MARCH 1986

- MUSIC
- MICROS
- MIDI
- MULTI-TRACK *and more!*

Electronic MUSICIAN

A MIX PUBLICATION

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Secrets of the
CASIO CZ-101

Interview:
HOWARD JONES

Software Reviews:
MAC and COMMODORE 64

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PCM RECORDING

Boost your
FOSTEX X15
output level

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And then there's the heartbeat of YCAMS, the RX11 (or RX15) digital rhythm programmer.

All its drum sounds are digital recordings which can be patterned in either real- or step-time. And controlled to a resolution of 1/192nd of a beat. Which along with the SWING function give the RX11 a distinctly human feel.

But there's really no way the printed word can fully convey the power, the scope, and the sound YCAMS offers the serious composer and musician. That can only be done by using the most powerful musical tool, your ear, to listen. So we suggest you go to your authorized Yamaha Professional Products dealer for a complete demonstration.

For complete information, write: Yamaha International Corporation, Professional Products Division, P.O. Box 6600, Buena Park, CA 90622. In Canada, Yamaha Canada Music Ltd., 135 Milner Ave., Scarborough, Ont. M1S 3R1.

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VOL. 2, NO. 3 MARCH 1986

Electronic MUSICIAN

A MIX PUBLICATION

DEPARTMENTS

COVER

Electronic musician Howard Jones seeks to combine high technology with a human touch. On his last world tour, he used a Yamaha KX5 remote MIDI keyboard not just to control multiple synths offstage but also to give him the freedom to mix with the audience. Photo by John Bellissimo.

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ARTICLES

Editor's Note

Did you know that you are a member of EM's editorial staff? You didn't?!?! Well then, let's sit down and discuss this for a bit.

EM is shaped by your comments, your feedback, your ideas, your articles, and your concerns and interests. All of us here at EM are working very hard to produce a publication that meets your needs, but we can't always do it by ourselves . . . which is where you come in.

We need your feedback—which articles you like, which ones you don't like, what you turn to as soon as you open up the magazine, even how you feel about the graphics and covers. Send us a postcard, a letter, or even a photocopy of the Table of Contents with comments about each article. Do you like do-it-yourself projects? What about reviews . . . is there any way we can enhance their usefulness to you? And what about the overall level of the magazine . . . do you find articles too advanced, too basic, a good combination of advanced and basic, or what?

We also need your feedback if you disagree with points of view or opinions presented in our articles. We don't censor authors as much as some magazines, because we feel that all points of view should be presented. But that means that if you disagree with something, write in! In such a rapidly-changing field, it's almost impossible to always be the Definitive Last Word about everything—so don't be shy if you feel we've overlooked something, or if you have an update to an item that already appeared in print.

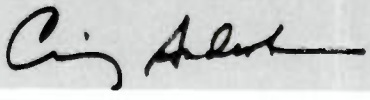
Now is the time when your comments will be most valuable-during those crucial months when the magazine is acquiring its new "personality." I realize, though, we're asking you to do a bit of work, so let me explain what's in it for you.

First, your letter will be read and carefully considered . . . and not just by one person, either. Your ideas will come up during editorial discussions, and possibly even form the basis for future issues. So don't think your writing is in vain—we are anxious to receive your comments, and you can bet we give them our undivided attention (because of this, though, we often don't have time left over to acknowledge your letter; we hope you understand).

Second, you'll get exactly the kind of magazine you want. One reader wrote in and wanted to read an article comparing the different types of synthesis; that seemed like a good idea, so we have an author working on it now.

We're working on putting out the niftiest music magazine in the world, but with that kind of ambitious goal, we could use some assistance. We need your feedback, so after you've had a chance to sit down with this issue, why not take a few minutes and jot down some comments on what you've seen so far? It will benefit us, it will benefit you, and it will benefit electronic musicians everywhere.

Thanks for your help . . . and a special thanks to those who have already written in with ideas and suggestions. Welcome to the editorial staff.



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THE ULTIMATE ACHIEVEMENT IN CONTROL ROLAND'S SRV-2000 DIGITAL REVERB

Roland

Reverb, truly realized. Consider this: assemble every conceivable parameter of natural and plate reverberation, incorporate the possibilities of non-linear (gated) reverberation, augment these with a parametric equalizer, use a 16-bit A/D/A converter and a 28-bit parallel-operation signal processor. Put all of that under computer control for one-button convenience, and complete the picture with MIDI control for (no button) convenience. Roland has not only considered these ideas, we have realized them, in the SRV-2000 MIDI Digital Reverb. Roland Corp. US, 7200 Dominion Circle, Los Angeles, CA 90040.

DIGITAL SIGNAL PROCESSING FROM ROLAND

Letters

Is This the Right Magazine?
I recently subscribed to *EM*. Somehow I got lost in all the digital talk, like MIDI. Could you refer me to a basic scenario on such a critter (MIDI) as I'm not familiar with one. I had hoped that all aspects of electronic music would be covered, i.e. vintage guitar amps, troubleshooting techniques, modifications, and hard-to-find schematics as well as the new stuff. I do have a background in electronics, so I hoped your magazine was what I was looking for—but I'm not sure.

Dave Lathom
Bellingham, WA

Dave—We have just the article for you in this month's issue; see what David Albin has to say about setting up a MIDI studio. For basic information on MIDI, check out the two books mentioned in last month's "Letters" column—I think you will find them very helpful. We will be having more MIDI coverage in the future, and the more you read about MIDI, the more the pieces will fall into place. As to whether we will cover all aspects of musical electronics, that is our goal—but remember, we can't cover it all in one issue. Over the course of the year we will have presented a wide variety of features, from basic to advanced, from guitar to keyboard to drums, from high-tech to low-tech.

Regarding basics, if something has already been published, there is usually no reason for us to be redundant; dated material will often be referenced in bibliographies, and you can then obtain copies of the articles that interest you. However, we also understand the need for basic-level tutorials—there are always new people coming into the field. In fact, you can expect more entry-level tutorials on a number of subjects in the months ahead.

As always, the magazine will be shaped by your input—if we get enough letters asking how to control microwave ovens with theremins, we'll do our best to find someone who can write an appropriate article.

More Just Intonation!
Is David Doty's article on Just Intonation in the August 1983 issue of *Polyphony* still available? Have you published any more articles on Just Intonation since the one of February 1984 by Vanessa Else? Are any in the works?

Pauline Phillips
Centralia, MO

Pauline—In the April 1985 issue, Polyphony *published David Doty's "Justly Tuned Guitar" article. Contact Polyphony Publishing, 1020 W. Wilshire Blvd., Oklahoma City, OK 73116 for information on back issue price and availability. While not that many people have experimented with JI, the subject is getting more attention than ever before; we have several JI-related articles planned for future issues, which touch on various aspects of the subject. For more information on JI in general, contact the Just Intonation Network, 535 Stevenson St., San Francisco, CA 94103 (tel. 415/864-8123).*

MIDI Volume Pedal?
Is there such a thing as a MIDI volume pedal? This would transmit pedal position as velocity data, and likewise receive velocity data and use it to control volume. This way, a MIDI keyboard with no velocity sensitivity could respond to dynamic information from a master controller, as well as send such data to other instruments. If no such device exists, how about a construction project along these lines?

Leslie Bildner
Howard Beach, NY

Leslie—The closest I've seen to what you want is Akai's ME15F four-channel MIDI fader, which uses four standard pots to control the mix of four keyboards by controlling velocity data. One of these pots could be mounted in a footpedal. However, I have also heard rumors that a device which corresponds exactly *to what you want is under development; I've been looking for something like this for quite some time myself, in order to add dynamics to an OB-8 by using a master keyboard with velocity. Concerning construction projects, if someone has something along these lines we would certainly like to take a look at it.*

RM/PLL Update
The RM/PLL article in the September 1985 issue looks great; I'll try building it. But I can't tell from the schematic where R8 and R17 are supposed to connect. Thanks for your help, and by the way, *great* magazine. Keep up the good work!

David Tunbo
Arlington, TX

David—R17 connects to V- and R8 connects to V+. Good luck with the project, and thanks for the compliments. EM

Operation Help

Operation Help *is dedicated to helping musicians help each other. If you need technical assistance, a schematic for some old piece of gear, or just want to connect with people having similar interests, send your name, address, phone number (optional), and nature of your request to* Operation Help, Electronic Musician, *2608 Ninth St., Berkeley, CA 94710. There is no charge for this service, but we cannot guarantee that all requests will be published.*

Tel-Ray Echo: I need information on the Tel-Ray Electronics ADD-N-ECHO Model 1001 (serial #11409); specifically, how does it work, how much oil should be in the can, and any repair techniques. John LeClaire, 114 Batoche Cres., Saskatoon, Sask. Canada S7M-5B3.

ARP Keyboard, CP/M MIDI: I am in search of any specifications, documentation, or prints for a three octave electronic music keyboard purchased from a computer products liquidator in 1983. The keyboard has the following information: ARP Made in USA Board E, ASSY 7203101, FAB 5402401, COMP SIDE and FAB 5402401A, CKT SIDE. The assembly frame is stamped 101078. The keyboard has a six-bit output and includes several LSTTL chips. I am also looking for any MIDI software that runs under CP/M 2.2. Joe Nash, c/o PORTEC ASD, 4500 Western Avenue, Lisle, IL 60532. Tel. 312/810-0500, ext. 213.

Mu-Tron Bi-Phase: I need a schematic for the Mu-Tron Bi-Phase. I am also interested in schematics for any equipment from the following manufacturers who are no longer in business: Electro-Harmonix, Mu-Tron, and MXR. Lee W. Hoover Jr., 325 Saude Ave., Essington, PA 19029. Tel. 215/521-2933.

Marshall Amp Effects Loop: I want to add a buffered effects loop to a Marshall 50 Watt master volume head; it is equipped with the Mark II series master volume rather than the hot rodded "California Master." I am also interested in corresponding with anyone who can steer me in the right direction on Marshall mods, specifically preamp/tone fine tuning and power amp compression. Gordon Power, #303-10021 W. Higgins Rd., Rosemont, IL 60018. Tel. 312/696-1797. EM

GET ON TRACK!

Whether you're recording original music scores, layering up sound effects, or synchronizing to video or film for audio-post sweetening, you need a tape recorder that's built especially for your new and exciting business. The MX-70 is the perfect multitrack for the synthesizer oriented studio tied together with MIDI.

The "70's" three-way design gives you 7.5, 15 and 30 ips in a 1" 16-track, a 1" 8-track, or a 1" 8-track prewired for 16. (An optional ½" 8-track is also available.) Noiseless, gapless, punch-ins and punch-outs provide quiet, inaudible inserts into pre-recorded program material. The MX-70's wide dynamic range makes for quiet, clean recordings. In fact, you'll find the "70" at 30 ips is so quiet that noise reduction just isn't necessary. And to complete this perfect package, you can add an optional autolocator to the standard full function remote for complete session control.

So if synthesizers with MIDI, or SMPTE with film and video, is part of your business, check the specs and don't settle for less than the MX-70. Call your nearest Otari dealer for **"Technology You Can Trust".**

Otari Corporation, 2 Davis Drive, Belmont, CA 94002, 415/592-8311, Telex 9103764890

OTARI®

Here's your chance to make suggestions to the industry. Is there some feature you would particularly like to see on equipment? Something you particularly don't like? Something you like which you wish more manufacturers would implement? Send your complaints, compliments, and ideas to: Suggestion Box, Electronic Musician, 2608 Ninth St., Berkeley, CA 94710. We also welcome suggestions from manufacturers to users of individual pieces of equipment.

MIDI interface cabling: Most MIDI interfaces that plug into the Commodore-64 have connectors mounted directly in the interface. Therefore, if you tug on the cord accidentally, a tremendous amount of stress is placed on the interface and the computer's edge connector. One exception I know of is the Passport interface, where the MIDI connectors mount on flexible wires which then connect to the interface cartridge. This is a much better way to do things, since you have a certain amount of slack should you accidentally pull on the MIDI cables. I hope more manufacturers adopt this approach. *Chuck Hosman, Los Angeles, CA.*

Availability of accessories: Stores should be more conscientious about carrying after-the-sale accessories for equipment. I recently bought a printer for a music computer, but ribbons were nowhere to be found. It took a call to Japan to straighten things out. Accessories may not make stores a lot of money and I wouldn't expect a store to stock every accessory in existence, but I believe that if a store sells you a piece of equipment, they have a responsibility to make the accessories to that equipment available with a minimum of hassle. *Pauline Anna Strom, San Francisco, CA.*

Computer program overlays and "cheat sheets": It would be great if computer programs had keyboard overlays that indicated the program functions of the computer keys. These already exist for computer programs like BASIC and word processors. You would simply place the overlay over the keyboard. If this approach is too expensive, then every program should alternately include a stiff cardboard or plastic "cheat sheet" listing the commands in short form. This would prevent having to constantly refer back to the manual when you're trying to find out the syntax for one specific command or function. *Robert Landry, Memphis, TN.* EM

Ecstacy

GET IT?

Too cute? Maybe, but a good description, since a demonstration isn't practical here. All we can give are the cold facts: 16Khz bandwidth (modifiable with the HF roll control), 8 programs, including gated drums, reverse reverb, some small structures, a few nice rooms and an absolutely huge hall. All very different and very usable. Zero to 200 Ms of predelay, continuously adjustable... variable decay time from very little on the small programs to over 10 seconds on the big ones... stereo inputs... stereo outputs... front panel mixing... infinite hold for looping vocals, strings or whatever into a never ending note. All the features and controls to duplicate any digital reverb sound you've ever heard, and a few you probably haven't. Want to know what Ecstacy sounds like? Hear it today at better music stores everywhere.

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The Mirage is versatile. There are 37 playback parameters to give you a wide range of control over envelopes, filters, tuning and key velocity effects. There's even an on-board sequencer that can sync with other units via MIDI. So the Mirage is everything a synthesizer is—and more.

The Mirage is affordable. Ensoniq's advanced technology makes the Mirage a reality for under $1700[1]. See your authorized Ensoniq dealer today for a complete demonstration.

1. Our legal department panics whenever we say "under $1700". The manufacturer's suggested retail price of the Mirage is exactly $1695.00. ENSONIQ and Mirage are registered trademarks of ENSONIQ Corp.

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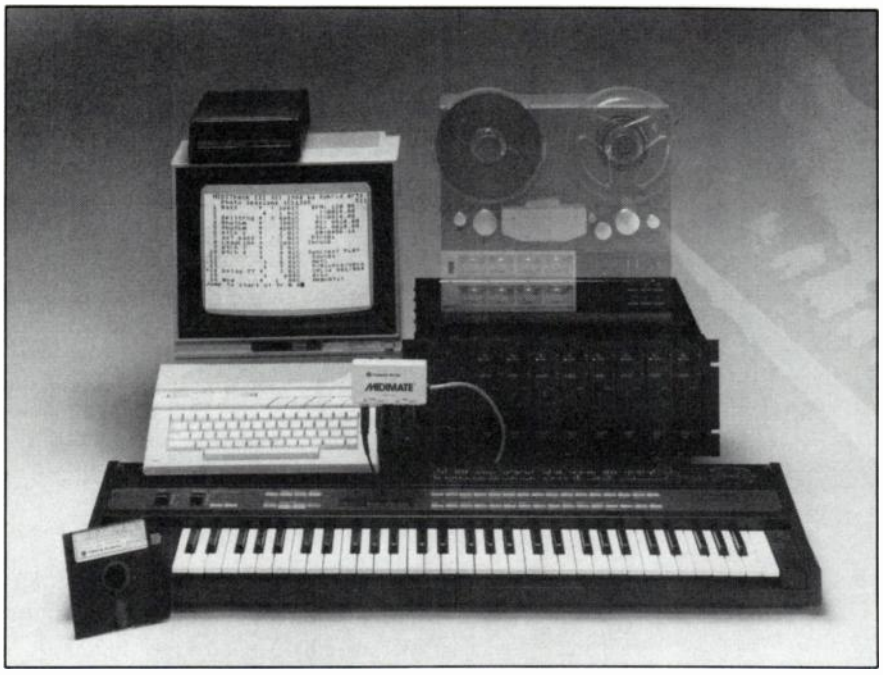

Hybrid Arts' MIDITrack III

COMPUTERS AND SOFTWARE

MidiTrack III ($374) for the Atari 130XE computer, an upgraded version of MidiTrack II, is an advanced sequencer that offers a capacity of over 10,500 notes, song position pointer, and SMPTE lock-up (but not SMPTE locate). The package includes the hardware interface, software, two MIDI cables, and documentation. Coming soon: **MidiTrack ST**, a 24-track MIDI/SMPTE recorder for the Atari 520ST. Hybrid Arts, 11920 Olympic Blvd., Los Angeles, CA 90064. Tel. 213/826-3777. Hybrid Arts also maintains a computer bulletin board service; call 213/826-4288.

Those interested in trading Kurzweil 250 samples should contact **Sweetwater Sampling Network**; those who aren't yet sampling can purchase disks for $5. Sweetwater Sound, 2350 Getz Road, Ft. Wayne, IN 46804. Tel. 219/432-8176.

The **Chroma MIDI Converter** ($349.95) attaches to the Chroma computer interface jack and provides MIDI In and Out ports (Out doubles as Thru). The converter can send and receive on up to eight MIDI channels simultaneously, thereby accessing the Chroma's multi-timbral capabilities. Also, the Commodore-64 compatible **Keyboard Controller** ($225.95) allows one master keyboard to slave MIDI instruments on any of the 16 available MIDI channels. Its main application is for multi-synth players who want fast preset setups for several slave keyboards; features include three levels of operation, and a stepper that stores combinations of setups. Syntech, 23958 Craftsman Rd., Calabasas, CA 91302. Tel. 818/704-8509.

The **PC-to-MIDI Card** ($295) for the IBM PC, PC/XT, PC/AT, and hardware-compatible "IBM clones" provides MIDI In and Out ports, sync to tape, four switch-selectable addresses for use of multiple cards, and full interrupt-driven capability for both timer and MIDI events. Also, **DXLIB** ($250), which runs on the above card, handles patch dumps and loads, stores over 2,700 sounds on one 5.25-inch disk, and provides voice and performance parameter editing. Noteworthy Systems from Gentle Electric, P.O. Box 132, Delta, CO 81416. Tel. 303/874-8054.

PERCUSSION

Drummers with **Linn 9000** drum machines will be pleased to hear that a library of over 100 drum sounds is now available on disk for the 9000. Linn Electronics, 18720 Oxnard St., Tarzana, CA 91356-1413. Tel. 818/708-8131.

Roger Nichols, whose "Wendel" drum machine has acquired a certain legendary status in the industry, is working on **Wendeljr.** This digital percussion instrument features 16 bit sampling and a 50 kHz sampling rate; it's designed to replace standard drum machine or live drum sounds with sounds of exceptional fidelity. Price and availability will be announced later. Contact Mike Clary, 143 West 89 Street, New York, NY 10024.

The **Duopad** ($225) has both a drum surface sensor and rim sensor; therefore, two different sounds can be played on a single pad. An additional "transition" control adjusts the coupling between the two sounds. Dynacord, distributed by Europa Technology, 1638 West Washington Blvd., Venice, CA 90921. Tel. 213/392-4985.

The **CX5** drum unit (not to be confused with the Yamaha CX5M music computer) memorizes up to 128 different "kits," which draw from 64 PCM-recorded digital drum sounds and 12 tympani sounds. Highlights include dynamics, programmable volume, sequencer functions, sync to tape, and a MIDI interface. An entire set, consisting of four drum pads and the CX5, is slated to list for under $2,000. Wersi, 1720 Hempstead Road, P.O. Box 5318, Lancaster, PA 17601. Tel. 717/299-4328.

Wersi's CX5 drum kit

The MK1 synthesizer from Wersi

Hohner's PK 250 keyboard

Shure's 16L-LC condenser microphone

SYNTHESIZERS AND MIDI

A new synthesizer, the **MK 1** (approximately $2,000), claims sound generation based on a mix of sampling, additive digital synthesis, and frequency modulation. The five-octave keyboard offers velocity and aftertouch. Wersi, 1720 Hempstead Road, P.O. Box 5318, Lancaster, PA 17601. Tel. 717/299-4328.

The five-octave **PK 250** keyboard also includes a MIDI interface and Arranger Accompaniment, a series of cartridges with built-in drums, bass, and synth sounds to provide a full-band sound. Hohner, Lakeridge Park, Sycamore Drive, Ashland, VA 23005. Tel. 804/798-4500.

MIDI Merger (approximately $250) "mixes" two MIDI signals together and sends them down a single line. This is useful for, say, mixing drum machine timing data and keyboard note data simultaneously into a sequencer. MIDI Merger also includes a channelizer to send MIDI info down any channel, along with Aftertouch and System Exclusive filters. Kamlet, P.O. Box 916, Indian Hills, CO 80454. Tel. 303/697-0296.

Mediasound now features a MIDI studio that provides music editing, scoring, sound library, and similar MIDI studio facilities using sequencers, computers, a Synclavier, guitar-to-MIDI controllers, and the like. Mediasound, Studio D, 311 West 57th St., New York, NY 10019. Tel. 212/765-4700.

The **MIDIMIX 6** MIDI splitter ($35) provides five buffered outputs from a single MIDI output. It draws power from the MIDI line itself, and therefore requires no power or batteries; it's also faster than many Thru boxes, since it includes no opto-isolators (it instead drives the ones used in the instruments themselves). MIDIMIX, P.O. Box 161, Ashland, OR 97520. Tel. 503/488-1023.

RECORDING

The **16L-LC** condenser microphone ($98) is a low-impedance, unidirectional, battery-powered microphone intended for home recording applications. Shure Brothers, 222 Hartrey Avenue, Evanston, IL 60204. Tel. 312/866-2573.

