign. The sounds include strings, synthesizers, orchestral samples, guitars, percussion, and sound effects.

In May OMI released the 'Universe of Sounds' Volume II containing samples gathered from specialists around the world — including an orchestra in Yugoslavia. Both Volume I and II are fully compatible with the DPX-1 and together represent a library of over 8,000 sounds! And OMI will publish future discs specializing in strings, sound effects and percussion.

The OMI CD-ROM with one 'Universe of Sounds' CD has a suggested retail price of $2,995, and Volume II disc sells for $995. This system requires the installation of a interface port in the DPX-1 — available at Authorized Oberheim Service Centers. To learn more about the CDS3 System, contact Optical Media International at 485 Alberto Way, Los Gatos, CA 95030 (408) 395-4332.

OberView.

DPX-1 Manuever

"That shouldn't be a problem, should it, Science Officer?"

"Correct, Captain. The Oberheim DPX-1 still has the largest library of sampled sounds available anywhere in the galaxy, without having to waste time sampling. And now with the CD-ROM System, there's no limit to the sounds available. The other new modifications are all very logical. Now the DPX-1 can make back-up copies of samples onto 3-1/2" floppy disks in the special Oberheim format..."

"Science Officer..."

"...which saves the entire DPX-1 configuration including MIDI channel, fine-tune, controller settings, patch number and extended functions. The Oberheim-format disks load faster because there's no conversion time, it's more economical and there are no head alignment problems because the disks are made on the DPX-1. The new software..."

"Science Officer..."

"...also backs up sounds from MIDI Data Dumps. It even formats the disks so you don't have to buy proprietary pre-formatted

disks. There's also error detection now when loading samples..."

"SCIENCE OFFICER!"

"Yes, Captain?"

"How do you know so much about the DPX-1? Isn't music a rather... human indulgence?"

"Actually, the people of my planet began developing musical instruments 427.849 centuries ago when..."

"Kiptin, the Russians first inwented MIDI in Leningrad over 5000 years before the Cossacks..."

"Gentlemen, gentlemen.... please! Helm, plot a course for the nearest Starbase with an Authorized Oberheim Dealer. I have a feeling they're going to need all the help they can get when the galaxy finds out about these new DPX-1 mods. Ahead Warp Factor 2."

Captain's Log, Stardate 2342.68. On reaching Starbase 12, we have been informed by Star Fleet Command that all new Oberheim DPX-1's support the Oberheim-format back-up disks with improved load time. DPX-1 owner's can also get the optional CD-ROM and eight individual voice output update kits installed at their nearest Authorized

Oberheim Service Center. Thanks to the DPX-1, the galaxy is a safe place to make music again! Captain out...
By Jeff Burger

Matrix-6
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The Matrix-6 is now being shipped with a library of 1,000 sound. With cassette mode, you can selectively load and hear all of these sounds without the need of a computer or a librarian program. These 1,000 sounds can also be purchased on cassette from Oberheim dealers or direct from Oberheim. The suggested retail price is $19.95. This is the largest patch library available from any manufacturer, ever. **Audition the great sounds of the Matrix-6 and Matrix-6R at your Oberheim dealer today.**

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EM

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burgeoning scene there, others decrying the apparent tacit acceptance of apartheid. Quite apart from the question of entangling politics and music, it seems likely that Johannesburg could become the next Kingston, attracting worldwide respect for the caliber of musicians there. *Graceland*, as a first mass export, is more Paul Simon than indigenous, but still gives indications of the potential. As a Paul Simon album, he's written stronger material before and he's previously been more daring. He joins Peter Gabriel and Steve Winwood in bending toward the mainstream.

John Scofield, *Electric Outlet* (Gramavision 8405); ***Still Warm*** (Gramavision 8508). Scofield is one of the busiest guitarists working today, performing prominently with John Abercrombie, Miles Davis, and others. These solo albums are straight-ahead electric jazz, giving everyone (synth, bass and drums with sax and trombone on *Outlet*) a chance to stretch out. His compositions are tasteful and pleasant without a hint of anything the least bit off-the-wall.

Michael William Gilbert, *The Light in the Clouds* (Gibex 003; CD). Michael has been a stalwart of independent electronic music since 1978. His albums (*Moving Pictures, The Call, In the Dreamtime*) all combine acoustic percussion, wooden flute, standup bass and synthesizers in a unique electroacoustic world-fusion music. *The Light in the Clouds* is debatably the least tuneful of the four, using the medium of digital recording to examine close-up the purity of sound, both real and sampled. Zen Ding an sich par excellence. 73 Spaulding Street, Amherst, MA 01002.

Craig Philips, *Moving by Numbers* (CSA 10185; EP). Six songs (about 22 minutes) of Top-40 style pop; written, sung, produced, recorded and programmed by Craig Philips Soukup. He used a MIDI-based, 16-track studio and what sounds like digital drums, although six guests (including two drummers) are credited. Craig's not a great singer and his songs are orthodox, but they're well-presented. $6 from PO Box 489, Mundelein, IL 60060.

Dick Bauerle Group, *Dick Bauerle Group* (MCA 5894). A while ago I got on MCA's promotional list for "fusion jazz," and for the life of me I can't imagine where they find it all. The DBG actually is one of the more tasteful apples in the bin, playing exuberant guitar jazz which entertains without really saying anything new. Then again, maybe on a debut album it's enough just to prove you can.

Alien Planetscapes, *Survival in the Nuclear Age* (cassette). Doug Walker believes in extreme music. Using synthesizers, organs, electronic flute, radio, tapes and a lot of distortion, he creates live, in the studio (no overdubs), an industrial noise-music which sounds for all the world like Kluster circa 1970 (See May '86

column). Loyal fans of the fringe will want Walker in their bombshelter. $5 from 479 Fifth Avenue, Brooklyn, NY 11215.

Slap, *Pratique* (Duotone 5); ***Songs from the Cross*** (Duotone 7). Slap is Stephen Nester on Yamaha synthesizers and sequencer, which he sets up with dense tribal rhythms using ring-modulated metallic percussion voices. There has been discussion in print about how DX's are only good for certain types of "pretty music." Nester puts the lie to this notion with his challenging and unique vision. *Songs from the Cross* in fact involves some non-rhythmic chording which resembles Stewart Copeland's Fairlighting on *The Rhythmatist*; not a bad connection considering the price. Duotone Records, PO Box 1168, Miami, FL 33243.
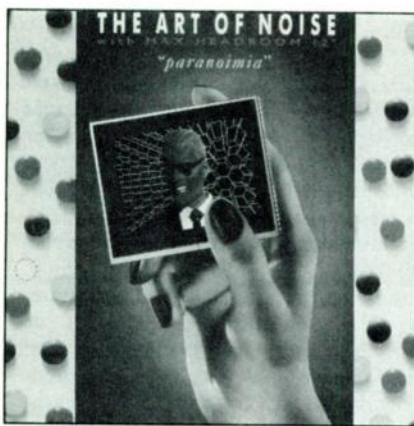
Art of Noise with Max Headroom, *Paranoimia* (Chrysalis 43017; 12-inch EP); ***Legacy*** (Chrysalis 43086; 12-inch EP). It makes perfect sense that Max Headroom, that electronically generated "TeeVee Personality," should show up on an Art of Noise EP, scion of sampling-gone-awry. After all, they're both creations of technology in the computer age. Each AON album generates a host of aftershocks, loosely based on the major disturbance but with new samples plugged into the LinnDrum formula. This seismic activity is necessary to relieve the tension in a changing electronic topography, and like a pressure vent it probably prevents anyone else from getting this extreme.