Industry-standard forms and labels are available from **StudioForms**. These include cassette labels, track sheets, MIDI sequencer track sheets, quote and work order sheets, equipment inventory, and considerably more; my favorite is the engraved self-sticking "No Drinks or Food on the Console Please" plaque. StudioForms, 186 Glen Cove Avenue, Suite

201/EM1, Glen Cove, NY 11542. Tel. 516/671-1047.

VIDEO

The **VC-C10UA** VHS camcorder ($1,899.95) features automatic focusing, exposure and white balance, 6X power zoom, and macro focus. It can be powered via AC adapter, battery pack, or car battery. Sharp Electronics, 10 Sharp Plaza, Paramus, NJ 07652. Tel. 201/599-3734.

Sharp's VC-C10UA VHS camcorder

SIGNAL PROCESSORS

Prices for the **Ursa Major StarGate 323** and **626** have been reduced to $1,300 and $1,800 respectively. Ursa Major, P.O. Box 28, New Town Branch, Boston, MA 02258. Tel. 617/924-7697.

The **MIDI/224 Interface** for the Lexicon 224XL allows for program change and assigning of reverb parameters to MIDI controller numbers. The **MIDI/224V** also provides eight voltage or resistance outputs, thus retrofitting existing processors and synthesizers to accept MIDI control. Electron Farm, Nelson Lane, Garrison, NY 10524. Tel. 914/424-4071.

PUBLICATIONS

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Industry Trends

BY STEVE SAGMAN

One of the hottest words in the electronic M.I. (musical instrument) trade during '85 was "sampling." Several American manufacturers (E-mu, Ensoniq, Kurzweil, Sequential and Linn) introduced new sampling instruments or options, and several smaller computer hardware/software vendors (Armonyx, Decillionix, Fractal, Mimetics, and SFX Computer Software) offered user-installed sampling boards and software for computers including the IBM PC, Apple II, Commodore-64, and Macintosh. Sampling is about to be further championed by the coincidence of two market forces.

Commodore's Amiga computer is still an unproven commodity despite strong sales to computer hobbyists. However, new software to direct its built-in sound sampling capabilities should soon appear. Commodore's marketing strength will ensure that the word "sampling" becomes even more conspicuous in both the music and computer media.

Alongside developments in the U.S., the Japanese have again demonstrated their remarkable ability to reproduce American technology at significant cost savings—particularly when it requires packing complex electronics into tight spaces. The result of their effort is usually comparable, or even superior, engineering at lower prices. The leader in this game, Casio, is about to do it again. Since their introduction of the CZ series digital synthesizers (see review in the August **EM** and related article in this issue), Casio has virtually dominated the low-end digital synthesizer market. Their next target appears to be the sampling market. Their weapon: a sampling keyboard priced for the home musician.

There are instruments from other manufacturers making their way to the U.S. market, too. Last year, Akai presented the S612, an inexpensive sampling rackmount module; now, watch for new budget sampling devices from Korg, Roland and Yamaha, as well as another sampler from Akai that is rumored to provide 48 seconds of sampling (albeit at reduced bandwidth). These developments will bring sampling to electronic musicians who last year could only dream of easily affording such stunningly sophisticated electronics.

In a similar vein, Casio's successful foray into the low-end digital synthesizer market (low end refers to expense, not quality) has attracted the attentions of other Japanese manufacturers. In addition to inexpensive samplers, several low-priced keyboard instruments from Yamaha, Roland, and Korg are about to appear at music dealers. Often, these instruments have the same technology as their more expensive bigger brothers and sisters, but include fewer features or are of a smaller size. Modern MIDIfied musicians have been quick to use these devices as slave synthesizer modules, connected to their master controllers through MIDI. So, although these instruments might appear to be aimed at junior musicians and home hobbyists, they are equally attractive to more serious players. With a market potential like that, it's clear why several companies are stepping in to fill the gap.

There are several other market gaps that are about to be filled, but that is a story for next month . . . see you then.

Steve Sagman is a New York-based keyboard player, marketing consultant, and editor of MIDI Marketer *(an industry newsletter covering the electronic musical instrument business). When he's not entangled in MIDI cables, he leads an acoustic jazz quartet.*

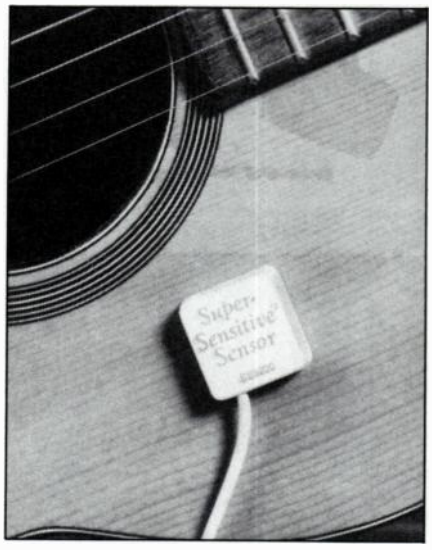
tions (single copies $3). P.O. Box 9911, Oakland, CA 94613.

Aftertouch, edited by former *Keyboard* editor Tom Darter, is Yamaha's free monthly publication that presents information on the use of the X series line of products. Aftertouch, P.O. Box 2338, Northridge, CA 91323-2338.

The **Canadian MIDI Users Group** puts out a newsletter that provides information on interfacing, hints, and software, as well as offering a "members only" classified section. CMUG, P.O. Box 1043, Belleville, Ontario, Canada K8N 5B6. Tel. 613/968-9559.

TELECOMMUNICATIONS

Bryan Bell (known for his work with Santana, Herbie Hancock, John McLaughlin, and many other artists) has started SynthBank™, a telecommunications service for patch sound downloading into instruments. Initial offerings include public domain software from a variety of manufacturers (Casio, Chroma, E-mu, Fairlight, Mirage, Oberheim, Synclavier, and Yamaha DX). For more information, contact Bryan Bell over the IMC network (BELL-US), PAN network (BRYANBELL), or write to 12080 Southwest Parkway, Portland, OR 97225.



OTHER NEWS

The **Super-Sensitive Sensor** ($39.95) is designed to pick up sound from any vibrating surface. A special adhesive allows for easy relocation without marring the instrument. Super-Sensitive Musical String Co., 6121 Porter Rd., Sarasota, FL 34240-9542. Tel. 813/371-0016.

Heard enough about instruments that sound *like* a piano? The **Bosendorfer 290 SE** combines a Bosendorfer piano with the efforts of engineer Wayne Stahnke and Kimball's R&D facilities. The result is a computerized system that scans the keyboard as the piano is being played, digitally codes all piano functions, then records the data on audio tape. Upon playback, the data activates the piano to reproduce the original performance with the exact same nuances. Kimball International, P.O. Box 460, Jasper, IN 47546. Tel. 812/482-1600.

RolandCorp US has donated CMU-800 CompuMusic Systems to nearly 500 U.S. school districts that excel in the integration of technology and education. These systems, which work with Apple II computers, are intended for use in existing music education programs. RolandCorp US, 7200 Dominion Circle, Los Angeles, CA 90040.

AEMMP Records is a not-for-profit project of Columbia College Chicago's Arts, Entertainment, and Media Management Graduate program. They are currently soliciting audition tapes from Midwestern musicians for their *Break Chicago* album project. Send tapes to: AEMMP Records, Columbia College Chicago, 7th Floor, 600 S. Michigan Ave., Chicago, IL 60605. Tel. 312/663-1600, ext. 351.

All prices are suggested retail prices, as supplied by the manufacturers. All prices and specifications are subject to change without notice. EM

BY ROBERT CARLBERG

Send records, tapes and videos for review to Robert Carlberg, P.O. Box 16211, Seattle, WA 98116.

D.L. Myers, *Electronic Guitar* (cassette). The obvious precedent for this tape is Fripp & Eno, based as it is around infinite-sustain guitar lines and rhythmic structures built up of layered overdubs. Yet this is only the jumping-off point for Myers (last reviewed 6/85). Side A is called "Vapors" and consists of six pieces, named after various gases, which takes F&E's formula into birdcalls, volume-pedal orchestrations and other territory they didn't get around to exploring. The other side, "Metals," is heavier, utilizing extensive electronic and tape-speed alterations to create some very un-guitar-like textures. Comparisons to Throbbing Gristle or Jean-Baptiste Barriere would not be out of bounds. In all, Myers' truth-in-advertising delivers some very "electronic guitar." $7 postpaid from 228 Bleecker Street, New York, NY 10014.

Mike + The Mechanics, *Mike + The Mechanics* (Atlantic 81287-1). Genesis has become both simpler and more popular since the departure of Peter Gabriel (1975) and Steve Hackett (1977). In fact, the ingredients these two took with them are evident in subsequent releases under their own names. The solo albums by the trio that remains (Phil Collins, Tony Banks and Mike Rutherford) reveal much less about the threads that hold them together and almost nothing about their differences. Each is populated by session musicians and bland to the point of being designed by committee. Rutherford's third, with compositional help on every track and two "name" vocalists in The Mechanics, is a case in point and probably full of AOR hits.

RMS, *Centennial Park* (MMC 004). RMS is a trio of Ray (Russel—guitar), Mo (Foster—bass), and Simon (Phillips—drums). They have worked, separately and together, as session musicians for dozens of well-known sessions. Hot progressive jazz-rock, right? Only partly—a brass quartet and stilted arrangements designed to leave room for them make RMS sound more like Jeff Lorber than I.O.U. They play their asses off but they don't cook.

Mark Egan, *Mosaic* (Hip Pocket 104). Weather Report and Jaco Pastorius opened the door to lead bassists, and Egan walks right through playing his double neck Pedulla. He first showed up in Pat Metheny's band, later splitting off with Danny Gottlieb to form Elements. This record could have been "Elements III," except three of the tracks really are all Egan. Tasteful overdubbing and the harmonically-complex tone of his 4 + 8 string instrument make his bass solos much more palatable than the idea sounds.

Ontario Electronic Music Tape Project 1985 (cassette). There is no doubt that some of the best electronic music—if not *most* of the best electronic music—comes from independent producers. If you have some doubts, an investment of $6.50 (USA) will prove it to you. D. Butler, Cedar Creek Sound, P.O. Box 1296, Woodstock, Ont. N4S 8R2 Canada.

Brian Eno, *Thursday Afternoon* (Editions EG 64; CD only). The Compact Disc offers three advantages over LPs: longer lengths, no wear, and wider dynamic range. This last means not only louder louds, but also quieter quiets—and it is this which attracted Eno to the medium. In the tradition of *Discrete Music* and *Music For Airports*, *Thursday Afternoon* is one long (nearly 61 minutes) droning background consisting of gently bonging reverbed piano over a steady synthetic string chord. It's boring as hell but that is the intention.

Gary Numan, *The Fury* (Numa 1003). Gary Numan hasn't scored any movies, guest-starred on Miami Vice or recorded a duet with Kenny Rogers. He has gotten big enough that Roland "supplies keyboards" in return for printing their logo on the lyric sheet. He also still turns out a finely-crafted record, adding a few new dimensions to his admittedly limited vocal prowess.

Yes, *9012Live* (Atco 90474). The title is a weak pun on *90125*, the album without a title from which these live tracks derive (also available as a video). Subtitled "The Solos," only two tracks from the album reappear here—the other five are solos spotlighting individual members (mercifully, the drummer's spotlight is shared with Chris Squire). While up to the usual Yes standards of precision, one wonders if the same megalomania that swallowed the old Yes is really put to bed. EM

Robert Carlberg is the national service manager for Audio Environments Inc., a nation-wide supplier of original-artist music for restaurants and fashion stores. His hobbies are electronics and music, and particularly electronic music. He was co-founder of SYNEX, *a newsletter for electronic musicians published during the late '70s.*

Are you interested in MIDI but don't know where to start? Here's how one person set up a MIDI studio—and more—for well under $1,000.

But How Do I Get Started?

BY DAVID ALBIN

Have you ever felt like you were on the outside looking in? Well, a lot of people feel that way about MIDI. When you hear all the talk about bits and bytes and disks and serial ports, if you don't know the lingo you may as well be listening to Martian.

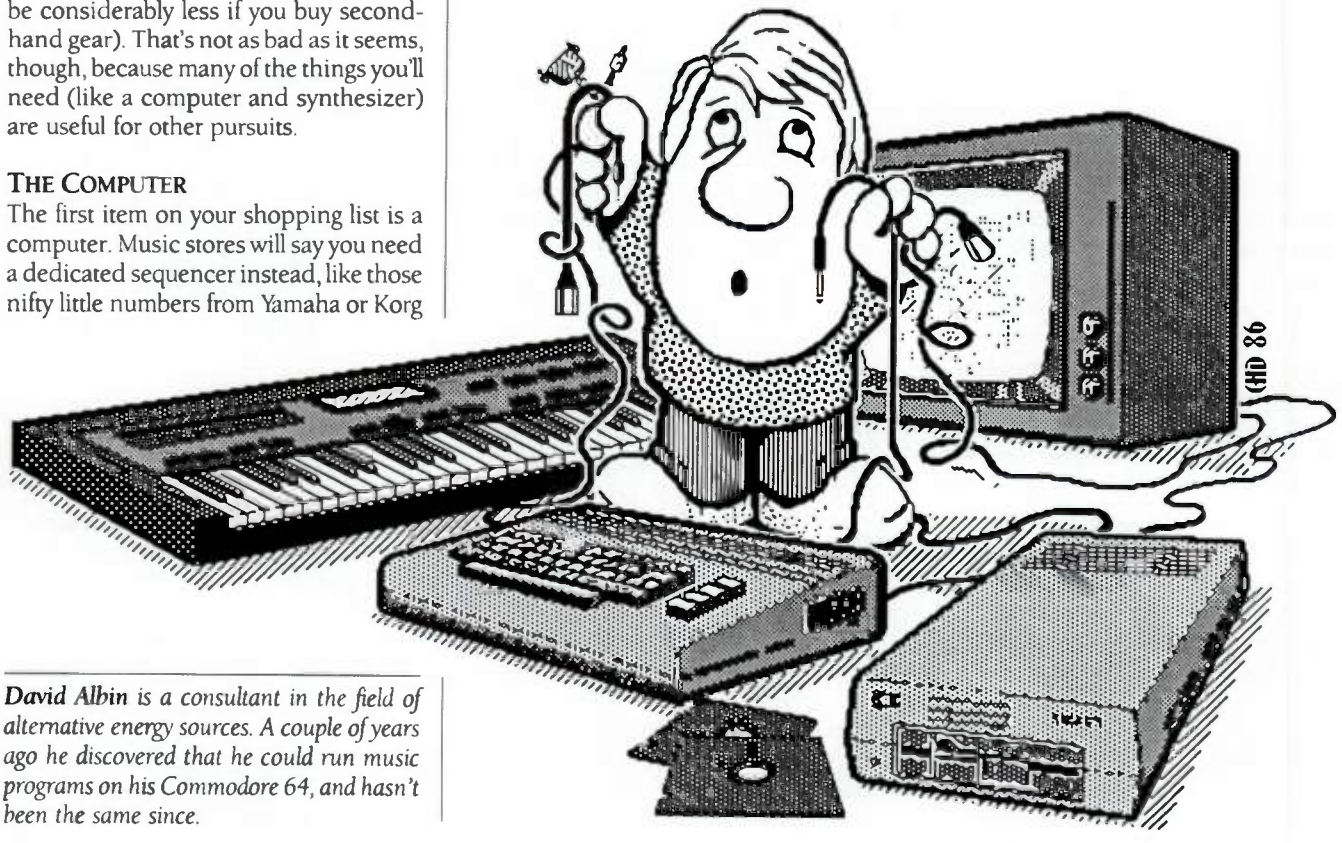
Take heart. I don't know all that much about computers, but over the past few years I've learned enough to be dangerous and certainly know enough to have a really great time with computerized music. It's not as difficult as you might think, but you need some gear—a "starter" MIDI lab, as it were—so that you can learn about what's going on. The whole point here is minimal cash outlay, but be forewarned: We're still talking about a realistic entry price of about $1,000 (this can be considerably less if you buy second-hand gear). That's not as bad as it seems, though, because many of the things you'll need (like a computer and synthesizer) are useful for other pursuits.

THE COMPUTER

The first item on your shopping list is a computer. Music stores will say you need a dedicated sequencer instead, like those nifty little numbers from Yamaha or Korg or Casio. But you can play Zaxxon as well as MIDI on a home computer, so as far as I'm concerned it's no contest . . . get the computer.

Everyone else is going to tell you that you need the Ultimo WhizBang computer with enough memory to put a herd of elephants to shame and operating speeds so fast you actually go backwards in time when you use the thing. But when you bought your first car, you didn't buy a Mercedes. The object here is to get something that works. Once you know more about the subject, you can move on to something better. By then, however, you'll know how to intelligently spend the bucks necessary to buy a good machine that fits your needs exactly.

So, go to your nearest discount toy store and head for the Commodore 64 display (or if you want to be upscale, go for the Commodore 128). The C-64 is the VW of computing, the people's computer that is ugly, slow and extremely functional. It's an 8-bit anachronism in a 32-bit world and its disk drive teaches you the true meaning of patience. So much for the bad news. The good news is that the computer regularly sells for $140 or less, and Commodore-compatible disk drives cost as little as $125 (the official version costs about $200). So, for $265 you have a new computer and drive. Because of its popularity over the years, the C-64 is well supported with software—from fun and games to serious stuff, like word processing and spreadsheets—and many small companies make all kinds of C-64 music software, including patch librarians and other useful stuff. You're not just buying into a computer, you're buying into a club with a bunch of members. Even my home town has a Commodore Users Group, and I can choose from several Commodore magazines at my local newsstand. Though Commodore's manuals are terrible—the disk drive manual is particularly atrocious—there are some excellent books on the C-64 available from other sources (such as the titles put out by Osborne/McGraw-Hill).



David Albin is a consultant in the field of alternative energy sources. A couple of years ago he discovered that he could run music programs on his Commodore 64, and hasn't been the same since.

Don't try to live without the disk drive; most good music software requires it. Grab at least a dozen disks, too. While you can take a year to fill up a disk with text, music applications use up disks fast . . . and you'll want enough disks to back up important data (never trust a computer).

Scan the papers and ask your friends about second-hand gear. Affluent musicians are always trading up to heavier-duty toys, and since the resale value for older stuff isn't really too good, you can often pick up a bargain. Lots of computers show up in suburban shopper papers for cheap, and are often in excellent condition because they were used once on Christmas and never again. (And of course, the C-64 isn't your only choice; Apple II computers, which are more sophisticated than the C-64, are often available for a good price; also, the Atari 130XE is quite inexpensive.)

The sound chip in the C-64 isn't bad, especially if your hearing has deteriorated somewhat over the years, and you like a little of that soothing white noise in the background. There are some decent software packages that let you get some good sounds out of the chip. In fact, if you're really tight for bucks all you need is the C-64 and some software like MusiCalc or Music Shop to get going. But you'll be much better off if you decide to commit to MIDI. This means more bucks, but more fun, and it also means you'll need . . .

The MIDI Interface

Most computers can't talk MIDI—just try plugging a MIDI cable into the back of the C-64 and you'll see what I mean. So, you need a MIDI Interface, a device designed to separate you from another $100. The interface has the unglamorous, but highly necessary job of translating computer talk to MIDI talk.

Some interfaces will cost you about as much as the computer. Sorry about that. I'd recommend a Sequential Model 64 MIDI interface, which works just fine and lets you synchronize drum machines to your playing; it cost me $100 second-hand. It also has a primitive sequencer built-in, but thankfully when you get tired of it (which doesn't take too long) several advanced sequencers and other programs will work just fine with this interface—all you need is a new program disk.

The Sequencer Program

As noted, you can just use the one in the Sequential 64, particularly if you have time on your hands and are easily satisfied. Later on, you can graduate to something like Syntech's Studio I or Dr. T's Keyboard Controlled Computer; neither are cheap but they both work well. Several other companies make music software for the C-64 (Passport, Hybrid Arts, Sight & Sound, etc.), but some programs only work with specific interfaces—let the buyer beware.

The Synthesizer

The Casio CZ-101 gets my choice as "best buy" for budget MIDI setups. While the keys are designed for midgets, the thing sounds great and most importantly from a MIDI standpoint, it lets you do multi-timbral work (i.e. you can sequence four independent monophonic sounds at once, so one synth makes a lot of sound). The only drawback is lack of a keyboard with dynamics, but you can't have everything. CZ-101s are even available discounted and are worth every penny. When you hit the big time, you can always use it as a MIDI bass box or expander for some other synthesizer.

The Other Stuff

Now you need two MIDI cables (go to your music store), and life is certainly much better if you have a drum machine of some kind. Cheap analog drum machines are often available secondhand, but new machines (like the inexpensive models from Roland or Korg) can still cost under $200. You also need a television for the C-64, but most people have access to one of those already.

The Reward

Remember, you haven't just bought a MIDI music-making system. You now have a computer on which you can learn programming, enter programs that appear in *EM*, maintain a mailing list, play games, get into graphics, write lots of letters (I have a friend who wrote a book using the C-64) or whatever. The CZ-101 is a portable keyboard, so you have something that's useful outside of the MIDI studio context . . . amaze your friends by plugging it into the Aux input on someone's stereo at the next party. Most importantly, you can learn a lot about MIDI with a simple setup like this, including sequencing, synchronization, and so on. As finances permit, you can add on goodies like patch library programs, which make the CZ a lot more fun with which to play and program. Try sequencing your own back-up band with the CZ and the drum machine. If the time comes to move up to something more substantial, congratulations—sell off your old system, which will let someone else get started for a minimal cash outlay.