Walter Holland, *Relativity* (Coriolis 0103); **Amber Route, *Ghost Tracks*** (Coriolis 0101). Synthesist/guitarist Walter Holland, and his group Amber Route, use digital and analog synthesizers with "control computers" (sequencer) to create jazz-rock with a distinctly electronic edge to it. He seems more interested in exploring large abstract sounds than the melodies he touches on, though his music is obviously grounded in European harmony. *Relativity*, dedicated to "Albert Einstein and all people who are not afraid to think non-traditionally," features a real live drummer (!) who keeps the momentum aggressive. *Ghost Tracks* adds clarinet/EVI-man Richard Watson and bassist Lou Rossi, and on four of the six tunes

Zappaesque growling vocals. Both share excellent production, deep bass and great covers. $8 each from Coriolis Records, Box 3528, Orange, CA 92665.

Mazzei Family, *The Sound Castle* (cassette). Anthems and short themes written for a horse auction (side 1) and Arabian horse video (side 2). The Mazzei family—mother, three daughters and one son—play Kurzweil synthesizer and Linn 9000 drumbox, or alternatively, sing very professional heavenly wordless chorales. It's hard to come up with anything new these days, but the Mazzeis are unique. 1142 Nevada Street, Pittsburgh, PA 15218.

Raymond Daniel Platt, *Fields of View* (Paradise Boutique 303; cassette). By combining sax, tabla, clay drums and hand-held percussion with Linn 9000, Oberheim Matrix-12 and Yamaha TX7, Ray Platt (and guests) create a nice fusion of electronics with the acoustic. It's a bit more "jazz" than Michael Gilbert or Jon Hassell, but not so much that it loses its exotic feel. 19100 Lassen Street, Northridge, CA 91324.

John Wiggins, *All The Truth At Once* (cassette). Sound-collage artist Wiggins is at it again, tossing together bleeps, blips, bongs, and boings, and blams in a noise salad of incredible variety—but no overall plan. Simply identifying (or speculating on) the noise sources could provide hours of family entertainment. Should be out in LP from RRRecords, 151 Paige Street, Lowell, MA 01852.

Dave Williams, *Flutejazz* (cassette). Dave Williams plays processed flute—altered, extended and electronified by foot-pedal controlled digital delays—manipulated in real time without overdubs, to the accompaniment of a standup bass. The technique is amazing, though the music (all barely recognizable standards) is less-so. 1055 Massachusetts Avenue, Arlington, MA 02174.

Paul Dolden, *Sonarchy* (Underwhich 26; cassette). Acoustic instruments (piano, woodwinds, strings and brass) get overdubbed— up to 280 times—until they sound like a vast unfiltered digital screech. The depth and texture varies constantly, making the traverse diverse if not particularly picturesque.

Michael Chocholak, *Owl Man Dreams* (Underwhich 29; cassette). Slow digital drone chords and fast repeated piano or analog synth, combined in compositions which are not strictly speaking "musical." Non-rhythmic percussion and open unlabored structures promote an enigmatic overall effect.

Mara Zibens, *Trance Resistance* (Underwhich 24; cassette). Drone-y "trance music" on analog synthesizers with occasional sequencing, not unlike early Dream or Froese

solo. All three Underwhich cassettes are available from PO Box 262, Adelaide Street Station, Toronto ONT M5C 2J4, Canada.

Patrick O'Hearn, *Between Two Worlds* (Private Music). Pat's first solo album, *Ancient Dreams*, was my choice for 1986's "Album of the Year" (in a column which, due to space restrictions, never appeared). Will his second be 87's A.O.Y.? Frankly, no—the music's a little too laid back, too mechanical this time around. But it still has some real nice Fairlight work in a digital recording of remarkable depth. I wouldn't write it off altogether.

Mute Beat, *Japanese Dub* (ROIR A-143; cassette). Japanese Reggae? Can you imagine them in dreadlocks? Leader Kazufumi Kodama's trumpet work at times sounds more like Herb Albert than Herb Smoker—especially when mashing up Paul Desmond's "Take Five" (miscredited to Dave Brubeck)—but the Japanese are known for perseverance.