You do have to spend a little something to get started with MIDI and computers, but that's true of anything (priced any sailboats, skis, or telescopes lately?). What you get in return is a lot of fun, and something that doesn't go obsolete in a month (just think how long some people have had their C-64s). Good luck! EM

MIDI Basics

For those who aren't familiar with MIDI, it's not all that complicated. MIDI is a computer "language" that expresses musical parameters in a computer-compatible format. For example, as you play a keyboard, the MIDI Out connector spews out data ("words" of the MIDI "language") describing your performance (when you push a key down, when you let it up, the dynamics of your playing, what synth patch you have selected, and so on). If you send this data to another keyboard's MIDI In connector, the second keyboard will "double" your playing as it receives the data being sent from the main keyboard.

What makes MIDI so powerful, though, is that MIDI data can be stored in computer memory and played back at a later time. If you play MIDI keyboard data into a suitably-equipped computer, it will "remember" the musical parameters of your performance, just as tape remembers a performance using analog techniques. Like tape, you can play back the recorded data (typically into the keyboard's MIDI In connector), whereupon the keyboard will replicate your original performance. This resembles the way in which a player piano re-creates a performance by storing note data on piano rolls. The only difference here is that we're storing note data in computer memory.

There's much more to MIDI than this, but those are the basics. To learn more, read about MIDI, reviews about MIDI gear, this magazine, and of course, whenever possible, experiment with MIDI equipment and techniques.

—*Editor*

Want to form a band, but can't find people with compatible musical personalities? Then find compatible machines and teach them to play exactly what you want. Our editor goes in search of new things to write about, and gets more than he bargained for...

Confessions of a MIDIot

(or, How I Learned to Stop Worrying and Love the Band)

BY CRAIG ANDERTON

It started off as a cross between a bar bet, a science project, and some as yet unexplained biological urge: after not playing live for over a decade, I was going to put together a live act...and not just any live act, but a MIDI band. Possible positive results would be fun, enough material for lots of articles, the chance to try out songs in front of a real live audience before committing them to tape, and personal satisfaction. Possible negative results would be equipment not working, spending excessive amounts of time trying to put the act together, and/or public humiliation.

Before MIDI had really taken hold, I had considered doing a solo act based around me, an Oberheim DSX sequencer, OB-8, and Drumulator. Unfortunately, lack of memory ended that—the concept of doing cassette loads every few songs seemed like an invitation for a quick, painful on-stage death. Then MIDI and the Emulator II (which includes a built-in MIDI sequencer) came along, which made the whole operation more manageable, as did adding another player to the band. The OB-8 now fills the function of a playing keyboard (it's not sequenced). The Emulator II's sequencer replaces the DSX as the "MIDI brains" of the organization; it provides timing pulses to the Drumulator and MIDI data to an SP-12, as well as bass MIDI data to a Casio CZ-101. (Of course, the E-II is pretty upscale, but similar results could be obtained with less expensive MIDI sequencers and a couple of decent MIDI synths.)

Overall, this experiment turned out to be well worth doing. The experience has been fun and educational, we got to make a low-budget video, and there are more bookings in the future. As far as I'm concerned, the MIDI band is definitely a low-stress way to get your music out to the public. If you decide to give it a try, here are some tips you might find useful.

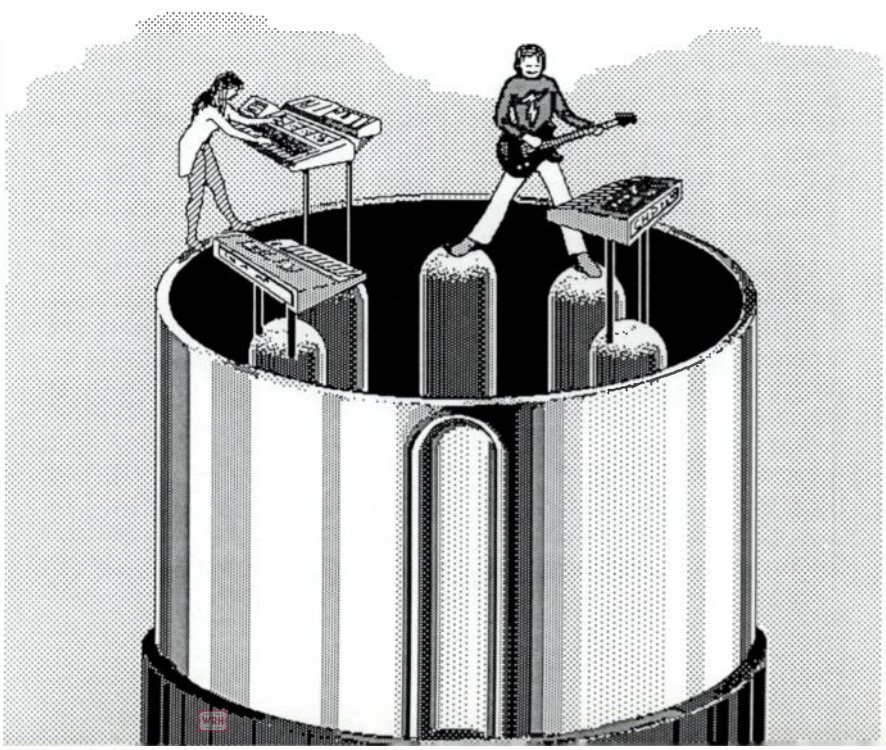
Memory is precious—conserve it. The concept of using several cassette interfaces on stage is horrifying, and cheap disk drives aren't much better. One solution is to expand an instrument's RAM; many service technicians offer this type of mod. For example, after installing a Steve Simpson 5X memory expander in my Drumulator, it could be programmed with an entire set—no more mid-set cassette loads.

As a sequencer, you might think that the E-II's half a megabyte of memory would be more than enough for anything. However, while sequences stored on disk don't take up a lot of memory, sampled sounds (which are stored on the same disk as the sequences using those sounds) do. To free up as much memory as possible, we sequenced other instruments via MIDI rather than just use the E-II's internal sampled sounds. By data compressing the Emulator II samples, and by not going overboard with overdubs, one disk change sufficed for 40 minutes of music.

With most sequencers, filtering out unneeded data conserves memory. In particular, avoid aftertouch, pitch bend, and velocity since recording these parameters uses up lots of memory. The tradeoff, unfortunately, is less expressiveness in the parts you sequence. (The memory tradeoff, which you run into every time you use a computer, is one of the best arguments for getting the most powerful computer you can afford.)

Programming takes forever. You might think that putting together a programmable back-up band is easy: just program a couple of bass parts, some drums, and some sweetening...and all the work is done. Well, as I added overdubs and took live performance into account, the nature of the song would shift, and I would have to go back and tweak up old parts to compensate. Sometimes I would realize that a song needed to be longer live than on record; going back and re-recording parts took a *considerable* amount of time. Recording a set's worth of music into sequencers takes as long as recording an album—maybe even longer, if you take the time needed to learn about the equipment into account. Expect to spend lots and lots and lots and lots of hours programming your parts, shifting patches around to make it easier to call up the right patch at the right time, pre-setting delay programs, etc.

Learn to talk to the audience. It takes time to change presets, load disks, switch



memory banks, and so on. Until everything implements MIDI totally (song select, song pointer, etc.), much of the setup work will have to be done manually. For a duo situation, I would suggest that one person do all the button-pushing while the other raps to the audience. This keeps up a smooth flow, and if the person doing the talking bores the audience, they can always look at the person who's running around madly and punching buttons. Practice transitions between songs as carefully as the songs themselves; remember, one wrong setting on a drum machine or sequencer, and you're in trouble.

Make back-up tapes. Audiences don't like performers who use tape, and I can see why. But they like acts that cancel halfway through a set even less. As you rehearse, make a cassette of the back-up tracks (drum machine, MIDI bass, whatever) and take them—along with a good quality cassette player—to the gig. Even if you never have to resort to using the tape, just knowing you're covered in case your sequencer blows up gives you one less thing to worry about on stage.

Every move counts. With several people in a band, there are plenty of distractions for the audience. However, an inanimate computer band isn't very lively—so you are the focus of attention *all the time.* Minimize "dead air," and make sure you're always doing something of interest.

Practicing is mostly easier. With a sequenced rhythm section, it's easy for any band member to push "play" and practice along with the work in progress. Practicing with a bunch of machines is an interesting concept, and they're an almost perfect back-up band—on time, no drugs, and no family problems to make them crazy (although they do get sick sometimes, or catch software bugs).

However, there is one big hassle—much MIDI gear does not accept *song position pointer* data, which allows various rhythmic devices to autolocate themselves to a specific point in a song. Therefore, you must generally start all sequenced devices at the very beginning of the song (whereupon they all count together and remain synchronized), even if you just want to practice (or, for that matter, overdub) the last two measures. For this type of practice situation, you're better off making a quick cassette of the two measures in question and practicing to the tape instead of the sequencer.

Expect the audience to be on your side. I didn't know how an audience would accept a combination of humans and machines as a performing band, but I needn't have worried—people find the whole concept rather intriguing, and respect you for being able to make sense out of all those buttons and wires.

Is It Worth the Effort?

Playing live has completed my transformation into a total MIDIot. I now use MIDI in recording, for saving patches to disk, for experimentation, and during the composing process...it has even coaxed me into getting back on stage again. You can fit a drum machine, sequencer, and bass keyboard in the space normally occupied by a single bass drum. In our band, every audio output (except microphones) goes directly into a mixing board. This eliminates lots of variables and is a sound engineer's dream. All our gear fits into a friend's van, and the whole process of setting up and knocking down is almost painless. Take along a VCR and PCM adapter, and you can record all your gigs with near-perfect fidelity.

If you're a songwriter or composer and you've always wanted to play live, a MIDI back-up band just might give you the perfect opportunity. Check it out—I did, and don't regret it one bit. EM

Thanks to MIDI, keyboard players can now strap on a lightweight, inexpensive remote keyboard, control a bunch of synths off-stage, and dance around as the mood strikes them. But is going mobile worth the effort and expense? Kirk shares his views on the subject, and also gives three mini-reviews of available controllers.

Mobile MIDI for Stage and Video

BY KIRK AUSTIN

So why would anybody want a keyboard that hangs around your neck, anyway? While many keyboard players may not want something like that, there are reasons why we are going to see more strap-on keyboards in the future. One of the biggest reasons is Video Rock. Whether you like it or not, Video Rock is here to stay, which means that the visual aspect of a particular act is more important now than it has ever been. The way an act *looks* can actually make it or break it! I'm not saying that I approve of this change, but I have to accept the fact that it is real.

For almost three decades, the electric guitar has been the outstanding instrument in pop music. Perhaps this is because the guitar is such a visual instrument; you can see the notes being articulated as you hear them. This is important—it means that facial expression and body language tie into something visual. Keyboard players normally don't have this visual link; however, portable keyboards do create the visual connection. You can see the keyboardist's finger on the particular note you are hearing, and watch the vibrato being added to that note by seeing the left hand on the wheel. Watching a person stand behind a bank of synthesizer gear is pretty pale by comparison.

Kirk Austin designed one of the first standalone MIDI keyboard controllers. He is employed as an assembly language programmer for an industrial controls company and creates much software for the Mac. In addition to writing for EM, *he also contributes to several Macintosh magazines.*

My feeling is that if keyboards are ever going to compete with electric guitars in Rock Video, the use of hand-held keyboards is going to be required (you heard it here first).

Another advantage of portable keyboards is the way they fit in with the trend towards less equipment on stage. A number of groups are part of this trend including Utopia and, to some extent, Howard Jones. It's interesting that as the technology gets more advanced, it also becomes less visible. Howard Jones, with his Yamaha KX5 keyboard and miniature over-the-ear microphone, is a far cry from the way bands appeared only a few short years ago; the emphasis is returning to the artists themselves instead of focusing on a mass of equipment. Walls of amplifiers are beginning to disappear as the obligatory stage setup. For synthesists, a portable keyboard controller gives the ability to experience the kind of freedom that guitar players have enjoyed for years. In fact, a completely cordless synthesizer setup is at last possible at a reasonable cost.

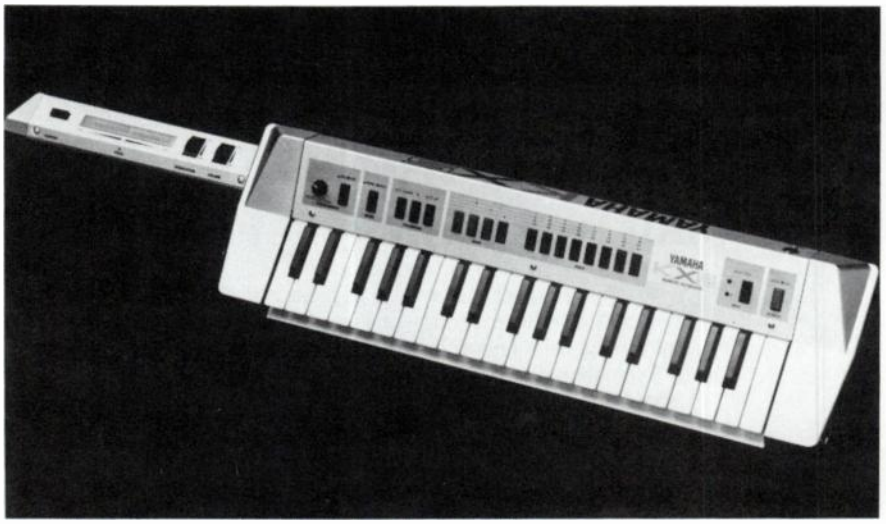
"It's interesting that as the technology gets more advanced it also becomes less visible"

THE EVOLUTION

Let's first consider the history of the hand-held keyboard controller. If memory serves me correctly, there was a hand-held controller made for the original Minimoog synthesizer. I don't remember too much about it except that it was pretty clunky and available as a special order item from Moog customer service. It wasn't really intended to be worn with a strap, but some people undoubtedly did so.

Then there was the one from ARP. The ARP 2600 had a keyboard that could be used as a remote controller, but it was pretty heavy and used two or three different cables. Again, it wasn't really designed to be worn around the neck, but I seem to remember Edgar Winter using it that way.

Another portable keyboard was the Moog Liberation. The Liberation was a complete synthesizer unto itself that could be worn like a guitar and also serve as a controller for other synthesizers. I played with one and found it to be pretty heavy; it was also kind of a drag if you didn't happen to like the particular synth sound that it produced (I didn't). It seemed to me that limiting the remote's function to just that of a separate controller was the



Yamaha KX5 remote keyboard

best way to go.

Prior to MIDI, the most successful implementation of the separate controller that I had seen was The Probe. The Probe keyboard was brought to the public's attention by Roger Powell. This controller, built by (I think) Jeremy Hill, was really the forerunner to what is currently appearing on the scene. Jan Hammer also used one of these controllers, and I'm sure that there would have been a lot more of them around except for the fact that they were pretty expensive. Not only was each case custom made, but the electronics had to be adapted to whatever synthesizer was going to be controlled. This made for a rather lengthy development time for each Probe, along with a distinct lack of flexibility.

MIDI is what made it possible to separate the controllers from the sound producing electronics, and to do so in a cost-effective, practical way. Now a controller can be built that will interface with almost any modern synthesizer, and only needs one cable to transmit all the necessary information. There are even companies that offer wireless transmitters for MIDI information.

I recently had the opportunity to try out three hand-held MIDI keyboard controllers, and was pretty taken with all of them . . . or maybe it's really the idea with which I am taken. The fact that I'm basically a guitar player who has taught himself to play keyboards probably has something to do with it, but that doesn't mean that I want the controller to be a guitar. I think the hand-held keyboard controller will become a valid instrument in its own right, and not just something that will enable keyboardists to "jump around like lead guitarists." To give you a better feel for what's available, let's look at some typical products.

YAMAHA KX1 AND KX5

The KX1 is the larger and more expensive of the two, and strangely enough, my least favorite. There is no way to change MIDI channels with the KX1, so you can't switch between different synthesizers by just using the controller. The KX5 at least has a switch that can switch between MIDI channels 1 and 2. One thing I have to say in favor of the KX1, though, is that the keyboard itself had the nicest "feel" of any of the controllers I tried. The velocity response was very predictable and the keys had a proper weight to them (unfortunately this is probably what makes the thing weigh 13 pounds). Both the KX1 and KX5 allow you to select 32 different patches via MIDI (which just happens to

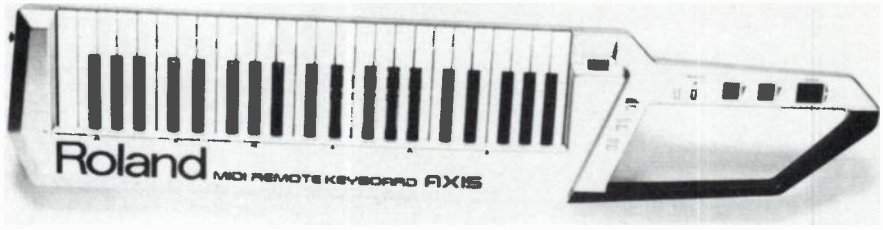
be the number of different patches available on a DX7).

The KX5's design is rather novel in that it uses "stubby" keys for its keyboard. The keys are as wide as normal piano keys, but not nearly as long. This allows the instrument to be made much thinner overall, which makes it easier to handle than the larger KX1. The feel of the keys is a little different than regular sized ones, but it didn't take long to get used to them. Another good feature about the KX5 is that it is very lightweight. The trade-off is that the keys are not weighted, so they don't feel as substantial as the KX1's keys.

The pitch bender on both Yamaha keyboards is a ribbon controller similar to the one found on the Micromoog and Polymoog. While I really liked this type of controller on the Micromoog, I found it rather difficult to handle on the portable keyboards. I guess I could get used to it if I practiced enough, but I found it difficult to hold on to the neck and bend the pitch at the same time; I had to support the keyboard with my right hand while doing pitch bending and vibrato with my left. I do vibrato with the pitch bender because using LFO vibrato sounds corny to my ears, and I like the subtle nuances that can be included in a carefully controlled vibrato. Jan Hammer was the first synthesist I remember seeing use the pitch wheel to get vibrato, and I like his vibrato very much.

THE ROLAND AXIS

The Roland "Axis" controller is my favorite of the three controllers I tried. It has an interesting shape, is lightweight, and offers programmability of certain functions. The programmability feature is both my favorite and least favorite aspect of this controller. I like it because it allows you to change the function of four switches and two wheels to any MIDI command you prefer, instead of having to settle for what Roland thought was most important. I don't like it because it is complicated to program these changes, and anyone who has read my articles for a while knows that I am against products that force musicians into becoming computer programmers. However, you *can* just use the Axis as it comes out of the box and never change any of the programmable aspects at all. In fact, the functions have been pretty well thought out and I think most people will probably just go with what is already there.



Roland AXIS controller

The Axis' pitch bender was the most comfortable of any I have played. The action on this bender feels the most like the movement that is required for bending on a guitar, so I was able to use it effectively without any practice at all. I was also able to use the bender while still supporting the instrument with my right hand (again, like a guitar) with no trouble at all.

The keyboard of the Axis felt pretty good; not as good as the Yamaha KX1, but better than the KX5. The keys are full size and mono aftertouch is included (as it is on all three keyboards I tried). The keys on the Axis also serve as switches for patch changes and other programmable features (it can switch between all 16 MIDI channels). When you hold down the Patch Change button and press a key at the same time, the patch change information is transmitted.

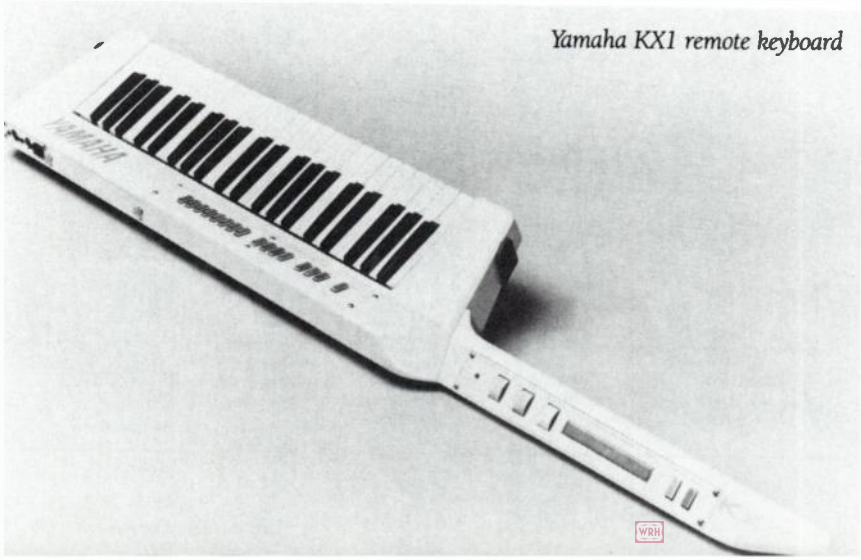
There is a useful two digit, seven-segment LED display that indicates which patch you are using at the time although the numbers can be a bit confusing. The first LED digit indicates one of four banks, as selected with the black keys; the second digit indicates an individual patch in that bank, as selected by pressing the appropriate white key. Once you get used to this numbering scheme there is no problem figuring out which patch you are using, and the method of operation allows you to change patches without having to take your hands off of the keyboard.

One other point that deserves mention is that the Roland Axis is AC powered while the Yamaha keyboards are battery powered. If you think you may someday want to use a cordless setup, you will probably want a battery-powered keyboard. The Axis comes with a heavy duty power supply in a footswitch enclosure. A standard MIDI cable connects your synth setup to the power supply, which serves as the Axis' "base unit"; a five pin Cannon connector connects the Axis itself to the power supply. A long (25 feet or so) cable comes with the Axis controller, and the Cannon connector offers the additional advantage of a snap lock feature. This could prevent substantial embarrassment, as it is not that hard to accidentally unplug a standard MIDI five pin DIN connector.

RECOMMENDATIONS

If the feel of the keys is your only consideration, check out the Yamaha KX1. At $1,295, it is not cheap, but it may be just what you want. If you don't want to spend very much money on a remote keyboard, then at $495 the KX5 is a good choice. If you are willing to spend a little more money and don't particularly care whether the keyboard is battery powered or not, take a look at the Roland Axis—at $695 it is a very versatile controller.