Polyrock, *No Love Lost* (ROIR A-144; cassette). This six-piece guitar rock band kicked around New York's club scene from 1979 to 1982, eventually putting out three uneventful albums before folding quietly. This tape of demos and live tracks helps explain their climb to obscurity. Both ROIR cassettes are available from ReachOut International Records, 611 Broadway, Suite 725, New York, NY 10012.

Billy Barber, *Lighthouse* (DMP 455; CD only). Keyboardist Barber (one of the "BBs" of "Flim &") plays Steinway grand piano plus Kurzweil, Prophet and Yamaha synthesizers slaved to a Mac and IBM-PC. His compositions tend to be very pianistic, even when the K-250 is soloing (which is often). Almost nothing is improvised, indicating the programs were all written (except for the piano) long before they were dumped direct to digital. As a result, the music is somewhat stiff, but oh so polished.

Flim & The BBs, *Big Notes* (DMP 454; CD only). Actually, Dick Oatts (saxes) has joined the trio, making the name a little inaccurate. It should be "Flim & D.O. & the BBs," but that doesn't have much of a ring to it, does it? This jazz quartet, back when they were a trio, became sort of a house band for 3M during their experiments with digital in the early '70s. They're not the most exciting band in the world, and they don't do much to test the limits of digital this time (unlike their first ultra-dynamic CD), but they do put together a journeyman studio jazz set. Barber's Kurzweil is spotlighted.

Stefan Nilsson, *Romantic Piano Dreams* (Breakthru ABCD5; CD or cassette). Digitally recorded piano and strings (of the Swedish Royal Opera House Orchestra) plus unspecified (but probably Wave) synthesizer. The strings are miked close-up, not like back-up but as equal partners, and Nilsson's jazz-tinted writing recalls George Gershwin or John Lewis. The synthesizer, when it appears, is used for contrasting color—sometimes very contrasting—but without altering the overall "romantic" direction.

David Panfili, *Other Worlds* (Paragenes CD8571; CD only). This compact disc is subtitled *Music For Guitar and Synthesizers*—sort of a William Ackerman meets Tony Banks affair. Gene Rabbai's synthesis (possibly a DX or large Casio) is fitted into Panfili's steel-string acoustic guitar compositions without a whole lot of interaction. A pleasant, though unpretentious set. $12 postpaid ($4 more for a deluxe cover and CD Mate giftpack) from 1134 Burbank Blvd., North Hollywood, CA 91601.

Katzer/Krause/Krcek, *Aide Memoire/Folk Music/Sonaty Slavickove* (Recommended Records 22; EP). Recommended Records regularly unearths and re-releases neglected masterpieces from around the world, including the past Faust and Slapp Happy. This 30-minute 45-rpm compilation presents experimental music from behind the Iron Curtain—music which is definitely *not* state-sponsored (see "Electronic Music Behind the Iron Curtain" EM, March 1987). Georg Katzer, from East Germany, wrote Aide Memoire to commemorate the 50th anniversary of Hitler's rise to power in 1933. It is a dense cacophonous collage of Hitler's speeches, military music, crowd noises, popular music of the day, and hundreds of other sounds taken from period recordings. Even without speaking German, one gets powerful impressions of the trauma still lingering a half-century later. Zygmunt Krause's *Folk Music*, from Poland, divides a symphony orchestra into 21 sections, each playing a different folk melody from start to finish. The result is oddly pastoral, as if the sum of the parts somehow remembers the origin of each part. Czechoslovakian Jaroslav Krcek sounds like he's using sampling in *Sonaty Slavickove*, but in 1970 there was only tape-music. Elaborate processing gives the sounds a digital complexity, as they wash in and out of a short five-minute "term paper" written on completing a course in Musique Concret by Miloslav Kabelac. The shortest piece on the disc is also the most interesting.