In any case, though, a remote keyboard gives a sense of freedom which at first might be intimidating but can end up being exhilarating. And if nothing else, you'll add that all-important visual element lacking from standard, stationary keyboard setups. EM



Yamaha KX1 remote keyboard

CHRISTINE McVIE ON FOSTEX

Christine is a singer, songwriter and, of course, a member of Fleetwood Mac.

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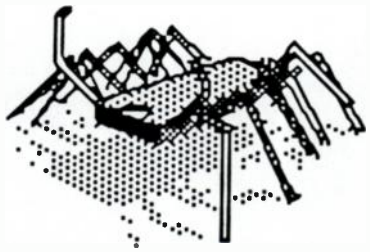
Getting intimate with a computer isn't the easiest thing in the world, but it's not the hardest either. Find out how a premier synthesist went from writing music to writing machine code—and back again.

Attack of the Microprocessors



BY LARRY FAST

I have always been hooked on gadgets—the more useful, the better. Electronic music is the ultimate territory for the gadgeteer, and one of the best gadgets ever created has to be the microprocessor.

My quest for control over electronic sound has been paralleled by a quest for control over microprocessors so that I could get them to execute musical tasks. A lot of the "behind the scenes" electronic housekeeping that we take for granted in current MIDI systems was, only a few short years ago, a techno-territory where only the brave and/or foolish dared tread. I'm not sure which category I fall into, but I dashed headlong into the world of microcomputers almost as soon as they appeared at reasonable prices.

I did have some preparation for this new world, such as a few college courses in higher language computer programming. Naturally, first came "batch-processed Fortran." This required hours of punching cards for the simplest computational task, and then a stack of cards were handed into the druids that ran the IBM mainframes. After sufficient turnaround time, usually overnight, it was time to check the big grey racks where the output paperwork was placed to see if the job had run. Usually it didn't because of a misplaced comma, or a left out space... not exactly a hacker's dream.

Larry Fast is one of the most respected synthesists in music today. In addition to his highly acclaimed work both in the studio and on the road with Peter Gabriel, Larry has recorded seven albums under the Synergy name and done sessions for hit songs by such artists as Bonnie Tyler, Foreigner, and Hall & Oates. He also writes computer software and invents gizmos that make his equipment run smoother and more efficiently.

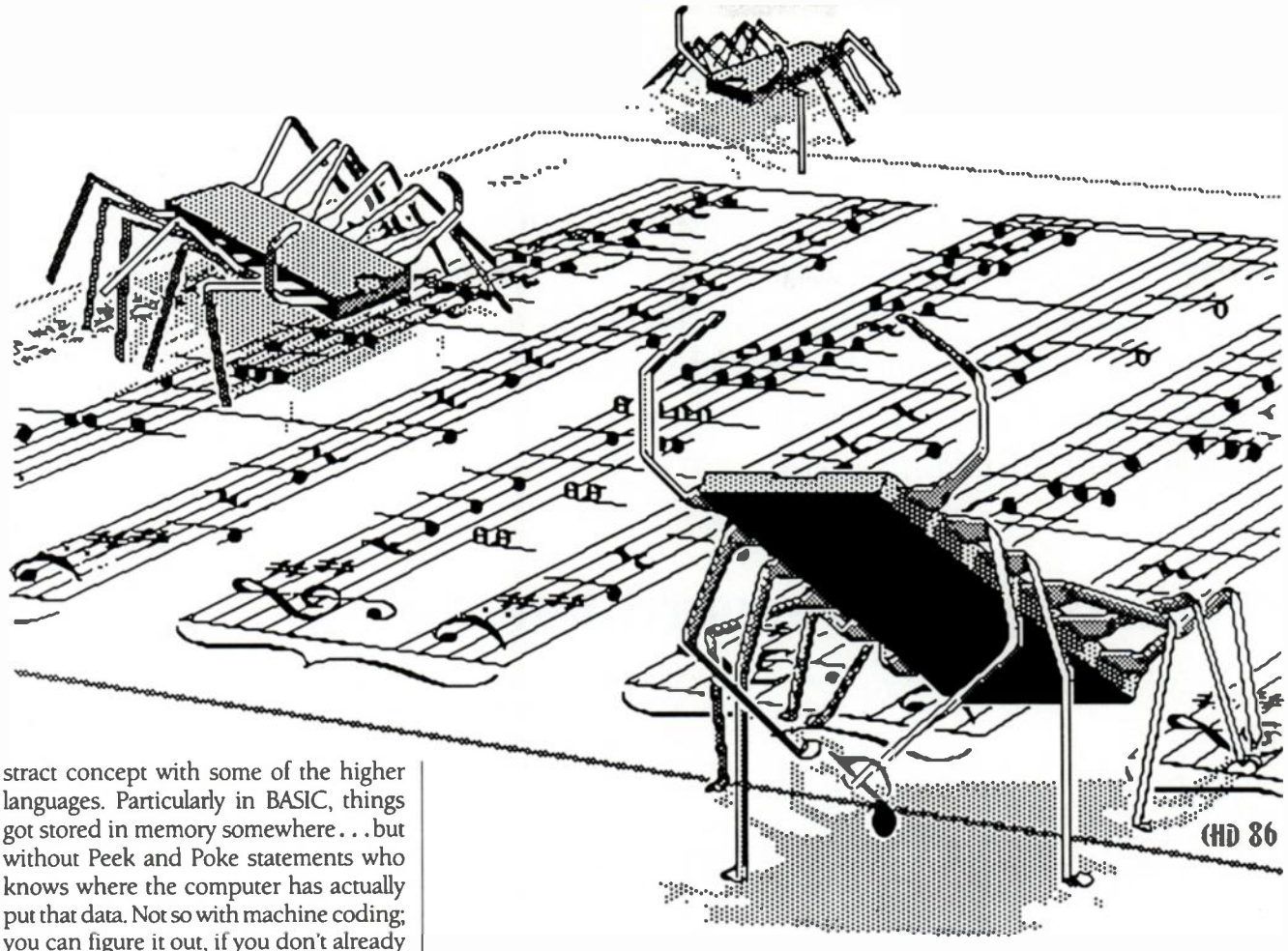
In 1972, the school installed a Hewlett-Packard 2100 minicomputer that ran an early, limited version of BASIC interactively to old-fashioned teletype terminals. It wasn't elegant, but the instant feedback from the machine hooked me immediately. My first all-nighters at the computer center happened then; I still have rolls of punched paper teletype tape (that was the popular storage medium then) with useless programs that I wrote at that time.

One of my biggest regrets at finishing college was losing free access to this great toy that answered back. So a few years later when the first single-board microprocessors were announced, I had to have one. I ended up with the KIM-1, a classic machine based on the 6502 microprocessor that had absolutely no frills at all—such as *higher languages* (BASIC, Pascal, Forth, etc.) that allowed for easy communication with the machine. This computer had to be talked to in the dreaded *machine language*. Machine language was one of those semi-religious things that only the high priests of computerdom could supposedly understand. In reality, it is a way of programming a computer that feeds binary or hexadecimal numbers to the computer. The computer directly uses this data to execute one of several dozen "tricks" that its designers have built into it. Writing machine language programs is the ability to string these binary numbered "tricks" from its repertoire into a useful collection of operations that do a required task—not unlike composition, where you create a complete song from a particular collection of notes and rests.

With this tiny machine, there was no other way to program except to learn the 6502's machine code and enter it in byte by byte. (Later 6500 family machines include the Apple II and Commodore computers, as well as the PAiA 8700 single-board machine which is similar in many ways to the KIM). I should also mention that the bare-bones KIM evaluation board didn't have anything elegant like a video screen or alphanumeric keyboard; instead, there was a six-digit LED readout and 24 key hexadecimal keypad. Most importantly, there were a couple of input/output data bus ports where a hardware hacker like me could hang the ADC and DACs (analog-to-digital and digital-to-analog converters) that could, in theory, control my analog synthesizers with programs for sequencers, arpeggiators and so on... if only I could learn to program in machine code.

Machine code turned out not to be so difficult after all. I learned the code specifically for the 6500 family of microprocessors, but the general concepts are remarkably similar for other microprocessor families like Z80s and 8085s, and even the more powerful 68000 family. Don't get me wrong, a lot of study time went into getting a handle on pushing the machine's most fundamental buttons, but none of the concepts to be mastered were anywhere near as abstract as some of the higher languages' more esoteric corners. The KIM had come with some good documentation for the 6502 microprocessor. With a slow, careful read through of that material and meticulous step-by-step experimentation to see exactly what action each of the 50-odd hexadecimal instructions would cause, within a few weeks I had enough of a feel for the system to begin stringing together short programs. These weren't anything more than a series of instructions to light an LED at predetermined intervals, or read the keypad and execute various useless subroutines for different keypresses. But in time I found a familiarity with the programming, and a real feel for the architecture of the machine. It was like driving a sports car with a 5-speed shift instead of riding in a big old mushy power-everything gas guzzler.

A lot of the mysterious places within the computer castle became familiar haunts. For example, when working with BASIC or Fortran, the Accumulator was a mythic place buried within the heart of the computer; somewhere where something or other was done to some digital value—a very abstract concept. The Accumulator, when dealing with machine language, is a real place with real functions that operate on the data exactly as you, the programmer specify. No mystery, no myths. Memory was also a somewhat ab-



stract concept with some of the higher languages. Particularly in BASIC, things got stored in memory somewhere . . . but without Peek and Poke statements who knows where the computer has actually put that data. Not so with machine coding; you can figure it out, if you don't already know exactly where every last byte resides.

An example would be in writing a sequencer program. Even the most basic sequencer, MIDI or otherwise, has two essential sections. One is the part to "read in" the musical data (what the notes are, how long they last, and in some cases dynamic and touch information). The other half "reads out" that stored data and plays it back to the synthesizer. The note and timing information is usually stored somewhat sequentially in memory. Since the "read out" part of the program is generally easier to write, we need some way to check our partially written program without having yet constructed the "read in" module. By pre-loading known data into the memory slots that will be used by the completed program with appropriate data, the playback half of the program can be tested to make sure that all the bugs are out. The memory slots selected by the programmer are the only ones that will be used, and the "read in" module of the program will be written later to store information in these same slots. Knowing exactly where the data is, and the ability to "follow" it through the program's path, allows the machine language programmer to visualize and execute a program with a precision unavailable with higher languages.

That kind of precision is what makes machine code so powerful. It is very efficient and allows the program to be written with no wasted effort in carrying out the more generic subroutines that the higher languages all must use. For instance, a "Poke" statement in BASIC should simply place a known value into memory, but a lot of time and computer energy is used in interpreting the BASIC syntax and converting it into a general purpose "store something into memory" subroutine. Many instruction cycles are used up, a lot of time is wasted, and in time-critical applications like sequencing, the programmer doesn't really know exactly how long the instruction might take to execute.

Machine code lets you bypass all of that and simply grab a value and stuff it into memory. It usually only takes two instructions and will always take the same number of machine cycles (the fundamental unit of time required to implement instructions), which means that the programmer can easily calculate—with microsecond precision—how long it will take to execute a subroutine. It is both quick and precise, which is essential in both synthesizer control and digital audio.

There are helpful devices for dealing with machine code such as the better *monitor* programs (i.e. system control programs that take care of housekeeping in even the smallest single-board development systems). These let you examine memory and machine registers (special memory locations inside the microprocessor), and single step through long programs so that programming errors can be caught as they crawl by in slow motion. Larger machines with CRTs and alphanumeric keyboards can often run *assemblers*. These special programs allow the programmer to structure machine code using slightly easier-to-use English-like notations on screen, which can be easily edited (to insert a forgotten instruction, for example). The computer then does the "drudge work" necessary to translate these English-like *Imnemonics* into the associated binary numbers that the machine program requires.

Once the initial fear of machine language has been overcome and control of the computer really understood, your outlook on all other ways of programming will probably change. There are many situations where using higher level languages still makes a lot of sense, but even there you will find your approach to programming more refined. And as a side benefit you will find a little more insight into why the machine language programs that reside in ROM and run commercially available instruments function the way they do (and you may have a bit more sympathy for any shortcomings you might have once intolerantly blamed on the manufacturer). Happy programming—now get back to the music! EM

Howard Jones has come a long way since he told his factory co-workers that one day he was going to quit his job and just play his music. From playing to five people in a pub only a few years ago, to headlining—and filling—major arenas, here's an electronic musician who turned his dream into action.

Keeping up with Howard Jones

BY CRAIG ANDERTON

Howard Jones broke on to the American music scene with a catchy, uplifting tune entitled "New Song." His first album, *Human's Lib*, was supported by extensive touring with Howard putting on essentially a one-man show that featured a battery of synchronized and synthesized instruments. The novel stage approach, coupled with Howard's ear for what makes a good tune, promptly garnered him both airplay and an ever-increasing number of listeners. His second album, *Dream Into Action*, was, if anything, stronger than the first and spawned a huge hit, "Things Will Only Get Better." Concert appearances in support of the second album, with Jones augmenting his lineup with Trevor Morais on drums (more about him later) and brother Martin on bass, have been receiving a tumultuous response just about everywhere he plays.

Nice guys don't have to finish last; Howard Jones proves that, and he was more than willing to share his ideas, as well as some practical advice, with the readers of *EM*.

EM: Since we set up this interview via electronic mail, let's start there. How do you use telecommunications?
HJ: We mostly use it to keep in touch with each other on tour. David (Stopps), my manager, is often off in a different city; and our production manager is in constant touch with the upcoming venues. I use it to keep in touch with people at home, and the fan club. As time goes by, we use it more and more. I think electronic mail is a great thing—it's a communications system where you can reach anywhere in the world instantly.

"I love turning people's preconceptions around"

EM: I plan to ask a lot about the live performance aspect of your show since live performance is the focus of this month's issue. However, it's clear that the whole underpinning to what you do is your songs, and I'm curious how you keep your songwriting chops up while you're on the road.
HJ: I think that if you are a songwriter, you need to constantly practice writing songs. I have an Akai MG1212 (12-track) set up in my dressing room, along with an Emulator II, DX7, Linn 9000, rackmount Super Jupiter, and some Roland effects. So I do demos everywhere I go—which has worked really well.

EM: Do you find that the instruments you work with ever influence the kinds of songs you write?
HJ: Yes, I always find that when I use a new piece of equipment it always inspires some kind of new thing...probably because one gets excited about the new sound.

EM: Do you think that these new tools could be dangerous, in the sense that they overwhelm you and take you away from the song and the music?
HJ: Well, I've always put 80 percent of my time into actual songwriting; about 20 percent goes into exploring new instruments and how to dress up the sound. I never get too technically bogged down in things...that does take you away from songwriting, which is the core of what I do. As one example, I've never bothered to work out how you program a DX7—I just couldn't afford to invest that amount of time, because it would have taken away from time I could have been writing.

EM: Besides, there are plenty of other people working on new DX7 patches, but there are no other people who are going to write your songs for you...
HJ: That's right. You can always obtain patches from the people who love to do that sort of thing.

EM: What kind of experiences outside the world of music shape your songs? What makes something click in your brain and say "I've got to write a song about this"?
HJ: (Long pause; laugh)...I don't know ...(pause)...it's really just the kind of things I think about. Just like anyone, whatever they do, they have thoughts as a result of the things that happen to them. That's all there is to it; there's nothing complicated about it at all. Sometimes it's triggered by feelings of anger, or sadness, or just because you feel very good sometimes so you write an exuberant song.

EM: Because of the importance your music places on the lyrics, you need a backup band that's totally sympathetic to the song. I was wondering if that was one of the reasons why you did the one-man band thing for some time.
HJ: It was partly that, and it was partly the *idea* of the one-man band itself. I wanted to show it could be done in a very human way. It was a real challenge for me to try and make it work. Besides, there just wasn't anybody around at the time that I particularly wanted to play with on a musical level. So, everything led me to work on the one-man thing.

EM: About the challenge aspect...why did you feel it was necessary to take up that challenge?
HJ: One was that I felt the idea of the normal rock and roll band had become such a cliché—everyone's done it for years, and also, it has been done so well. It's been brought to the peak of perfec-

Years ago, before editing EM, *Craig Anderton was synthesist for the Group Motion Berlin Dance company and did session work in New York. Presently, he plays guitar and keyboards in the performance group* Transmitter *and is working on an album project with pianist Spencer Brewer.*

PHOTO: JOHN BELLISSIMO

tion, really. I thought it would be great to do something radically different, but make it work on a popular level, without being involved in a kind of "fringe" thing. . . to show it could be done almost like in the mainstream of pop music. That was the challenge, and it was difficult, but I'm really glad I did it. I learned so much.

EM: Did it also provide more intimacy with the audience?
HJ: Yes, it did, and it was during that time that I really learned how to communicate with an audience—I had to, or I would lose the gig. All the stuff I learned doing the one-man band has been so valuable to me now that I have a band. It has helped me keep that intimacy, even though there are now other people on the stage.

EM: Based on the concert I saw, I think you definitely pulled it off. I think everybody had a feeling of personal attention. . .
HJ: It's very much my intention to do that. I always want to reach the people in the back row.

EM: Would you recommend that musicians try a solo act?
HJ: I would recommend it if you're the type of person who is perhaps a little bit lacking in confidence and also in ability to take on an audience. I learned to lose my fear of an audience, to completely relax. I think one of the most enjoyable things about going to a live concert is feeling that you get to know the performer much better. It's something you can't get from a record or video; it's a unique thing that should be exploited to the fullest.

EM: What do you think you've sacrificed, or gained, by using other players? What are the tradeoffs?
HJ: I don't really feel that I've lost anything. I was worried I'd lose the intimacy, but that hasn't happened. It's mainly gains, I suppose, in that I can devote more time to the audience; I don't have to worry so much about controlling the machines. And I can concentrate more on playing—more solos and stuff.

EM: Speaking of solos, that guitar thing you do towards the end of the set is really excellent (this is where Howard straps on his KX5 and plays the Emulator II electric guitar disk; the effect is not unlike Jeff Beck meets Godzilla). I've seen a lot of synthesizer players who are reasonably good at playing guitar leads, but you're the first keyboard player I've seen who can play keyboard like Pete Townshend or something. . . power chords, feedback, the whole shot.
HJ: (Laughs) To me, it's just so much fun to do it. (Laughs) I love turning people's preconceptions around, really, and this is an ideal way for me to do that musically.

EM: But you really do it well; I feel you've captured the *attitude* of the guitar. Did you play guitar at one time?
HJ: When I was just starting off playing rock and roll piano, at about age 13 or 14, I was playing the piano but I was always imagining that I was playing the guitar. I would try to voice the piano like a guitar, such as playing chords without thirds—the classic guitar sound, really. So when I discovered this Emulator disk with the guitar on it, you couldn't keep me away from it—I'd just be playing for hours and hours. Finally, I was getting out all the sounds I was imagining in my head.

On this tour, there would always be these kinds of macho security guys at the front, always with their back to you because they're watching the audience. And

I noticed that every time I started out with that sound, they would turn around because they couldn't believe those sounds were coming from a keyboard. I love the whole concept because it really smashes a few preconceptions.

EM: You look like you're having a good time too...
HJ: Well, I'm going to be playing it with my teeth next (laughs).

EM: Getting back to reality here, I compared notes with a friend who had seen a different concert and there seems to have been several differences. How much of the show is sequenced, and how do you allow room for improvisation when you're working with machines?
HJ: Right. Well...contrary to what people think, there is actually only one sequenced line playing in most of the songs, which is the MSQ sequencer playing two DX7s stacked together for more texture. With a few exceptions, the lines have been doctored in such a way you couldn't physically play it with your hands... which I think is the ideal way of using machines: to add a new dimension, or to play something very repetitive that you would never insult a keyboard player by asking him to play it. For example, in "Things Will Only Get Better" the sequencer is playing the sort of guitarish-type part. The song is structured until we reach the end, and at any time during that section, I switch off the sequencer and we go free. From then on, anything can happen. I use that structure in most of the songs, where we have room at the end to play around. People think there's a lot of sequencing, but there really isn't...it's down to the arrangement that makes the music sound full.

EM: I also noticed that your live sound isn't particularly cluttered, but it has a fair amount of power because there is *not* a lot of stuff going on...
HJ: I very much believe in that for live work. You can fit loads of things onto a record, but in a large, boomy, reverbery hall, the more clarity you have and the more concise your arrangements, the more enjoyable it is because you can hear every little detail that's going on. We also give the vocals top priority.

EM: How do you like working with the wireless mic?
HJ: I could never go back to an ordinary mic. Wireless gives me such freedom, although we do have a problem with the monitors since that mic is live all the time. I think we get a pretty good sound with it; but then again we've done a lot of experimentation.

"People think there's a lot of sequencing, but there really isn't...it's down to the arrangement that makes the music sound full"

EM: What is your current stage setup?
HJ: Let's start with the rhythmic side of it. The MSQ is the essential brain of the thing; it drives the Simmons SDS6 to give Trevor a click track, and plays the two DX7s as we mentioned before. My keyboard setup is Jupiter 8 on top, a Juno 60 in the middle, and a Pro 1. We just got a memory facility built into it specially to do little sequences. I use that a lot in "New Song" and "Things Will Only Get Better" since they both have short repeating sequences. On my left, there's a Prophet T8—I don't use the sounds, I just use it because I think it has a lovely keyboard. It can be routed to anything; backstage I've got an Emulator II, the two DX7s, and eight TX modules; the T8 can be routed to those or play the Juno 60. I've also got the Drumulator which I use for "New Song" and "What Is Love." Finally, I've got a CP80 piano out front.

It's going to be very different next year, 'cause I'm going to look at everything from scratch again...that was really this year's setup. I'd like to make things more modular, so that there are only one or two keyboards and the rest is more modular stuff, because it's easier to cart around and not as bulky.

EM: Trevor Morais is a remarkable drummer (Trevor plays standing up in a sort of "cage," whose framework holds what appears to be a near-infinite number of Simmons drum pads). Are the drum parts mostly programmed, or is Trevor playing all the parts?
HJ: In "Life in One Day" there's some extra bits of percussion programmed on the SDS6, which runs off the MSQ. On "Hunt the Self," the last song we do, the bass drum and snare are programmed and Trevor plays on top of that. But aside from that, Trevor is doing all the playing.

EM: I was real impressed by Trevor's setup. Who came up with that concept?
HJ: Trevor came up with the idea of a stand-up kit several years ago, and he used to have a bass drum and things hanging all around him. When Simmons came out, he had a combination of Simmons and real drums, all on this kind of rack system. So I thought well, I've never seen a drummer play like that before, and it really does shake some preconceived ideas about drumming. We decided to develop it and really go to town.

Often there's the comment that Trevor is just miming because people don't believe you can play drums like that, but he's playing everything. And it's very difficult, it's an absolute feat of coordination what he does.

EM: It really amazed me...
HJ: We're thinking the next stage for Trevor's setup is to use MIDI to trigger some sampling machines such as the Emulator...we're just exploring that now, as well as getting better quality samples.

EM: In the studio, do you play mostly live or do you do a lot of overdubs?
HJ: The way I did the last album was that often three-quarters of the track would be worked out before any tape was run; I had programmed and sequenced everything, and had done all the editing before taping. After that was down on the tape, though, there would be lots of overdubs and vocals ...it was basically a combination.

EM: So you approach the studio and concert quite differently; yet the overall sound is consistent. I don't think people would be disappointed seeing you live after hearing the record.
HJ: I do approach it differently, although I try and be as authentic as possible with the actual sounds. I think that things like reverb perspective should be accurate. I take a lot (of parts) out for live performance, because you can't fit all that in—you just end up cluttering the sound. There's nothing worse than a big blob of sound. It often takes a lot more nerve to strip things down—it gives you confidence to have tons of things all playing behind you. But I think it's much more effective if you make sure that the song is very well arranged, and that everything has its place in the sound spectrum.

EM: I noticed that you use the word confidence a lot; you certainly seemed confident during your concert. Was it always that way?
HJ: No, I used to be a wreck, a heap of jelly when I came off stage...and that wasn't so long ago. I couldn't look the audience in the eye, I couldn't talk, I'd

just be gibbering. I probably put more work into all that than into the music in a way. Overcoming the psychological problems and fear of being onstage was my biggest challenge. And that's what I've constantly been working on.

EM: What kind of directions do you want to pursue in the future?
HJ: I'm very much into songs and classic song formats; I really do love working in that way. I suppose my thing is to work with classic song shapes, but use them with exciting new sounds.

EM: What music do you like listening to?
HJ: Recently I've been listening to the Scritti Politti album, which I love, China Crisis, and the new Grace Jones album. I like music that's got energy in it. I don't listen to music for relaxation purposes; I can't get into music that's too laid back. I like things that are intense, even if they're slow tempo. I want to feel something very strong emotionally.

EM: Who do you look up to...do you have any heroes?
HJ: Well...for songwriters, I've always sort of admired McCartney. He's such an incredible melodic writer. And there have been two sort of genius producers that pop music has produced: George Martin and Trevor Horn. I think Trevor is a visionary producer, I really do. I wouldn't call these people heroes, though—I don't believe in having heroes. I think that can almost be a negative thing, to have a hero; it's almost too comfortable having heroes, because it takes the onus off you to do something. So I think heroes are a slightly dangerous concept (laughs). But there *are* people I really admire.

"...I used to be a wreck, a heap of jelly when I came off stage...and that wasn't so long ago. I couldn't look the audience in the eye, I couldn't talk, I'd just be gibbering"

EM: Would you like to elaborate on what you're trying to get across in your songs? For example, I recall reading somewhere you were involved in charity work, and I wondered if your general attitude was connected with that at all.
HJ: Well, I do make sure that every year I devote a decent amount of time to raising money for causes I believe in. I feel that's important for me personally to do that. But the general subject matter of the songs is trying to promote an attitude of questioning and thinking about everything, and not just accepting the status quo. And in having faith in idealism; I'm trying to promote idealism, because in a slightly cynical world where ideals are being trodden on and forgotten, I think it's absolutely vital to have our ideals and dreams. I do talk about that a lot, because I see ideals as such a motivating force for everyone. Even if they're not realized, they're practical things to have around (laughs). It almost acknowledges a faith in the future, a faith that things can change and can get better, and that the world can improve.

EM: Was there ever a time when you felt like giving it all up, and found those dreams kept you going?
HJ: Yes, oh yes, definitely. Before I got a record deal I would get desperate. I really thought that I was doing the wrong thing and that I really didn't have anything to offer musically at all, because I'd been

rejected so much. I couldn't even get gigs at the London clubs as a support act. There were a lot of times I felt like packing it all in and getting an office job. But I suppose I'd always wake up the next morning and feel a spark of optimism that I could do it, and that was enough to keep me going.

Another way I look at it is that even if I never did get a record deal, or actually managed to do what I'm doing now, I still think that I would have enjoyed trying to get there...knowing that I'd had a go and I hadn't given up.

EM: Otherwise you would always ask yourself, "well maybe if I had tried..."
HJ: Yes, exactly. I never wanted to get old and be full of regrets about things. But I think also one has to be practical. Sometimes you can work at a certain thing for years and years and it doesn't seem to get anywhere. Then I think that you have to sometimes say, well maybe this isn't for me and I should try something else...but still maintain the same kind of attitude that kept you going in the first place.

EM: I noticed you even involved some other people in your dream, like when you brought up two strangers from the audience during "New Song" to play bass behind your leads...you don't do that every show, do you?
HJ: No, just when I sort of feel like it.

EM: There was something about that that seemed to make a particular point. With so many people getting into little Casio instruments and such, do you think more people will be able to appreciate and get involved with music than ever before?
HJ: I think that because people—people who wouldn't have normally spent six or seven years learning how to play an instrument—can mess around with music in their own living room, that music will become more accessible. I think that's one thing synthesizers have done; someone who would normally never even dabble in music can enjoy it, you know, in a very simple way and...I don't know, that can only be good.

EM: That's what I think. Some people feel it trivializes music, but I just don't think that's possible.
HJ: (emphasis) It's about time music was de-mystified! I think musicians often play the role of god too much. I think the reason for that is that music itself is so powerful, and the musician is lucky to be able to tap into this kind of—I don't know, *force*—it can almost hypnotize people in a way. I think musicians often tend to take personal credit for something they're just tapping into.

EM: Well, I know we could go on a lot longer, but I want to keep this to a manageable length. I must admit, when it comes to interviews, I always feel somewhat uncomfortable just barging into someone's life and asking a bunch of questions. I'd like to make sure you've been able to say what's on your mind...is there anything you'd like to be asked that you haven't been asked before?
HJ: No, I'm quite happy with everything you've asked me and I don't think interviews are a drag. Sure, sometimes you get interviews where the person has already made up their mind about you, and there's nothing you can do to change their minds. But I like being on the sort of cutting edge of things, and I'm quite prepared to take this on. That's one reason I like doing interviews like this: between us, we can let people understand a little bit more what this is all about. EM

Now you can record with quality approaching that of a CD for about the same price as a quality reel-to-reel mastering deck. Sooner or later, you're going to be using a digital recorder—here's the info you need to understand what's going on.

Particularly Clean Music: The PCM Story

BY FREFF

(In this month's installment, Freff looks at the history of digital recording and why PCM recording does what it does so well. In the next installment, we'll get the lowdown on PCM adapters—what's available, and how to use them.)

Good gosh, look at the clock on the studio wall. Know what time it is? That's right, cyber-kids...it's time for another *Great Moment In Audio History You Most Likely Never Heard Of!* (cue wild applause):

You are at the NHK Research Institute of Japan. In front of you, on a wheeled cart, a frame holds the prototype of the first stereo digital tape recorder. It uses a technique called PCM, short for Pulse Code Modulation, and a clunkier-looking beast would be tough to find. It has three big rack-mounted metal boxes, lots of knobs, meters, switches, and pull-out circuit cards, a pro-class VCR the size of a pirate's chest, and so many cables you just *know* they'll be airbrushed out of the publicity photos. Though messy, everything in this hodge-podge is doing a specific and necessary job. The three boxes convert incoming analog audio signals into the 0s and 1s of digital data for recording, then reverse the process on playback. The VCR does the actual recording, its one-inch videotape storing the digital data in a format similar—though not identical—to a television picture. As for the crazy cabling, it connects everything together and gives the technicians something to fuss over.

This prototype is a monster; the parts alone cost $40,000. A commercial model would run $300,000 or higher, assuming anyone would be crazy enough to buy a 2-track machine that A) can't be edited and B) sounds only marginally better, if that, than a standard analog deck.

Price, however, is not important here. What's important is that the prototype works, publicly demonstrating the viability of digital recording and justifying further research. You and the attending audience of businessmen, audio engineers, acoustic scientists, and circuit designers are delighted. Applause all around, and everybody back to work.

Which is exactly how it was...in May, 1967!

Right. Nineteen years ago. Before the Summer Of Love, before Woodstock, before Gerry Dorsey changed his name to Englebert Humperdinck, exactly four months after the first Super Bowl (and just about the same time Sweden changed from left-side to right-side driving), that's when PCM recording made its bow.

The roots go back even further. Digital recording has two fathers, both American (Claude Shannon and Harry Nyquist), and one mother: the phone company.

Back in the 1930s, Mama Bell faced an overwhelming technical problem—how do you audibly and understandably send somebody's voice down a phone line? We're not talking sending a power chord down ten feet of guitar cable, but whipping millions of human voices back and forth around the world. Analog transmission can hack it only up to a specific point, for the same two reasons that bedevil all communication (even that between your guitar and your amp): noise and bandwidth.

Noise is inherent in every known form of signal transmission. Ask why and you invite getting bogged down in some fairly esoteric physics, so instead let's fall back on folk wisdom. *You can't get something for nothing.* You can certainly try (that's the history of science), but you can't succeed; the best you can do is lower the cost. It's a matter of efficiency—nothing works at 100 percent. Looking specifically at the telephone problem, there's the air between your throat and the mouthpiece (which absorbs and scatters some of the energy, introducing random variations, i.e., noise); the carbon microphone in the mouthpiece (which converts changing air pressure to changing electrical energy, but not perfectly, introducing more noise); the handset wire (short, yes, but all wires have some resistance and the difference between the electrons going in and the electrons coming out is—you guessed it—noise); the phone electronics (ditto); the wires from phone to wall, wall to house main, main to trunk, trunk to switching station (ditto, ditto, ditto, ditto); and after that, who can say? Depending on where you call, your voice could be bouncing off a satellite, streaking beneath the sea, or coming back to you from Cleveland, at every connection losing energy and gaining noise. If they didn't reamplify the signal regularly you couldn't even call the next county—and as you know from direct musical experience, amplifiers aren't terribly discriminating about what they amplify, treating all inputs (signal *and* noise) as equal... not to mention introducing new distortion and noise all their own. Frankly, it's a miracle that phones work at all.

"...in (the) combination of sampling and binary coding, Ma Bell laid the bedrock on which all digital recording has been built"

As for "bandwidth," it means something subtly different when referring to a signal source than it does when referring to a transmission medium. For signals, it is essentially a measure of their maximum size, of how wide a range of vibration they take up. It represents their "space,"

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in much the same way that you are so many inches tall or a piece of music is so many bars long. The bandwidth of a transmission medium, conversely, is a measurement not of how big it is but of how big it *can* be.

The problem with bandwidth is that if your signal bandwidth is greater than the transmission medium bandwidth, you are in trouble. You can't send the whole signal; only part of it...and if enough gets lost in transmission, what's left may not be recognizable. (Can you identify that song in *only one note*, game show fans?)

So there's poor Ma Bell, desperation incarnate, struggling with rising tides of noise and an incredible bandwidth crunch. Though crisis city hadn't been reached in the '30s, it was clear that sooner or later analog methods would get to a point of diminishing return. So what do you do, if you're a multinational corporation with a ton of money and the sense to spend a good chunk of it on far-sighted research?

Answer #1: invent sampling. Pretty much as we know it today, in fact, and approximately on par with an Ensoniq Mirage set to its lowest bandwidth (actually, at 4 kHz the Mirage is better; the phone company stopped at a meager 3.3 kHz). Sampling finessed the bandwidth problem by reducing a voice to discretely spaced air pressure measurements, and then sending only those selected few down the line instead of the whole thing. The trick was to transmit as little information as possible and still get an understandable voice out at the other end. (Which analog experience has shown to be remarkably little, thanks to a useful quirk of human physiology: the ear will reconstruct fundamental tones when presented with the right set of upper harmonics. The phone company takes advantage of this and sends only those harmonics, nothing else, thus letting your brain do the heavy-duty processing. Remember that, next phone call. You only think you're hearing what you're hearing—literally!)

Answer #2: use binary coding. Before sending the selected pressure measurements, convert them into "words" made from strings of 0s and 1s (actually, voltage level pulses of specific lengths that are referred to, for convenience, by number). Binary coding offered distinct advantages, including working miracles in the fight against noise. A binary signal is either On or Off. There's nothing in between. This meant that Ma Bell could build some "smarts" into the system in the form of circuits that would ignore voltage fluctuations outside the patterns chosen to represent 0 and 1. In other words, with binary signals it took much more line noise to screw the message up in transit.

Right there, in that combination of sampling and binary coding, Ma Bell laid the bedrock on which all digital recording has been built.

Only...it's one thing to push 3 kHz down a line, to an ear that will do the rest of the work for you, so that you can be understood (if not necessarily believed) when you tell someone the check is in the mail. Capturing the full bandwidth and dynamic range of music is several quantum leaps beyond that, and for a long time neither theory nor hardware were up to the task.

Theory came first. It began with communications engineer Harry Nyquist, who proved mathematically that any sound could be perfectly reproduced, given a high enough sampling rate. Then, in 1948, a scientist at Bell Labs named Claude Shannon published two papers on something called "information theory." Summing up the importance of these papers is close to impossible (aside from mentioning that they have profoundly altered and influenced fields of science all the way from computers to astrophysics to genetic engineering); what matters to us is that they dealt, among other things, with how to encode and decode information in as compressed and error-free a fashion as the laws of physics will allow.

As for the hardware, well, that was just a matter of time—a time that the electronics revolution shortened beyond anyone's capacity to foresee. In 1967, the NHK PCM prototype (described earlier) turned digital recrding theory into engineering fact. Eleven years later, in 1978, the first professional 2-track PCM recorders went on sale. By 1983, as the Compact Disc entered the marketplace, PCM technology was firmly in place as the system to beat for digital multi-track recording and digital mastering. As for today, even a gypsy with a bowling ball for a crystal can see that the future of recorded sound—indeed, of recording itself, from multitrack right down to portable studio—is digital. Period. End of statement, end of argument.

As a matter of strict fact, some of it isn't the future at all. *Right now* you—yes, you!—can buy, for anywhere from $1,000 to $3,000, the hardware to do stereo digital mixdowns and recordings of CD quality.

Is it magic? Not a chance—simply the right combination of videotape, VCR, and one of several available "consumer" PCM encoders. The encoders are the key. Who makes them, how they differ, and how to get the most out of them will be covered next issue; for now, it's important to lay out a clear explanation of how they work.

The PCM encoder's first job is to send each channel of the incoming audio signal through a line amp designed to adjust signal level and impedance, while at the same time keeping frequency response and phase flat within all audible frequencies. This is nothing special. Most audio equipment does it.

Next comes something new, a "dither generator" circuit. Dither is white noise which is deliberately added to the input signal. As to why you want to go crudding up your input, consider this: When a sound is converted from analog to digital, as will happen a little further down the chain, it is "quantized"—that is, broken up into discrete steps. This automatically and unavoidably creates a series of measuring inaccuracies called *quantization noise*, the net difference between the original analog sound and the stairstep series of measurements it has been converted into. In the normal course of things, this noise is filtered out...so long as the signal level is high enough, that is. When level drops, the difference between q-noise and signal gets so low the filtering system can't tell what's what, and lets both through. Result: audible high-frequency distortion.

How Much Sampling Rate Is Enough?

Sampling rate is the biggest controversy going in digital audio. The argument is that the industry standard rate of 44.056 kHz is just not fast enough. At first glance it seems it ought to be. Even extraordinary human hearing doesn't pick up frequencies beyond 20 kHz, well within the Nyquist limit for the industry rate. Nevertheless, in ways we don't fully understand—but which may be related to that harmonics/fundamental trick telephone calls rely on—recent tests indicate that harmonics above 20 kHz do have a measurable impact on aspects of timbre and phase perception. If these test results hold up, recording and sampling rates will surely rise.

In one of life's bizarre ironies (like taking one poison to neutralize another) adding just the right amount of white noise in front of the conversion process creates a signal that, when converted, has enough q-noise in all frequencies so the filters can always detect and remove it, even at incredibly low signal levels.

A low-pass filter, located post-dither (but still pre-conversion), sharply limits any incoming frequencies higher than half the encoder's sampling rate. This helps control *aliasing noise*, which we'll cover in just a moment.

At last, prepped and processed, the analog audio signal enters the electronic Cuisinart: first a sample and hold circuit, where the sound is sampled thousands of times a second; and then an A/D converter, which gathers up those samples, looks at them, and spits out slices of digital data. The number of slices is the sampling rate. The slices themselves are in the form of 0s and 1s that have been bunched together in binary words. These two factors—sampling rate and word length—set the principal limits on PCM audio quality.

"At last, prepped and processed, the analog audio signal enters the electronic Cuisinart"

Consider sampling rate first. As Harry Nyquist proved, to accurately reproduce a waveform you have to sample it at least twice as fast as its highest frequency. Cut the rate, and frequencies start to fall between the cracks. This is not necessarily bad. If the low-pass filter has done its job, the result is no more than reduced bandwidth, with fewer high frequencies and therefore a less realistic sound. If the filter isn't adequate, however, the mathematical errors pop up as audible, out-of-pitch sounds which have come to be known, because they involve a case of mistaken identity, as aliasing noise.

Binary wordlength is important because it sets an upper limit on how accurate each sample can be. A word one "bit" long can describe only two different states, those being either 0 or 1. On or off. Two-bit words can describe four possible states—00, 01, 10, and 11. Each bit you add doubles the possible number of measurements a binary word can express. In today's PCM recording there are two different lengths in use, 14 bits (16,384 levels of measurement) and Sony's more demanding 16 bits (which increases measuring accuracy to 65,536 levels). As a general rule of thumb, each bit of word length buys you 6 dB of signal-to-noise ratio, with 16 bits providing a potential signal-to-noise rating of 96 dB! But (and here's another dose of irony) remember why we added dither? Right. Quantization noise. Though it seems you should be able to increase digital recording quality simply by increasing word length, it won't work. Instead, what happens if you start dealing in 18- and 20-bit words is that the steps between levels of measurement get so fine that q-noise can no longer be identified and separated from signal. Both get through, and the result is serious buzzout.

Conversion is only a portion of the game. Now the converted signal has to be recorded, which introduces a whole new string of problems and solutions.

First there's error correction to add to the signal (this is where Shannon's work comes into heavy play). In essence, error correction means duplicating the binary words of the digital audio signal several times, in one of a variety of complicated schemes, so that on playback the circuitry can look at, and compare, several different examples of the same signal. Why? Because there is a tradeoff in using binary coding. Yes, it is far less susceptible to distortion from noise. But if something *does* create an error, that error is absolute, total, and unpredictable. . . aside from the general fact that the longer the binary word being used, the more radically inaccurate an error can be. This is bad news, and according to Shannon the only fix is redundancy.

However, massive error correction solves one problem but slams us smack into another—bandwidth. To achieve enough redundancy means processing over *two million bits of information a second.* This is about as many bits as there are letters in the entire *Lord of The Rings* trilogy (plus *The Hobbit*). . . which is more than moderately mind-boggling. How do you store all of that? Standard tape recorders won't do—even multi-track machines, running at 30 ips, don't have enough bandwidth for the job.

". . . if something does create an error, that error is absolute, total, and unpredictable"

The answer is a VCR. Television signals also have a huge bandwidth. VCRs cope with this by using multiple recording heads positioned on a drum that rotates at an angle to the passing videotape. Taken together, head and tape motion give you the equivalent of 16 *meters* of tape per second. Which is bandwidth enough, to be sure. . . but since VCRs record signal at an angle across the width of the tape, instead of down its length, the digital audio signal must be broken into pieces exactly as long as the angled track, then compressed and time-shuffled so that nothing gets lost in the small but measurable gap between where one strip ends and the next begins. (This is called the *switchover time*, and you can see it as a black stripe that creeps into view on your TV screen whenever the vertical hold goes wonky). Lastly, a synchronization signal has to be added, so the encoder can correctly interpret the signal even if the VCR transport isn't stable in speed or vibration.

Playback isn't any easier technically than recording, but it's much easier to describe! First the sync signal is read and used as a guide; by its numbers the digital data is analyzed, expanded, pieced back together, and error-corrected; and finally the result goes to a D/A converter, which changes this data from digital back to analog form. From there, the signal passes through some circuitry designed to improve its frequency characteristics and a final low-pass filter that serves, primarily, as an extra ounce of prevention (high-frequency signals created by the D/A process should be well above hearing range, but might—just barely might—induce cross-modulation noise in amplifiers, speakers, or signal processors that aren't perfectly linear; safer to shoot down those frequencies here).

Those are the engineering basics. By any standard, PCM encoders are technological marvels. That you can pick them up with one hand, or buy one for well less than the cost of a DX7, just makes them all the niftier.

Frankly, no cyber-kid worth his or her salt should be without one because the *really* neat thing is not how they work, it's what they let you do. . . which is what we'll cover next time. EM

Emulator II. We set the standard. Now we've raised it.