Bill Rhodes, *Outside Looking In* (cassette); ***Rupprechts Werke*** (Jazzical 86003); ***Mind Break*** (Jazzical 85023). Keyboardist, columnist (*Electronic Musician* and *Polyphony*) and author (*Applied Synthesis*), Bill Rhodes has a number of releases out now of his DX7, Kurzweil, Juno-80, Memorymoog, Prophet and piano neo-classical compositions. *Outside Looking In* and *Mind Break* are both fusion-jazz efforts, and *Rupprechts Werke* is an anthology of Bill's work, 1978 through 1985. Although his style is a bit flashy for my tastes, he certainly knows his chops. 1 Wyndmere Road, Piscataway, NJ 08854.

Fairport Convention, *Expletive Delighted!* (Varrick/Rounder 029). This is the 15th official (umpteenth unofficial) album by the Convention, who in their 20-year history have themselves become a tradition in traditional music. It merits a review in *Electronic Musician* because: 1) it is all instrumental this time and, 2) it utilizes a number of electronic devices (delays, processors and synthesizers) in a blend of acoustic jigs and reels with contemporary electric jazz-rock. Fairport remains anything but conventional—as is traditional for them. **EM**

notes can be set globally (for the machine) or locally (for one patch). There are 20 factory and 20 user patches and more on cartridge (40 in RAM, 80 in ROM). Six different sounds are available, each voice an independent synthesizer. Pitch is set by front panel controls or a received MIDI note number. The voice split mode allocates the six voices to two sections, so you can, for example, reserve two voices for a bass line and the other four voices for a vibes part. As in the TMI and MTM, up to 99 patches can be strung together to form a sequence accessible via a footpedal.

Voice programming on the SDE is not difficult but I did have some trouble getting the sounds I wanted. Seven "Digital Generators," each with six "Multi Parameter Controls" are used to synthesize sounds. The descriptions of presets can sometimes be ambiguous, but naming sounds is a notably inaccurate undertaking. Naming and knob twiddling aside, the SDE is capable of making a wide variety of interesting and useful sounds. My one complaint is that some of the sounds contain excess "digital grunge."

Like the TMI and MTM, the SDE suffers from a rather confusing owner's manual. At times you're not sure if they're talking about a MIDI channel, a trigger channel, or a voice channel. I know that there's more than one word available to describe these functions! (How about a thesaurus, guys?)

But any minor complaints notwithstanding, I think Simmons has done well with their venture into the MIDI realm, and I'm sure that many musicians will agree. **EM**

AC coupling: An interconnection technique between stages in an electronic circuit that blocks direct current (i.e. a constant current flow) but allows alternating current (AC, such as an audio signal) to pass. A series capacitor, which blocks DC but not AC, is ideal for interconnecting stages in this manner and essentially acts like a high pass filter.

ADSR: Abbreviation for Attack, Decay, Sustain, Release, the four adjustable parameters found on the most common type of analog synthesizer envelope generator.

Aftertouch: Modulation added by exerting pressure on a MIDI keyboard key after it has already been pressed down (i.e. at the bottom of the key travel). This modulation typically modifies the filter cutoff, degree of LFO modulation, level, etc. Also called pressure sensitivity.

Attack: The initial portion of a sound. Example: a plucked or struck sound has a fast, percussive attack; a wind instrument has a slower attack since it takes a finite amount of time to pump enough air into the instrument to play a note. When applied to an envelope generator, the attack is usually the first phase produced by the envelope generator. It determines the amount of time required for the envelope generator signal to go from full off to full on.

Chorus: A signal processing technique that makes one instrument sound like several instruments playing in ensemble.

Controller: (1) A device that operates synthesizer electronics (i.e. keyboard, guitar with appropriate interface, etc.). (2) A device that operates some specific parameter of a synthesizer (i.e. modulation wheel, pitch bender, ribbon controller, etc.). (3) MIDI data that describes the synthesizer's physical controller movements. (Also see MIDI Volume, below.)