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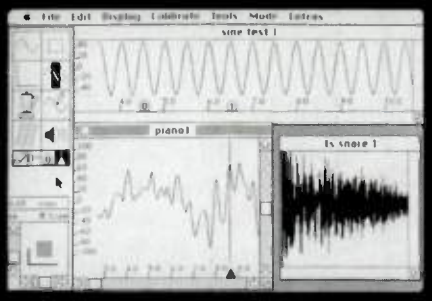

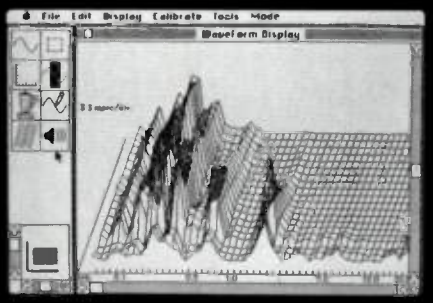

The Digidesign Sound Designer™ turns your Emulator II into a complete computer music system

with punch-in, punch out, and track bouncing. What's more, it can also function as a stand-alone multichannel MIDI sequencer capable of controlling up to eight other MIDI instruments and drum machines with a storage capacity of over 90,000 notes!

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Add to all this an RS-422 computer interface, a 24 pulse per quarter note clock, a full MIDI implementation, and a built-in SMPTE code reader/generator and you have a complete music and sound effects production facility in a single portable package.

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To find out more about the Emulator II (and there's a lot more to find out) visit your nearest E-mu Systems dealer for a complete demonstration. Or send us $2.00 for a demo record and color brochure.

The Emulator II. The new standard from E-mu Systems.

Delve into the world of after-envelopes, MIDI echo, synthesized delay, automatic double bends, FM, and much more as Alan shows how to squeeze every last bit of sound from the Casio CZs.

CZ Secrets

BY ALAN GARY CAMPBELL

The CZ-101 MIDI synth offers a lot for a little. If you haven't yet played one, or have only casually stepped through its factory presets, you're in for quite a surprise! Not only are the bass and lead sounds absolutely great, the CZ can sound "digital," "analog," or—better yet—"none of the above." It can do Minimoog-like "fat" patches, FM, MIDI delay effects, single/multi-trigger leads, envelope delay, and lots of other stuff it probably wasn't even *supposed* to do.

This article unveils CZ programming "secrets" and helpful hints (a follow-up article will cover CZ-101 modifications). Assuming you have a basic familiarity with the instrument, let's make some noise!

ENVELOPE GENERATORS

The CZ-101 envelope generators (EGs) are similiar to those found in competitive hybrid and digital synths (Yamaha DX-series, Korg Poly-800 and DW-series). The CZ envelope generators are exceptionally versatile, however, with eight envelope segments (instead of the usual four to six) along with the ability to insert arbitrary SUSTAIN and END points.

When programming the EGs, remember that each envelope rate value represents a fixed rate-of-change. That is, an ENVELOPE RATE of "50," starting from an ENVELOPE LEVEL of "00," will take considerably longer to reach an ENVELOPE LEVEL of "90" than an ENVELOPE LEVEL of "20" (**Fig. 1**). Therefore, changing an envelope level forces a change in the associated "attack," "decay," or "release" times (for lack of better terminology). To compensate, you'll have to adjust the appropriate envelope rates after changing an envelope level.

Inserting an END POINT automatically sets a level of "00," which prevents you from inadvertently setting up DCA envelopes that never turn off (although that *could* be useful). Also note that the DCW KEY FOLLOW function works oppositely to the way keyboard tracking affects a VCF: KEY FOLLOW values *greater than zero* make the timbre progressively *less bright* as you play higher on the keyboard.

***Alan Campbell** is the owner of Musitech™, an electronic music consulting firm. He is also a synthesist and technical writer.*

COMPARE/RECALL FUNCTION

When editing a patch, the COMPARE/RECALL function stores the altered version in a memory "workspace." The altered patch stays there until you edit another patch, even if the unit has been turned off. You can call up a patch and step through its parameters (to write out a patch sheet, for instance) without disturbing the COMPARE/RECALL memory, as long as you don't *edit* any parameters.

PITCH BEND/MODULATION

When "slaving" the CZ-101 via MIDI, the LFO does *respond* to modulation wheels and similar controllers on the master. However, you can't control the *amount* of modulation. As you engage the master's modulation controller, the LFO "jumps" on at some point to the full modulation depth set in the patch, and doesn't turn off until the modulation controller is returned to the "zero" position. Some modulation controllers (e.g. DX7) require excessive travel to activate the LFO, while others (e.g. Emulator II) engage the LFO with just the slightest "push." (A simple LFO switch, as incorporated in various Roland left-hand controllers, would probably be as effective as a wheel with the CZ-101.) Adding a small amount of LFO delay will "soften up" the abruptness with which the LFO turns on. Note that the CZ-5000's built-in mod wheel avoids these problems.

The spring-loaded pitch bend wheel of the CZ-101 doesn't have the best tactile feedback for those "Jan Hammer" bends, but it does lend itself to "flicking" with the thumb, to simulate an electric guitar's twang bar. You can modify the pitch bend wheel to remove the spring-loading and add a mechanical center detent; we'll show you how in the CZ modifications article.

RING/NOISE MOD, DISABLING DCOs

The CZ's digital "ring" modulation is actually only a software approximation of balanced modulation. Even so, it's a welcome addition to the timbral capabilities of most synths. One "classic" patch is to ring-modulate one oscillator with another, and listen to the ring mod output while sweeping the pitch of one of the oscillators with an envelope generator. Such effects are easy with the CZ-101: Line 1 ring-modulates Line 2′, and the DCO 2 ENV sweeps the pitch.

The NOISE MOD function tracks the keyboard with the perceived pitch center of Line 1′ or Line 2′. Thus, DCO envelopes, pitch bend, and modulation effects can control the "noise pitch." When Line 2′ is noise-modulated, DCA 2 can control the

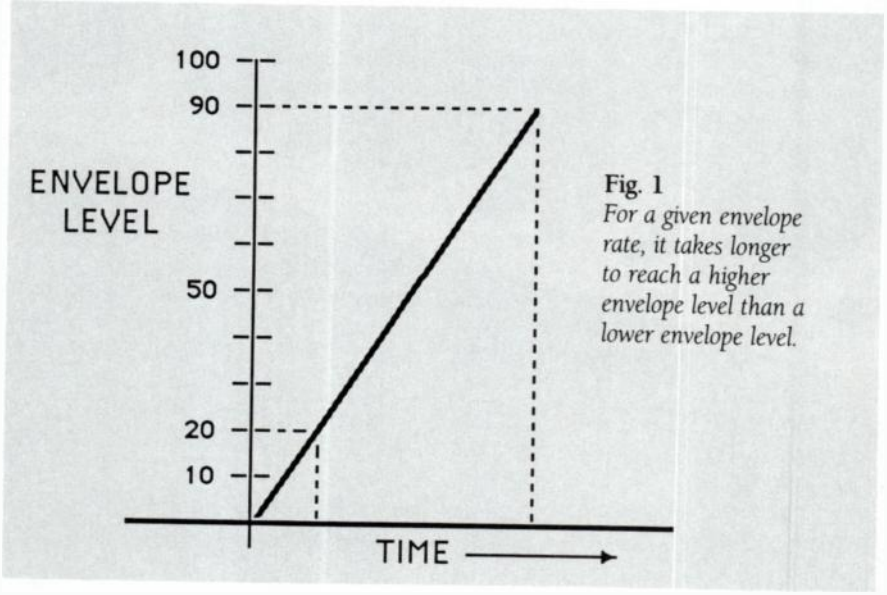

Fig. 1
For a given envelope rate, it takes longer to reach a higher envelope level than a lower envelope level.

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REQUIREMENTS: Commodore 64 or Apple IIe/+ Computer, One Disk Drive, One Monitor (color optional), One Syntech MIDI Standard Interface*

The Studio will operate on MIDI Interfaces from the following companies: Syntech, Yamaha, Korg, Passport, Mimetics, and Sequential 64 and 242.

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amplitude envelope of the noise modulation, but the DCW function has no effect.

Ring and noise modulation affect only Lines 1′ and 2′. But, Line 1′ or Line 2′ can only be selected in conjunction with Line 1. So, to hear the modulation effect by itself without also hearing Line 1 (unmodulated), you'll have to select Lines 1 + 1′ or 1 + 2′ then turn DCO 1 off. DCA 1 could be set to disable the DCO, but this doesn't work for the RING MOD function since DCA 1 controls both DCO 1's level and that of the RING MOD effect. The following DCA routine is useful for isolating the NOISE MOD of Line 2′, though::

DCA 1 ENV
Rate 1: 00
Level 1: 00 END

It really doesn't matter what values are assigned to Rate 1 and Level 1; when you place the endpoint in the first envelope segment, the DCA envelope just doesn't "happen," and the associated DCO is effectively off.

MIDI "ECHO" EFFECTS

Use a MIDI cable to connect the CZ's MIDI In and MIDI Out together. As you play, the keyboard control algorithm assigns a voice to each key; a moment later the MIDI control algorithm assigns a second voice to the same key. The resulting time delay produces a subtle "slapback" echo. You can hear a mild chorus- or flange-like effect (due to the phase difference) during the portions of the envelope where the harmonics are changing amplitude.

This subtle delay seems to produce just the right amount of timbral animation for certain patches. But you don't get something for nothing; this does cut the number of available voices in half. If you've selected LINES 1 + 2′ or 1 + 1′, this means you'll only have two voices—or does it? Sometimes you can get three notes to play: one assigned to the keyboard, one to the MIDI In, and one to both. It depends on the phase of the moon, or the color socks you're wearing, or something.

FAT PATCHES

The TONE MIX function normally "stacks" one patch on top of another, but it can be used to stack a patch *on top of itself*. The associated voice-assignment delay and phase offset produce a really fat sound that's great for lead and bass patches; it combines digital "punch" with analog fatness. With this technique, I find myself applying the CZ where I might have used a Minimoog. This is not to imply that the CZ-101 replaces the mini, but the Casio is certainly an excellent timbral resource with which to augment it.

SINGLE/MULTIPLE TRIGGER MODES

In SOLO or TONE MIX mode, the CZ-101 functions as a multiple-trigger, last-note priority, monophonic synth. For single-trigger control when the instrument is in SOLO or TONE MIX mode (to mimic Minimoog operation, for example), turn the Portamento on and set the Portamento Rate to "00." The CZ is now in single-trigger, last-note priority mode. To return to multi-trigger mode, just turn the portamento off.

ENVELOPE DELAY

At the lowest possible DCA Envelope level (01), the sound from the DCO is virtually inaudible. We can use this property to generate pseudo-envelope-delay:

DCA 1
KEY FOLLOW: 0
Rate 1: 50
Level 1: 01
Rate 2: 99
Level 2: 00
Rate 3: 90
Level 3: 99 SUS
Rate 4: 60
Level 4: 00 END

The sum of Rates 1 and 2 determines the total delay time. (In this specific example, Rate 1 alone determines the delay, since Rate 2 is set at its fastest possible rate). The remaining ENV stages make

Fig. 2

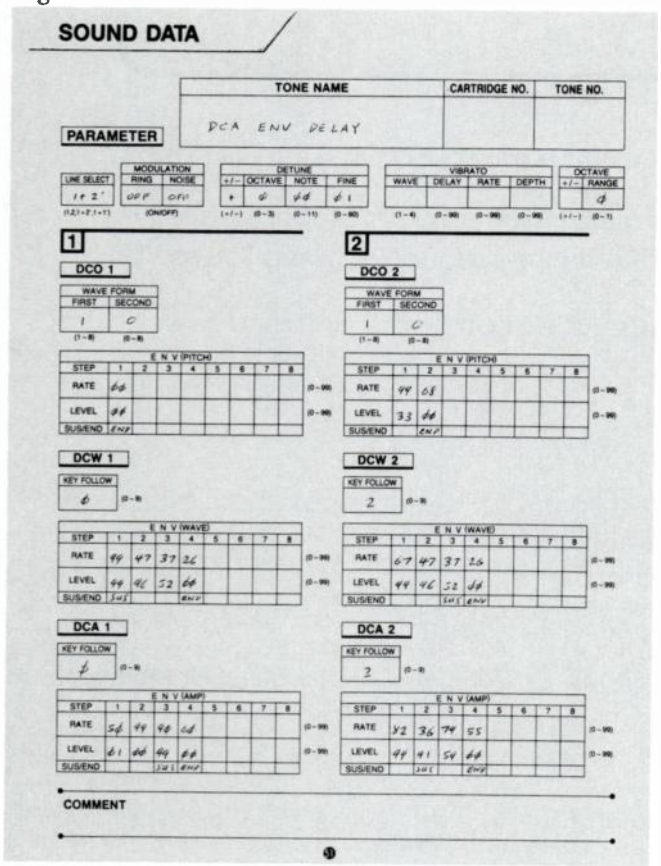
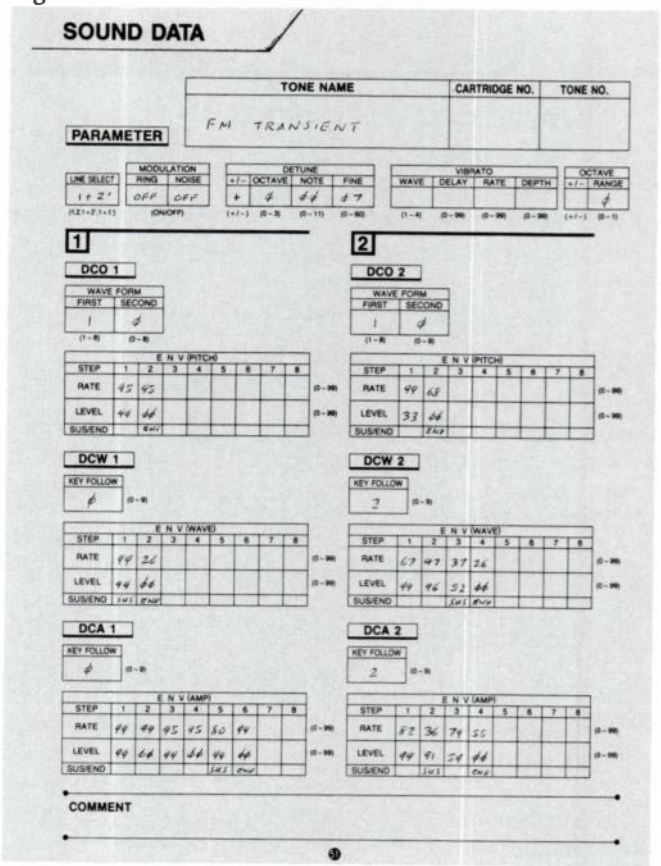


Fig. 3



up a standard ASR-type envelope: Rate 3 controls attack, Level 3 sets sustain, and Rate 4 controls release. In this example and those that follow, setting the KEY FOLLOW parameter to a value greater than zero will cause higher notes on the keyboard to have shorter envelope delays, as well as shorter envelopes.

Envelope delay in this range sounds a lot like slapback echo; the sample patch in **Fig. 2** demos this effect. For longer delay times, simply add more envelope delay stages:

DCA 1
KEY FOLLOW: 0
Rate 1: 50
Level 1: 01
Rate 2: 50
Level 2: 00
Rate 3: 50
Level 3: 01
Rate 4: 50
Level 4: 00
Rate 5: 90
Level 5: 99 SUS
Rate 6: 60
Level 6: 00 END

The sum of Rates 1, 2, 3, and 4 determines the total delay time (the settings in the above table are arbitrary). The remaining envelope stages (Rates/Levels 5 and 6) make up an ASR envelope, as in the previous example.

You can also generate delayed "after-envelopes":

DCA 1
KEY FOLLOW: 0
Rate 1: 99
Level 1: 01 SUS
Rate 2: 50
Level 2: 00
Rate 3: 50
Level 3: 01
Rate 4: 50
Level 4: 00
Rate 5: 65
Level 5: 99
Rate 6: 75
Level 6: 00 END

Upon releasing a key, an after-envelope occurs at a time determined by the sum of Rates 3, 4, and 5. This example creates an AD-type after-envelope, made up of envelope segments 5 and 6. Rate 5 controls the after-envelope attack, Level 5 sets the after-envelope peak, and Rate 6 controls the decay.

Pseudo-envelope-delay is a useful technique. The only trade-off is that it uses up envelope segments.

FM

The CZ-101 is basically a *subtractive* synthesis machine. But, at their maximum settings, the DCO EGs are so fast that they can directly generate FM transients—a useful technique for synthesizing brass and plucked- or struck-instrument timbres. For example:

DCO ENV
Rate 1: 99
Level 1: 99
Rate 2: 95
Level 2: 00 END

Rate 1 and Level 1 control the attack portion of the FM transient; Rate 2 and Level 2 control the decay. For this effect, useful DCO ENV values range from about 80-99 for Rate 1 and Rate 2, and 50-99 for Level 1. Normally, Level 2 remains set at 00. The asymmetry between the R1 and R2 values is intentional, but arbitrary. Try anything.

In this case, the ear perceives the FM transient as something "interesting" that happens during the attack, and it's not too concerned about the actual FM spectrum or the spectral envelope. For two excellent references on this type of synthesis technique, see Patrick Gleeson's column "Synthesizer Technique: Simulating Plucked Timbres" (*Contemporary Keyboard*, December, 1978, page 64), and Steve Porcaro's "Rock Technique: The Jacks on the Back" (*Keyboard*, February, 1983, page 65).

In the above example, and those that follow, the DCA ENVs must be set so that the DCOs are audible when the FM transient occurs. (Often, this requires a *fast* DCA attack, since the FM transient occurs immediately upon key-down at the fastest rate possible.) **Fig. 3**'s synth-brass patch demos the FM transient effect.

For a *delayed* FM transient, use the pseudo-envelope-delay techniques described in the previous section:

DCO ENV
Rate 1: 25
Level 1: 01
Rate 2: 99
Level 2: 00
Rate 3: 99
Level 3: 99
Rate 4: 95
Level 4: 00 END

Rates 1 and 2 control the overall delay; in this example, Rate 2 is set as fast as possible, thus Rate 1 controls the total delay. Rate 3 and Level 3 control the attack portion of the FM transient; Rate 4 and Level 4 control the decay.

For longer delay times, add more envelope segments. FM "after-blips" are possible, too. However, the envelope delay techniques described earlier are not as generally applicable to the DCO envelopes, since even a sustain level of "01" will produce a noticeable detuning effect.

Automatic Double Bends

This playing technique is normally associated with contemporary stringed instruments. A "double bend" involves playing a note on one of the higher strings of the instrument, while simultaneously playing a second note (usually a whole step lower than the first) on the next lower string. As soon as the two notes are plucked, the lower string is "bent" sharp to match the pitch of the higher string. The resulting sound is very dramatic and penetrating, generating intense "beating" (difference-frequency amplitude modulation) as the pitch of the lower string approaches that of the upper.

The DCO ENVELOPE GENERATOR can create a "double bend" effect. Select a brass or guitar-like patch with a relatively fast, simple attack; choose a patch with similar or identical timbres for lines 1 and 2. Turn off DCO 1 ENV, then set the DCO 2 ENV:

DCO 1 ENV
Rate 1: 00
Level 1: 00 END
DCO 2 ENV
Rate 1: 36
Level 1: 20 SUS
Rate 2: 00
Level 2: 00 END

Select lines 1 + 2′. Set the DETUNE function as follows:

DETUNE
Polarity: - (minus)
Oct : 00
Note : 02
Fine : 27

The DCO 2 ENV "warps" the pitch upward, at a rate set by Rate 1, to the sustain level; the detune function compensates for the sustain level pitch offset. (Alert readers may wonder why I didn't simply specify a lower DCO 2 ENV SUS level, a slower rate for DCO 2 ENV Rate 1, and less detuning. Well, it turns out that the perceived DCO ENV curve is not the same at low SUS levels. To my ear, the above routine simply sounds better.)

This technique requires DCA ENVs with relatively fast release times, since DCO 2, at least, needs to be inaudible during the release segment as the DCO 2 ENV decays to zero and grossly flattens the pitch. The ASR (attack, sustain, release)-type DCA ENV below is typical:

DCA ENV
Rate 1: 99
Level 1: 99 SUS
Rate 2: 85
Level 2: 00 END

Rate 1 controls the attack time, Level 1 sets the sustain level, and Rate 2 controls

the release time.

Fig. 4's patch demos the "Auto-Double-Bend" effect. The DCO 2 resonant waveform, swept by the DCW, makes the patch more interesting but obscures the double-bend effect somewhat. You might want to disable the resonant DCO waveform and select an alternate DCW ENV, as follows:

DCO 2 WAVEFORM
First: 1
Second: 0
DCW 2 ENV
KEY FOLLOW: 2
Rate 1: 99
Level 1: 99 SUS
Rate 2: 49
Level 2: 00 END

Applying similar pitch-warp envelopes to both DCOs provides an "auto-glide" or "auto-bend" function. The detune function can then be used normally.

Envelope-Controlled Chorusing

This effect uses one of the Pitch EGs to change the relative detuning of DCOs 1 and 2 over time. For this example, the DCO 1 ENV is off:

DCO 1 ENV
Rate 1: 00
Level 1: 00 END
DCO 2 ENV
Rate 1: 08
Level 1: 08
Rate 2: 08
Level 2: 00 END
DETUNE
Polarity: + (plus)
Oct : 00
Note : 00
Fine : 27

The sound starts with a given amount of negative detuning, and progresses through zero detuning on to a similar amount of positive detuning. During the "release" segment, the process is reversed. Since the detuning envelope incorporates a sustain level, staccato playing yields a relatively fixed chorus rate/depth, while legato playing emphasizes the envelope-controlled chorus effect.