Derate the current: Many manufacturers' specifications list a maximum usable current that pushes the limits of the units' capabilities. It is safer to "derate" the current by a certain amount to determine a practical usable current. Therefore, if you want to derate the specifications' rated current by 20%, you would multiply the rated current by 0.8 (100% -20%=80%)

to determine the usable current. I(usable) = 0.x × I(rated) where x is percentage expressed in hundredths.

DCA: Digitally Controlled Amplifier. A device that controls the amplitude of a signal according to digital data.

DCO: Digitally Controlled Oscillator. A device that generates a signal whose frequency is determined by digital data. This is the main type of tone generating element found in many low-cost, computer-based synthesizers.

Decay: The second phase of a standard ADSR Envelope Generator. This determines the time a signal takes to fall from the maximum voltage attained during the attack phase to an arbitrarily set sustain (see below) voltage.

Envelope: The boundary or contour of a signal over time. Example: devices called envelope generators create specifically shaped envelopes used to control a VCA's amplitude, VCF's cutoff frequency, VCO's pitch, etc.

Envelope Generator: A synthesizer circuit that generates a control voltage of a predictable nature, typically used for controlling the amplitude of a VCA, cutoff frequency of a VCF, or pitch of a VCO.

EQ: Slang for equalization.

Equalization: The process of altering the distribution of energy in various frequency bands of a signal.

Filter: A device that removes part of a signal or data stream. Example: an audio filter removes part of the energy present in a particular frequency band; a data filter removes a particular type of data from a data stream.

High frequency shelving: see Highpass Filter.

Highpass Filter: A circuit that passes frequencies above a certain point; the lower the frequency below that point, the greater the degree of attenuation.

International Parts Specification Standard: A system of specifying electronic parts' values used throughout the world and in EM. This standard avoids the un-

necessary use of zeroes, decimal points, and stating Ohms (Ω) or Farads (F) when implicitly understood. For example:

USA	Int'l
1.5 kΩ	1k5
2.2 MΩ	2M2
10μF	10μ
0.01μF	10n
3300pf	3n3
0.00228μF	2n2
10pF	10p

where:

k	=	kilo	=	10^3 Ohms
M	=	Mega	=	10^6 Ohms
μ	=	micro	=	10^{-6} Farads
n	=	nano	=	10^{-9} Farads
p	=	pico	=	10^{-12} Farads

Therefore, 100 nF is equivalent to 0.1 μF; and 20 nF equals 0.02 μF. Be *sure* when ordering parts based on construction articles in this magazine that you understand the difference between, say, a 20 nF cap and a 20 μF capacitor. Don't expect a parts clerk to know what a nanofarad is.

LFO: Abbreviation for Low Frequency Oscillator, a synthesizer module or sub-circuit that generates a periodic, generally sub-audio control voltage used to control some other synthesizer parameter. Example: applying an LFO signal of around 7 Hz to vary the pitch of an oscillator produces vibrato effects. Applying the same signal to control the gain of a VCA produces tremolo effects.

Low frequency shelving: see Lowpass Filter.

Lowpass Filter: A circuit that passes frequencies below a certain point; the higher the frequency above that point, the greater the degree of attenuation.

MIDI Mapping: The process of altering MIDI data specific to one device to make it compatible with another MIDI device. Example: suppose triggering a drum machine snare sound sends note-on data as MIDI note number 41, and that you record this data in a sequencer. Now suppose you play back that sequencer track through a drum machine with mappable drum sounds. With this mappable drum machine, you could "map" MIDI note number 41 to any of the available drum

sounds. Mapping is also used with program change commands. Example: calling up program number 2 on one synthesizer might be mapped by another synth to call up a different program number.

MIDI Volume (controller #7): MIDI can send data that affects particular synthesizer parameters—modulation, portamento on/off, aftertouch, and so on. In pre-MIDI days, these were normally controlled by physical synthesizer controllers such as pedals, levers, wheels, knobs, switches, etc. Nowadays, these parameters can be controlled via physical devices (as before), or by MIDI data that mimics the effect of physically turning a control. MIDI makes provisions for several *continuous controllers* (which cover a range of values, like a potentiometer) and *switch controllers* (which cover one of two possible values, on and off). Each controller is identified by a unique number so that data intended for one parameter affects that parameter only. In the case of MIDI volume, overall level messages are identified as controller number #7. Thus, if data tagged as controller #7 is increasing in value and feeds a synthesizer, the synth level will increase. Conversely, if the data is decreasing in value, the synth level will decrease. Note that not all synths implement this function; check the MIDI implementation sheet to check whether this function is available on a given machine.

Mono mode: This MIDI mode assigns each voice (sound-generating element) of a synthesizer, drum machine, etc. to a single MIDI channel. Thus, each voice can play an independent part according to the data being sent to it over its own MIDI channel.

Multi-timbral: A multiple-voice device that can operate in MIDI Mono mode, and offers the additional advantage of being able to assign an individual timbre (as well as MIDI channel number) to each voice.

Multi-timbral synthesizer: An instrument that can provide a unique timbre (bass, violin, trumpet, etc.) for each voice in the synthesizer, as well as assign each voice its own MIDI channel. Thus, multi-timbral instruments can play independent melody lines, each with a different timbre, when driven by a multi-track MIDI sequencer.

9 Volt battery eliminator: A device that converts the AC voltage coming out of wall sockets to 9 Volts DC. This DC voltage may be used in electronic circuits in place of a 9 Volt battery; this is desirable because operating a device from AC power costs less than buying batteries to provide an equivalent amount of power. Using an AC adapter is generally better for the environment as well.

Omni mode: A MIDI mode in which a MIDI receiver will react to, and attempt to play, MIDI data appearing on any of the 16 MIDI channels.

Power supply rejection ratio: The specification that defines the ability of an electronic circuit to reject spurious signals carried on the power supply lines (hum, noise, voltage spikes, etc.). If a circuit has a poor power supply rejection ratio, then these spurious signals could show up in the output as part of the total signal.

Pro net price: A retail price, set by the manufacturer, that in theory is not subject to discounting as are many list prices.

PCM adapter: Pulse Code Modulation adapter. A unit that converts audio signals into digital data in a form that is stored on the video tracks of a video tape recorder. This provides high quality digital recording with the aid of a common VCR.

Release: The portion of an ADSR envelope that, upon releasing a key, goes from the sustain value back to zero and thus terminates the envelope generator cycle.

Stereo plug: A connector that carries two audio signals, typically left and right, and ground.
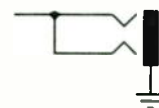
SCHEMATIC:
STEREO JACK

HOT #1
HOT #2
GND
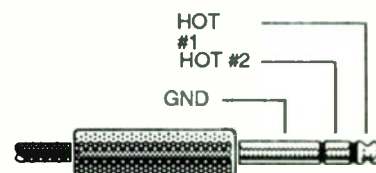
STEREO 1/4" PHONE PLUG

Sustain: In the context of an envelope generator, sustain is the level at which an envelope will remain during the time that a keyboard key is held down. Releasing one's finger from the keyboard key initiates the release cycle (see above).

Tolerance rating for resistors and capacitors: A manufacturing run of electronic components will not always produce components with totally consistent values. As a result, manufactured parts are tested and assigned a particular tolerance, i.e., how close they are to a standard value. For example, the actual value of a 10k resistor with a 10% tolerance rating could be anywhere between 9k and 11k. A 10k resistor with a 5% tolerance rating could be anywhere between 9.5k and 11.5k. In most applications, 5% precision is adequate.