For double-bends we wanted to make sure that there wasn't any sound during the release segment, but for ENV-controlled chorusing, we want to make sure that there *is*. Select a piano- or organ-like DCA envelope with a fairly long release time.

Fig. 5, an electric piano patch, demos the envelope-controlled chorus effect.

AC Adapters, Batteries

The recommended AD-5 9 Volt adapter is rated at a *substantial* 850 mA DC (the specs are not given in the *Operation Manual*). Many alternate adapters (e.g., Roland "Boss," Radio Shack units) cannot deliver this much current, and will cause the batteries to be depleted prematurely. This won't hurt anything, but the batteries may become so weak that the APO (automatic power off) function will turn off the CZ even with the adapter plugged in. If this happens to you, don't think the machine is broken—just replace the batteries. Use heavy-duty alkaline batteries (the *Manual* refers to these as "manganese dry batteries"). Of course, there's nothing sacred about the Casio adapter. You can use any quality 9V adapter rated for at least 850 mA, and wired for the *proper polarity*—negative tip (inner connection), positive sleeve (outer connection). (*Warning: reverse polarity can severely damage your instrument.*) We'll show you how to build your own "adapter"—an IC-regulated power supply with onboard surge suppression—in the next article, as well as how to send power over MIDI or audio cables.

The six D-cells add considerable weight to the CZ-101 (especially when playing the unit as a strap-on keyboard), and they're *expensive*—but they can be eliminated. When using the proper AC adapter, the only function of the batteries is to back up the internal RAM, which requires about 0.01 mA of current. So, a single alkaline 9V rectangular battery could back-up the RAM

Fig. 4

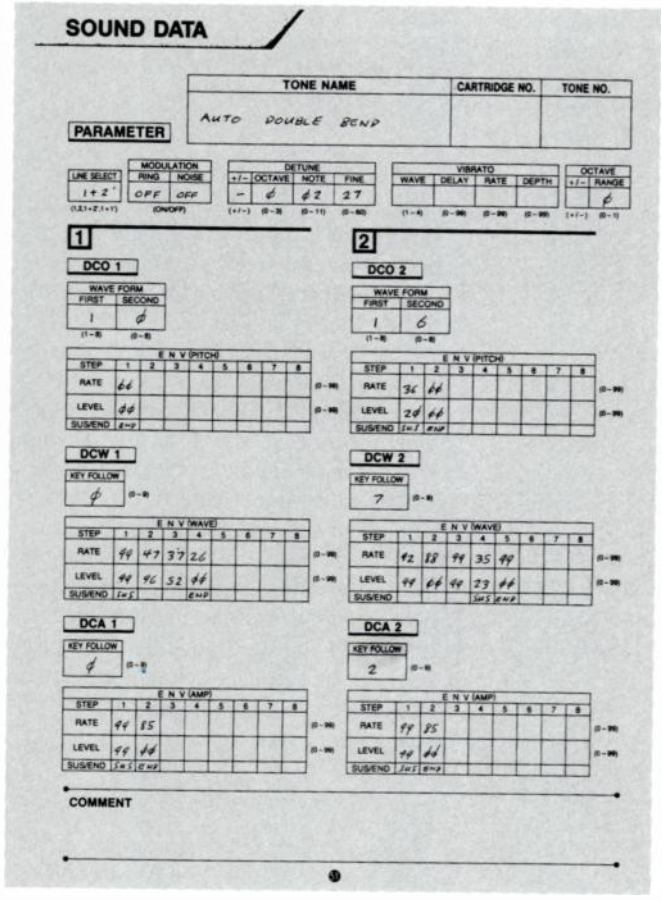
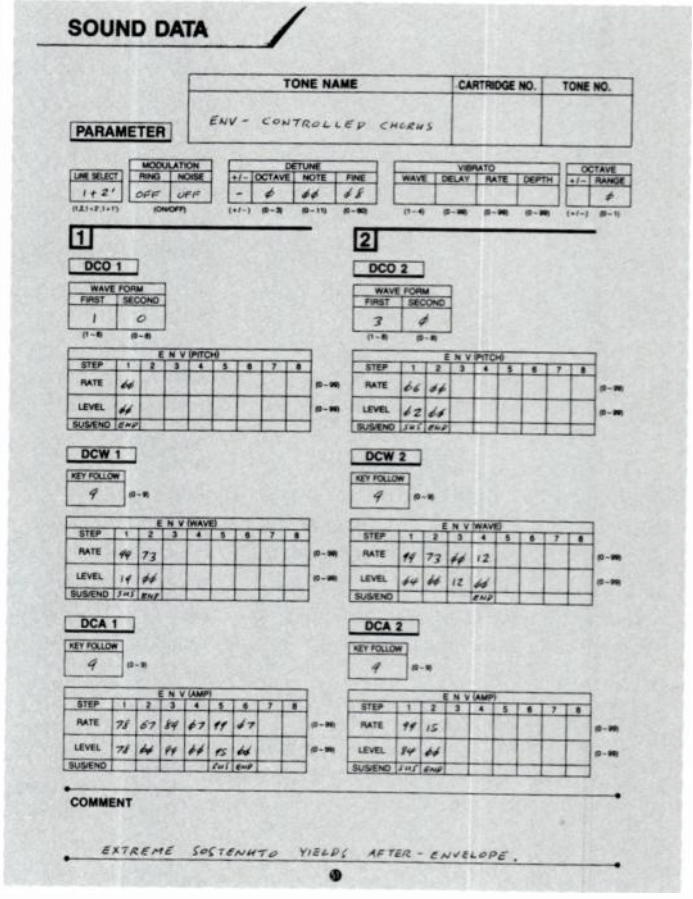


Fig. 5



for virtually the shelf-life of the battery! The power capacity of those D-cells is not needed for memory back-up at all. Casio needed the D-cells as a primary power source to keep the instrument as inexpensive as possible. (You buy the adapter *separately*, remember?) The one-year battery-life estimated in the manual is just the average shelf-life of the batteries.

RAM Cartridge/Patch Librarians

Since the CZ-101 has only 16 user-writeable internal memory locations, program storage can become a problem in a hurry.

The dimunitive RA-3 plug-in RAM Cartridge stores 16 additional programs. The RA-3 contains RAM (not EEPROMs), "backed-up" by a BR-2016 lithium battery. If the battery goes flat, you'll lose the RAM cartridge programs. (Similarly, if the CZ-101's batteries go flat, you'll lose its programs, too.) Since all of the CZ system's RAM is potentially volatile, and there's no cassette interface, the instrument by itself is not exactly a testament to data security. I'm sure that Casio bypassed EEPROMS, in favor of RAM's, to reduce cost. But, even so, the RA-3 isn't inexpensive.

In general, I'm a little disappointed with the RA-3. To replace the battery, you have to take the cartridge apart The case is held together with miniature wood screws that thread directly into styrene plastic. Even with the proper tools and care it's easy to ruin the screw heads while trying to produce enough torque to overcome the grip of the plastic. To make matters worse, the battery and holder exert pressure on the top half of the case, causing it to "bow."

Well, one could always record patch sheets by hand, on paper. (You should do this anyway, for important patches.) But, if you have a MIDI-interfaced personal computer, there's an easier way to store lots of patches. Patch Librarian software (such as programs available from Hybrid Arts, Op-code Systems, and Dr. T) can store hundreds of CZ patches on inexpensive floppy disks instead of costly RAM cartridges. Patch Librarians transmit, receive, display, modify, print *and* store program data via MIDI. If you have a personal computer with MIDI, this is the way to go.

Printed Matter and Accessories

Jerry Kovarsky, product manager for Casio E.M.I., informed me that some of the early CZ-101s were inadvertently shipped without the "Sound Data Book." This 51-page manual contains patch sheets for the 16 sounds in the Internal (P-button) memory and 32 others, plus some blank patch sheets you can photocopy. If your copy was missing, contact Casio's Customer Service Division at the address below. A new "Pro-Tone Data Book, Volume I" containing 81 alternate sounds for the CZ instruments, will be available early this year. This manual will be sold as an accessory by Casio E.M.I. dealers, and may be ordered directly from Casio. All accessories (*not* keyboards) may be ordered direct; refer to the address below.

The CZ-101 MIDI System Exclusive Information is available in a pamphlet entitled "Cosmo Synthesizer CZ-1000/101, MIDI System Exclusive." The SYS EX commands are fairly extensive, including program data send and receive functions. "Hackers" will definitely want to get a copy of this reference. Also available are: "How to Calculate Real Time of CZ Envelope," by M. Fukuda; "Cosmo Synthesizer CZ-5000 MIDI System Exclusive;" MIDI Implementation charts for the CZ-5000, SZ-1 Sequencer, and CT-6000 Keyboard, and a list of Authorized Service Centers. To contact Casio, write: Customer Service Division, CASIO, INC., 15 Gardner Road, P.O. Box 1386, Fairfield, NJ 07007. Telephone 201/575-7400. EM

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Yes, you can buy a cathedral in a rack mount box for under $1,500...in fact, you have several choices of sonic cathedrals in this price range. Last month, Part 1 of this two-part series told you what to look for in a digital reverb; this month, Part 2 tells you where to look.

Sing a Song of Reverb (Part 2): Architectural Spaces Available Cheap

BY LARRY OPPENHEIMER

For most of us, spending $5,000 or more for a signal processing device is a dream. Fortunately, some of the hottest action in digital signal processing is occurring at prices under $1,500. This is not only a much more realistic amount of money for most small studio budgets, but the increasing quality of these instruments makes them useable as second and third reverbs in larger studios that want to "inexpensively" supplement their more costly processors. In this report, we'll limit ourselves to digital reverbs (DRs) costing $1,500 retail or less. As of this writing, the roster includes nine DRs: the Alesis XT:c; ART DR1, DR2a, and 01a; DOD RDS-6400; Lexicon PCM60; Roland SRV-2000; Ursa Major StarGate 323; and Yamaha REV7. Although all of these units share certain crucial features, each also has its own approach and unique features.

DEFINITIONS

Before describing these products, let's define terms. There is very little agreed-upon DR terminology; each manufacturer uses whatever seems appropriate, which makes life difficult for those of us who try to describe these units (not to mention use them). While this is confusing at first, you should have a pretty good idea of what's going on by the time we get to the product descriptions.

Perhaps the most confusing point (at least to me) is the way in which the term "program" is used. For example, suppose one manufacturer advertises a reverb as having "five different plate programs." This could mean that there are five programs designed to create the sound of five different plates; or, there could be a single plate program that is manipulated into five additional variations. You will also see the term "algorithm" used, such as a unit which claims to use plate, hall, and room algorithms. From a literal, computer programming standpoint, "algorithm" and "program" are virtually synonymous: an algorithm is a general procedure, executed by a computer, that achieves a desired result; a program is the specific sequence of computer language instructions that actually implements this procedure.

So that we can keep terms straight among ourselves, in this article "program" will refer to a simulation of a particular reverb system (plate, room, hall, etc.); variations (small hall, large hall, bright hall, etc.) could be different algorithms designed to simulate the same system, or they could just as easily be variations of the same algorithm (only the manufacturer knows for sure). Therefore, if a reverb claims "five hall programs, three plate programs, and three room programs," we will consider that unit as having three programs (hall, plate, and room); the other sounds are considered variations, regardless of whether they use their own algorithm or are manipulated versions of the "core" programs. In the chart of features accompanying this article, the terminology is mine, and conflicting terms used by manufacturers are ignored. Ready? Let's take a look at these wonderful toys.

ALESIS XT:c

Alesis was founded by Keith Barr, a founder of MXR and creator of the original MXR 01 (possibly the first under-$1,500 digital reverb; Yamaha's R-1000 is the other possibility). Barr left MXR, which then transmuted into ART, and formed Alesis. He quickly brought out the XT, but soon devised improvements that resulted in the recent release of the XT:c. At $750, the XT:c is at the bottom of the price

Larry Oppenheimer is a musician, sound engineer, electronics technician, and consultant, as well as a writer. His San Francisco-based company, Toys in the Attic, offers a variety of consulting services in the fields of MIDI, product documentation, and signal processing. He grows hair under his lower lip and still owns a considerable number of ARP 2500 synthesizer modules.

What is MIDI Program Mapping?

MIDI program change commands simplify life when slaving two synthesizers together; for example, if you change from program 1 to program 2 on the master, the slave will also automatically switch from its program 1 to program 2. Thus, it is only necessary to push one button to change sounds on two synthesizers.

What happens, though, if you want the slave to change from program 1 to, say, program 14? You could simply copy slave program 14 into program slot 2; then, selecting 2 on the master would call up program 2 on the slave, which would now contain the sound of program 14. However, since copying programs around can be time-consuming, manufacturers devised the concept of *mapping* program numbers.

In the case of the REV7, for example, you can program which reverb preset will be called up by a given program number on your master synth. There need be no correlation between these numbers; different synth programs could call up different reverb presets or for that matter, the same reverb preset...all you need to do is program in this information, and the REV7 will translate the program number it receives into the desired REV7 preset number. —*Editor*

barrel, but its sound compares favorably with other units in this price range.

Perhaps the XT:c's most interesting feature is the way its programs are set up. Two buttons choose between four programs; while these programs are not specifically constructed to imitate the common hall, room, and plate programs, they fill the same basic sonic niches. There's also gated reverb. Pressing a "Mod" button modifies each of the four programs significantly, yielding four additional programs. These additional programs have extended decay times and, in the case of program four, gated reverb becomes reverse reverb. For a reverb at this price to offer eight different programs is great. The front panel also includes a "Hold" pushbutton to create infinite reverb.

The XT:c also offers variable predelay adjustment (up to 250 milliseconds!), decay time, and variable high frequency rolloff. Other front panel controls include a low frequency cut filter, high frequency damping switch (HF rolloff sets frequency, HF damping sets time), and knobs for input level, output level, and dry/reverb mix. The rear panel includes a pair of jacks for an insert loop. The XT:c's two inputs are actually summed to mono internally, as are most stereo inputs on units in this price bracket, but there are true stereo outputs...all in all, a lot of bang for the buck.

Barr is working very hard at increasing DR performance, both in this price range and in the next higher bracket (around $5,000). Look for more to come from Alesis.

Applied Research and Technology (ART)

ART currently offers three under-$1,500 models. The top-of-the-line DR-1 ($1,295) is a programmable, MIDI-controllable processor with a number of programs and a compact, full-function remote control. The DR-1 has been designed to allow ART some flexibility and room for growth in their software, a ploy which is useful in maximizing the number of effects that a given hardware design can perform. The current software revision offers hall, plate and room reverb programs (with five variations of each program) plus four reverb "effects" (cavernous, reverse, gated, and resonant) and a stereo delay line.

Most of the standard parameters (decay, predelay, HF damping) are controllable, as well as several less common ones, such as diffusion and position (depth). Some features, such as decay time and memory, are quite robust—there's no skimping here. The maximum decay time is 25 seconds, and you'd be surprised how often you use such absurdly long times when they are available. For storage there are 100 user memories in addition to the 30 factory presets.

Two additional features are unique to this box. If the minimum decay time parameter (normally set equal to the regular decay time parameter) is changed to a value lower than the regular decay time, then whenever the input signal exceeds a software-defined threshold level it is processed with the minimum delay time; if the input goes below the threshold, the regular (longer) decay switches in. This is similar to the "dynamic decay" feature pioneered in Lexicon's 224X, and is very helpful to prevent muddying up of a sound while nonetheless preserving the ability to have a long "tail" when the music stops. The other unique feature is the Kill/Inf button, which can be in any of four modes: inactive, reverb kill, decay kill (leaving only early reflections), or infinite reverb hold.

The 01a ($1,095) is the next step below the DR-1. It is the descendant of the original 01 and the forerunner of the DR-1. The 01a has five room program algorithms, two "dynamic rooms" (as in the DR-1), reverse and gated reverb, most of the standard parameters, and 49 user memories. A rear panel "hold" jack allows infinite reverb. The 01a, being older and less expensive, does have limitations compared to the DR-1 (10 kHz bandwidth instead of 14 kHz, and a 12-bit A/D converter system instead of the DR-1's 16-bit converter).

The DR-2a, at $795, is competing with the Alesis XT:c and DOD RDS-6400. But even at this price, ART includes three user memories, five variable parameters, and several programs (three Plate, two Room, and two Hall algorithms, plus the now-common gated and reverse reverb). The decay time can even be pushed out to a more than reasonable 13 seconds.

ART includes some of the less significant yet practical features on their reverbs, such as user memory protection to avoid accidental erasure or change, and reverb

Fig. 1 Digital Reverberators under $1,500: Audio Features

Make	Model & Price	band width	A/D	#of I/O, Bal/Unbal	lvl ctrl	gain boost	mutes	cut filters	mix ctrl	eq	insert loop
Alesis	XT:c $750	14kHz	16 bit linear	2U/2U	S	-	-	H, L	Yes	-	Yes
ART	DR-1 $1,295	14kHz	16 bit linear	2B/2U	-	I/O	0	L**	Yes	-	-
	01a $1,095	10kHz	12 bit linear	1B/2U	-	-	0 *	-	-	-	-
	DR-2a $795	10kHz	12 bit linear	1B/3U	-	I/O	0	L**	Yes	-	-
DOD/ Digitech	RDS-6400 $699.95	10kHz	12 bit linear	1B/4B	S	I/O	-	-	Yes	-	-
Lexicon	PCM-60 $1,040	10kHz	16 bit linear	1B/2U	S	S	B	-	Yes	-	Yes
Roland	SRV-2000 $1,495	10kHz	16 bit linear	1U/2U	I	I/O	S, B	-	Yes	3 band digital: Sweepable LF shelving, MF & HF parametric	-
Urse Major	StarGate 323 $1,300	15kHz	15 bit linear	2B/2U	I	-	I, 0, 0*	-	Yes		
Yamaha	REV-7 $1,195	12kHz	16 bit linear	2B/2B	I	-	0, B	-	Yes	3 band analog, sweepable	-

Notes and Abbreviations

* 01a and StarGate 323 output mutes actually reduce decay time to minimum allowable for the program.

** Low cut filter switch located inside chassis.

1. # of inputs and outputs and balancing are shown in the form: (# of inputs)(B or U)/(# of outputs)(B or U)
2. For level control, gain boost, and mutes: I = input control only, 0 = output control only, I/0 = single control for both, S = seperate controls for each, B = bypass
3. An input mute is defined here as muting input sent to the reverb processor. An output mute is defined here as muting output from the reverb processor. In these cases, the dry signal is unaffected.
4. For cut filters: H = high frequency cut, L = low frequency cut
5. All units use 1u of rack space, except REV-7 and StarGate 323, which are 2u.
6. All units use 1/4" phone jacks for input and output, except for StarGate 323, which uses XLR-type. In addition, the REV-7 has XLR type connectors, and the 01a has barrier strip connections for input and output.

kill jacks and/or buttons. They do not include power switches, however.

DOD/DIGITECH RDS-6400

DOD is the newest entrant into the "cheap" DR sweepstakes; in fact, as of this writing the RDS-6400 was not yet available for evaluation. DOD did show a prototype at the October Audio Engineering Society convention in New York, but the technology is new to them and, as has been stated before, it is a rather tricky and laborious undertaking to make a good DR at any price, much less at the $699.95 quoted for the 6400.

In terms of features, the approach (and even the front panel) of the RDS-6400 is extremely reminiscent of Lexicon's PCM60 (see below), which has a small set of parameters, each with only two or four settings selected by pushbuttons. As in the PCM60, there are four room sizes, four decay times, and two programs (plate and room). As you are reading this, the RDS-6400 is probably arriving at your music store.

LEXICON PCM60

If you've only ever heard one DR, chances are it was a Lexicon. Although not the first company to market one (that distinction belongs to EMT), Lexicon's 224 established digital reverb as a standard element in the sound engineer's arsenal with its combination of versatility, ease of use, portability (very few EMTs were ever taken on tour), and price (the first under-$10,000 DR). With the introduction of the PCM-60 in 1984, Lexicon entered their fourth generation of digital signal processors.

The company's concern for quality had kept prices on most of their line out of reach of small studios and musicians for a number of years. If the PCM41, introduced in 1980, was their attempt to crack the market for low cost digital delays, the PCM60 is their bid to do the same for digital reverb. This was re-emphasized by Lexicon's recent price cut on the 60, from $1,495 to $1,040. Consequently, the PCM60 is somewhat sparse on features, both for ease of use and as a cost-cutting compromise.

As with the DOD unit, the 60 has only two programs, Room and Plate, with four different choices for room size, decay time, and frequency contouring. Thus, this arrangement can produce a maximum of 128 different setups. The PCM60 has been well received because of its sound and Lexicon's reputation, but there is a more subtle point here.

Lexicon is now offering the heart of the PCM60 as a small printed circuit board available to OEMs (an OEM, or Original Equipment Manufacturer, item is a sub-system made available to other manufacturers for inclusion in their products; Dolby noise reduction chips for cassette recorders are one example of an OEM item). Is this the dawning of synthesizers and mixing consoles with onboard digital reverb? Although it now takes a whole PC board, it is not inconceivable that a basic but high quality digital reverberator may shrink to a small PC card scarcely larger than a 6x8 file card. Intriguing possibilities come to mind...

ROLAND SRV-2000

The number of features on this machine is almost obscene. Programmability, several modes of operation, onboard digital equalization, MIDI compatibility, and a large number of variable parameters make this machine extremely versatile. Fortunately, the SRV-2000 has been thought out well enough that the features are generally accessible without a significant hassle. This is largely due to a good front panel layout, which uses a combination of fixed function global controls (for storing setups, selecting mode, etc.), software defined parameter controls, and a display that shows six parameters simultaneously.