VCA: Abbreviation for Voltage Controlled Amplifier. An element that varies gain in response to the voltage applied to a control voltage input.

VCF: Voltage Controlled Filter. An element that varies a filter parameter (usually cut-off frequency) in response to the voltage applied to a control voltage input.

VCO: Voltage Controlled Oscillator. A circuit whose frequency varies in response to the voltage applied to a control voltage input.

WVDC: Abbreviation for Working Volts DC, the maximum operating voltage of a capacitor. For example, a capacitor with a 15 WVDC rating can accept a maximum potential difference of 15 Volts or less across its two terminals.

XLR connector: A multi-pin connector used primarily in pro audio work.
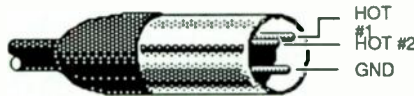
XLR PLUG

353

Wide Bandwidth
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OUTPUT A	1	8	V+
INVERTING INPUT A	2	7	OUTPUT B
NON-INVERTING INPUT A	3	6	INVERTING INPUT B
V-	4	5	NON-INVERTING INPUT B

4136

Quad Op Amp

- IN A	1	14	- IN D
+ IN A	2	13	+ IN D
OUT A	3	12	OUT D
OUT B	4	11	+15 V
+ IN B	5	10	OUT C
- IN B	6	9	+ IN C
- 15 V	7	8	- IN C
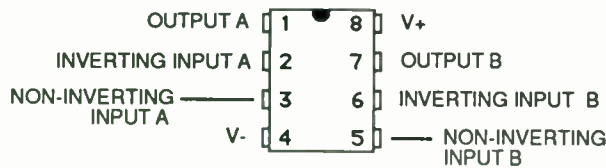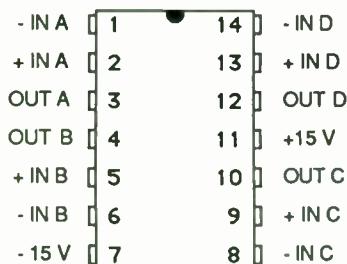
LM 386

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+ INPUT	3	6	+ SUPPLY
GROUND	4	5	OUTPUT

EM

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OR CONSEQUENCES.

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CONSEQUENCES: Bad mixes. Re-mixes. Having to "trash" an entire session. Or worst of all, no mixes because clients simply don't come back.

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CONSEQUENCES: When pushed hard, most tweeters simply fail. Transient detail blurs, and the material itself deforms and breaks down. Other materials can't take the stress, and crack under pressure.

TRUTH: The Frequency Dividing Network in each 4400 Series monitor allows optimum transitions between drivers in both amplitude and phase. The precisely calibrated reference controls let you adjust for personal preferences, room variations, and specific equalization.

CONSEQUENCES: When the interaction between drivers is not carefully orchestrated, the results can be edgy, indistinctive, or simply "false" sound.

TRUTH: All 4400 Studio Monitors feature JBL's exclusive Symmetrical Field Geometry magnetic structure, which dramatically reduces second harmonic distortion, and is key in producing the 4400's deep, powerful, clean bass.

CONSEQUENCES: Conventional magnetic structures utilize non-symmetrical magnetic fields, which add significantly to distortion due to a nonlinear pull on the voice coil.

TRUTH: 4400 Series monitors also feature special low diffraction grill frame designs, which reduce time delay distortion. Extra-large voice coils and ultra-rigid cast frames result in both mechanical and thermal stability under heavy professional use.

CONSEQUENCES: For reasons of economics, monitors will often use stamped rather than cast frames, resulting in both mechanical distortion and power compression.

TRUTH: The JBL 4400 Studio Monitor Series captures the full dynamic range, extended high frequency, and precise character of your sound as no other monitors in the business. Experience the 4400 Series Studio Monitors at your JBL dealer's today.

CONSEQUENCES: You'll never know the "truth" until you do.