There are three modes: Reverb, Non-linear, and Room Simulate. Reverb mode provides the ubiquitous Hall, Plate (two different kinds), and Room programs, each available in several sizes. Instead of having a room size control, the Reverb Select button steps through several sizes of one program before going on to the next. The Non-linear mode provides the gated and reverse reverb sounds without which you must still be living in the '60s (horror of horrors!). The Room Simulate mode provides "ambience" (early reflections) without the buildup of a reverberant "tail." How about parameters? Although the SRV-2000 includes all the standard stuff (decay time, HF damping, predelay), it is the extended parameter control (called "Further Level") that really sets it apart. In this mode, reverb density can be varied, and a variety of parameters pertaining to

Fig. 2 Digital Reverberators under $1,500: Software Features

Make & model	programs	Parameters								
		decay time			diffusion	depth	max predelay	size	HF damping	early reflections
		max	type	HF/LF						
Alesis XT:c	H(2), P, R(2), G, RY, I, FX	15 s	Y	-	-	-	250 ms	-	S	-
ART DR-1	H(5), P(5) R(5), G, RY, I, FX(2)	25 s	Y	-	Y	Y	200 ms	-	Y	-
01e	H(2), P(2) R, G, RY, I	24 s	Y	-	Y	Y	180 ms	-	Y	-
DR-2e	H, P, R, G, RY	12.8 s	Y	-	S	S	75 ms	-	S	-
DOD/Digitech RDS-6400	P, R	10 s	S	-	-	-	-	S	S	-
Lexicon PCM-60	P, R	4.5 s	S	S	-	-	-	S	-	-
Roland SRV-2000	H, P(2), R(2), G, RY, I	99 s	Y	-	Y	Y	160 ms	Y*	Y	lvl, density, attack time and gain
Urse Major StarGate 323	H(3), P(2) R, FX(2)	10s	Y	Y	-	-	200 ms	-	-	-
Yamaha REV-7	H, P, R(2), FX(3)	24 s	Y	Y	Y	-	100 ms	Y	-	6 "modes", 1st refl. dly and lvl

Notes and Abbreviations

* The SRV-2000 has five hall and eight room sizes.

1. Programs: H = hall, P = plate, R = room, G = gated, RY = reverse, I = infinite, FX = effects
   Number in parentheses indicates number of algorithms of each type available.
2. For all parameters: S = Selectable (four or less choices of value), Y = Variable (more than four choices of value)
3. Decay time: HF/LF = contouring of high frequency and low frequency decay times
4. A dash (-) = parameter not user-adjustable or not present.
5. All units with predelay are variable except the DR-2e, which is selectable.

Fig. 3 Digital Reverberators under $1,500: Programmability, MIDI, and other features

Make & model	Programmability # of factory presets	Programmability # of user memories	MIDI program change	MIDI advanced features	other features
ART DR-1	30	100	Y	-	Dynamic rooms, remote control
01e	7*	7	-	-	Dynamic rooms
DR-2a	0	3	-	-	
Roland SRV-2000	16*	32	Y	Y	Footswitch operation of mutes and memory recall.
Yamaha REV-7	30	60	Y	-	Remote control

Notes

* Factory presets on these units require the use of user memory space. Loading the factory presets destroys the contents of those memories.

1. Factory presets are defined here as presets stored in ROM.

early reflections can be set, including Attack Gain, Attack Time, Density, and Level. Unfortunately, to get into this mode, it is necessary to actually turn the power off and back on while holding down several keys, then press those same keys again. This seems a little screwy, but Roland gets away with it because the SRV-2000 does not put out glitches on power down/up (or bypass, amazingly enough), and, with its memory capabilities, nothing is lost in the transition. Returning to normal operation requires the same procedure.

The SRV-2000 also has three bands of pre-reverb digital EQ: low frequency shelving and both mid and high frequency peaking. The EQ is fully parametric, providing from -24 dB cut to +12 dB boost in each range, adjustable frequency, and adjustable Q (for the mid and high peaking filters).

As if all this weren't enough, the machine is also MIDI controllable... however, what exactly does "MIDI controllable" mean? As we discussed last month, for most signal processors it means the ability to recognize program change information and call up a preset from memory. But a signal processor is subject to the same problem with MIDI program change as a synthesizer, namely, the difference in program numbering schemes used in various machines. What does the machine do when it has 32 memories and receives a command to change to program 62? Most manufacturers of MIDI-controllable processors are providing some means of mapping MIDI program change numbers to the processor's memory or preset numbers to deal with this problem (see sidebar).

In the case of the SRV-2000, though, the situation is further complicated by the fact that Roland claims that parameters can also be varied through MIDI System Exclusive information. Unfortunately, a processor that can recognize MIDI information does not necessarily transmit it, as is the case with the SRV-2000. In fact, it doesn't even have a MIDI Out jack, although there is room next to the MIDI In and Thru jacks to add one later. Roland states that they are developing a personal computer interface for the SRV-2000, and anticipate the development by themselves or a third party software house of PC software to generate the necessary System Exclusive codes. In the meantime, there isn't any way to access this wonderful ability. In spite of this limitation, the SRV-2000, at $1,495, is a very impressive machine.

Ursa Major Stargate 323

Ursa Major was one of the first companies to get involved with digital reverb, anticipating the age of cheap digital signal processing with their Space Station, which sold for around $2,000. A recent price cut has brought the StarGate 323, one of their most recent models, down to $1,300—a necessary move to remain competitive in this explosive arena.

The 323 stays true to Ursa Major founder and chief engineer Christopher Moore's idea of creating one algorithm flexible enough to perform many tasks. Indeed, the 323 can create spaces ranging from extremely small (with decay times as short as 100 milliseconds) to extremely large (with decay times as long as ten seconds).

MIDI Parameter Control: Which Approach Is Best?

It has been this writer's contention for a number of years that external control of reverb and effects parameters would be a very useful feature, but until MIDI came along, no manufacturer was willing to commit to an interface to accomplish this. Naturally, as soon as a "standard" interface arrived, in the form of MIDI, people wanted to do things that were far beyond MIDI's originally conceived uses, such as control entire orchestras of instruments from a personal computer...or externally control signal processors. Of course, there is no definition in MIDI for most of the appropriate parameters, so the question then became: should the processor define its own MIDI codes using System Exclusive information, or should it apply a unique interpretation to standard codes defined in the MIDI specification?

Roland seems to have chosen the former approach in the SRV-2000, while Lexicon seems to be opting for the latter. In the PCM70, it is possible to assign up to ten parameters in each program to be individually or collectively controlled by any of the controllers defined in the MIDI specification. These assignments are stored as part of a setup in a user memory. For example, a reverb program can be set up so that decay time increases as you play up the keyboard, or chorusing modulation becomes stronger as you increase a synthesizer's modulation wheel.

The concept of creating dynamic signal processing by using controllers that are more natural than those typically found on a DR is new, and opens up a number of areas for creative investigation. Additionally, using standard MIDI control codes allows the information to be recorded in a stand-alone or personal computer-based MIDI sequencer and replayed automatically, or even synched with SMPTE time code if you add a SMPTE-to-MIDI box (SBX-80, SMPL Lock, et al).

Another function that would be useful to implement on MIDI-equipped processors is parameter data dumping, so that user memories can be off-loaded to a sequencer or computer for storage. Although it is possible to dump parameters from one PCM70 to another over the MIDI port, it is not now done with MIDI codes. Hence, the PCM70 does not, at this time, allow offloading of parameters into a computer, which the System Exclusive approach does. Once again, the jury is still out on this, but it may never come back with a solid verdict of "guilty" or "innocent;" Lexicon does intend to add other parameter dumping options later on, which are likely to include MIDI.

Even with this wide variety of sounds, the 323 is very simple to operate. The front panel is logically laid out and uses color-coded knobs: blue for variable parameters, and red for selectable parameters (stepped pots). The 323 offers a selectable overall decay time control for "coarse" adjustment, and separate LF and HF decay time controls for "fine" adjustment. It also offers three kinds of mutes: Input Mute, which cuts off input to the processor and dry signal; Dry Only, which cuts off the reverb output; and Reverb Clear, which forces the reverb decay time to 0, leaving only early reflections.

It is also worth noting that Ursa Major provides particularly helpful documentation that covers the unit's performance, interfacing suggestions, and tutorial information on reverb. In fact, the 323 even has such useful information as phasing of the inputs and outputs printed right on the back of the chassis.

YAMAHA REV7

Yamaha leapt into the fray with both feet in 1983, introducing both high- and low-end DRs. If the REV-1 was a no-compromise machine, then the R-1000 ($795) was a complete compromise. The REV-1 had more parameters than anything on the market except the Lexicon 224X, but the R-1000 had only one: decay time. Four "Mode" buttons on the R-1000 selected decay times ranging from 1.5-2.4 seconds, which, according to Yamaha's literature, "should be more than sufficient for just about any reverb requirements." The machine did include three bands of sweepable analog EQ following the reverb processor, which was quite useful in getting some variation; nonetheless, the R-1000 was limited at best.

But it sometimes seems Yamaha brings out a first-generation product with some

obvious flaws, only to follow it up later with a killer second-generation machine. At the Spring 1985 AES show in Anaheim, Yamaha showed their killer—the REV7. This machine is set up to be the most general-purpose digital signal processor in its price range. Of course there are plate and hall programs, but there are also two "Early Reflection" programs, delay, flanging, chorusing and tremolo programs, all with a good number of variable parameters. Six parameter buttons, three dedicated and three "soft" (software-defined), access a total of about 16 aspects of the programs that can be manipulated (including HF and LF decay time, diffusion, first reflection delay time and level, number and "mode" of early reflections, room size, modulation frequency and depth for the effects, etc.). Of course, not all parameters are available on all programs; in fact, no program has more than seven parameters. Still, that's pretty good in a $1,195 package.

The three-band sweepable analog EQ is still there, although it is now pre-processor like the Roland's, but the maximum decay time is about 24 seconds. The REV7 is programmable, with 60 user memories in addition to 30 factory programs. It is also MIDI-controllable, as one might expect; although limited to program changes, the different REV7 patches can be mapped to different program numbers. The REV7 also comes with a remote control, although it can only step through user memories.

Lexicon's PCM70 and The User Interface

I know what you're thinking . . . the PCM70 is neither $1,500 and under (it retails for $2,300) nor intended primarily as a reverb (it is, first and foremost, an effects machine). It is, however, a rather remarkable piece of gear and worthy of inclusion here for several reasons, not the least of which is the value of looking ahead at what kinds of features might trickle down in the next few years. The 70 is essentially the first of Lexicon's fifth generation, and it shows. The original concept was to make an affordable box with the effects programs of the 224XL and a full MIDI implementation. Because it was adapted from a digital reverb architecture in the first place, it was not too difficult to put reverb in the 70, although the idea was added late enough to almost be considered an afterthought. Fortunately, the PCM70 is still quite a nice reverb, featuring three reverb programs and advanced features like dynamic decay. More significantly, though, the PCM70 addresses several sticky design issues. One is the question of the user interface.

As inexpensive reverbs get increasingly sophisticated, they run into one of the toughest design problems imaginable: how can you give extensive control over a machine, yet still have it be easy enough for a fast-paced engineer (who doesn't have time to read the manual) to use proficiently? The PCM70's solution is to have a small amount of largely software-definable front panel controls, combined with a matrix organization of the programs and parameters.

Is this approach effective? The jury is still out, as the product is quite new. It is still necessary to have external documentation on hand (i.e. the chart of the program and parameter matrices) until you get to know the machine well. This was also a problem with the 224X, but does not seem to have impeded its success or usefulness. In fairness, it should be noted that the organization of the matrices is quite logical and the number of programs relatively small; with regular usage of the machine, this information can be assimilated on an instinctive level.

Additionally, Lexicon is developing a PCM70 personal computer interface to allow more flexible manipulation. As has been amply demonstrated by recent releases of software for controlling complex instruments like the DX7 and the Emulator II, this can be a very effective compromise between cost—graphic displays are expensive to build into a low-cost unit—and control. Of course, it costs money to build in a computer interface, so this may be an option; but it is reasonable to expect that this feature may start to become available in lower-cost devices (such as the aforementioned Roland SRV-2000).

Conclusions

Amazing things are happening in the area of low-cost digital signal processing. Each box is at least slightly different than the rest, but all have something to offer—so look closely at features. Most of all, listen as carefully as you can when deciding what to use. Play a variety of source material through the machine, from test sources such as clicks to dry tracks (snare drum, vocals, piano, guitar, synthesizer, etc). Play some familiar material and fiddle with the knobs to gauge their effect. Try cutting off the source material suddenly (using the input mute if present) and listening to the reverberant decay. If possible, do your evaluation in your own facility or one which is familiar to you. And above all, don't forget to dig those crazy sounds. EM

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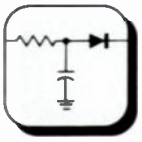

Any hardware hacker knows why audio range VCOs need to follow an exponential response. But what's to be gained by having an exponential oscillator response in a range that even dogs can't hear? Plenty, says our Practical Circuitry columnist . . . read on and find out why.

Build a High Frequency VCO

BY THOMAS HENRY

Many electronic music circuits (flangers, chorus units, echo units, top octave divider circuits, and so on) require a source of high frequency clock pulses. It is easy to make manually-adjustable clock circuits for various frequencies, but adding voltage control over the output frequency starts to get a little trickier. One of the more common ways to accomplish this is to use the VCO of the 4046 phase locked loop IC, since it features not only a control voltage input, but can also oscillate up to about 1 MHz. This was the approach taken for the Synthesizer Delay Line described earlier in this column.[1] The one drawback of this method is that the output frequency changes linearly with respect to a change of input control voltage. While linearity is often a desirable quality, for most music applications it leads to unnatural sounds. In fact, most all musical parameters are exponential in nature!

Here's a circuit that opens up a whole new world of exciting projects by overcoming the disadvantage of linear response. The High Frequency VCO (HFVCO) provides a train of pulses suitable for clocking most of the circuits mentioned earlier, and best of all, the output frequency can be modulated in a precise one volt per octave fashion, over a seven octave range. When you throw in the capability to do linear or exponential FM, coarse and fine tuning, and temperature compensation, the HFVCO starts to appear as the most attractive way to generate precise clock signals for modern electronic music circuits.

Thomas Henry is a contributor to several magazines including Radio-Electronics, Compute!, Keyboard, *and* Run. *He played guitar professionally for ten years in order to put himself through school; currently he is an assistant professor of Computer Science at Mankato State University in Minnesota. Aside from computers and electronics, Thomas has a passion for Victorian literature, Sherlock Holmes, and the works of Oscar Wilde.*

WHAT'S IT GOOD FOR, ANYWAY?

At first, a circuit like this probably only seems exciting to the designer, so let me take a moment to try to convince you of its great potential. To do so, let's consider flangers. One of my favorite effects is to pump a square wave from my synthesizer into a flanger, turn the modulation *down* all of the way, adjust the delay time manually until I hit an interesting timbre, and then just leave the flanger at that setting. This technique generates all sorts of hollow, resonant sounds that are very useful. But unfortunately, this effect is not *global*. That is, some of the notes sound just right, while others will cancel out entirely! So, playing a far-ranging lead line becomes quite unpredictable. This, of course, is not a quirk of the delay line, but is a natural consequence of the time delayed signal adding in with the dry signal. Now imagine a delay line incorporating the HFVCO. If you apply a 1V per octave control signal not only to the synthesizer VCO *but also to the delay line modulation input*, the waveform thus generated would remain fixed (or at least manageable) over the entire length of the keyboard!

Still not convinced that the HFVCO is for you? Well, come back next month to "Practical Circuitry" when we'll use it to make a combo organ think it's a synthesizer! For now, let's see how the basic circuit works.

HOW IT WORKS

Refer to Fig. 1, the HFVCO schematic. The circuit can essentially be divided into two parts. There's the oscillator itself, comprised of the SSM2031 (and virtually nothing else!), and the exponential current source. Those of you who have been involved with electronic music for a while will instantly recognize the current source: this top quality configuration has been well honed by a variety of designers over the past ten years and is a proven winner. Since the exponential current source forms the heart of countless synthesizer modules, both commercial and homemade, this would be a good time to give thanks to those early designers who have made our work that much easier. In particular, Terry Mikulic, Bernie Hutchins, and Dave Rossum deserve a bit of tip-of-the-hat.[2,3]

Let's work through the circuit now. To properly control the oscillator, we need to convert a control voltage input to an exponentially related current. This is the task of transistor Q1. The basic idea here is that Q1's collector current is exponentially related to the base-emitter voltage. If only Q1 were used, we would have real troubles, since the emitter saturation current changes quite drastically with respect to temperature. Adding in transistor Q2 cancels this temperature dependence. The cancellation is quite effective since Q1 and Q2 are matched transistors and share the same substrate. It's beyond the scope of this article to work through the equations, but it's enough to keep in mind the following two concepts: (1) the collector current of Q1 varies exponentially with respect to base-emitter voltage changes; (2) Q2 helps cancel a devious temperature effect.[4]

Q1's collector current doubles with every 18 mV change in the base-emitter voltage. To match this with our normal 1V per octave response, op amp A1 and related components provide suitable scaling. J1 is the 1V per octave input and this feeds R7. R7 in conjunction with thermistor R3 sets an attenuation factor of 50; therefore, a 1V change at the input leads to a 20mV change at the output. Trimpots R1 and R2 attenuate this still further; properly adjusting R1 yields a precise 18 mV response.

Potentiometers R10 and R11 provide fine and coarse tuning adjustments, respectively. Notice that summing resistor R16, at 3M3, is considerably larger than R6 at 100k; this is what gives the pots the "fine" or "coarse" feel when setting an initial tuning. One other point to note is

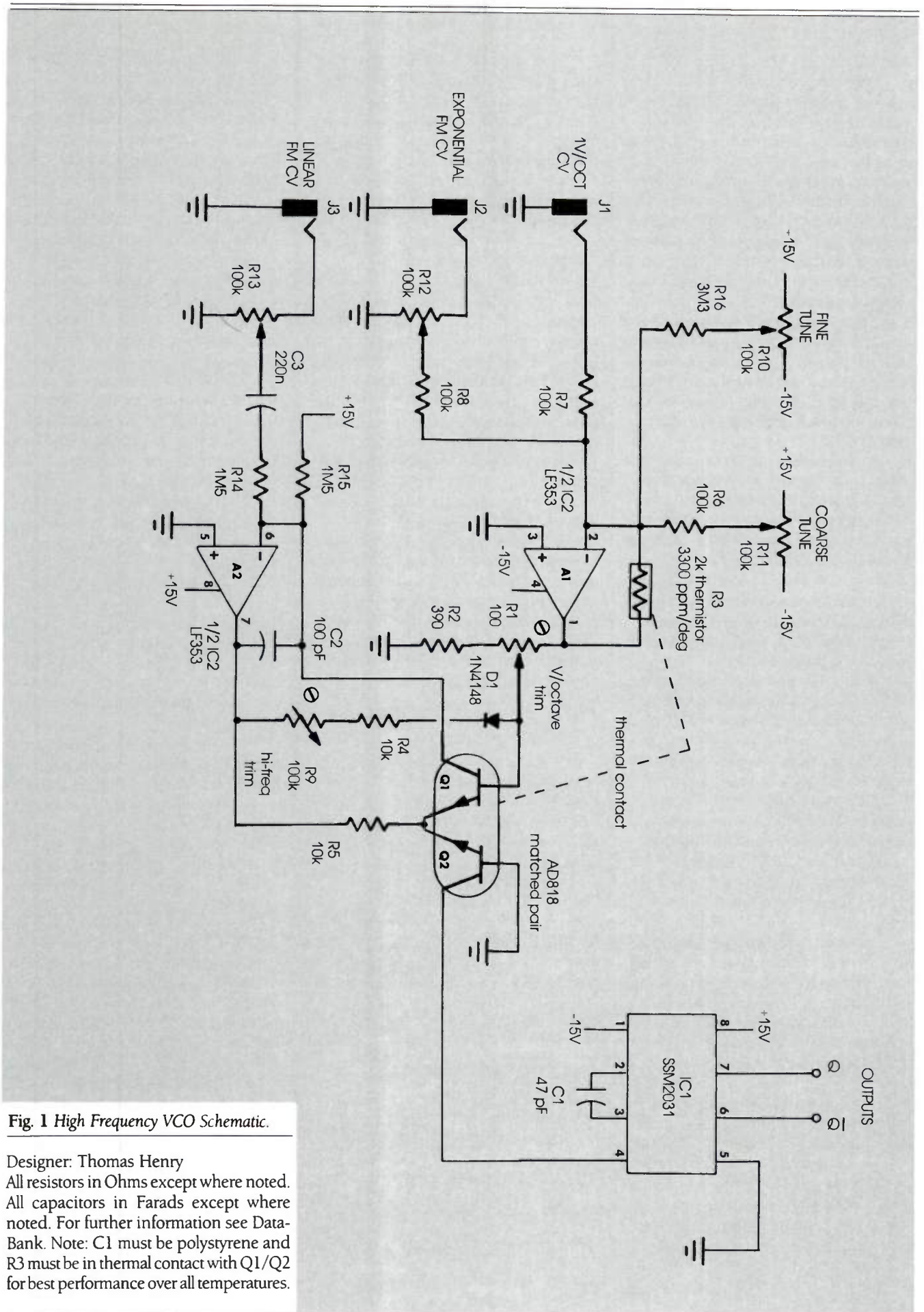

Fig. 1 *High Frequency VCO Schematic.*

Designer: Thomas Henry
All resistors in Ohms except where noted. All capacitors in Farads except where noted. For further information see Data-Bank. Note: C1 must be polystyrene and R3 must be in thermal contact with Q1/Q2 for best performance over all temperatures.

that R11, the coarse tuning pot, covers a very large range and will in fact not provide any useful tuning control at the two extremes. The reason for leaving this apparently undesirable feature in is that there may be times when you will apply various control signals containing offsets via jacks J1 and J2. The wide swing of the coarse control will allow you to overcome these offsets, thus bringing the VCO into its more reliable operating range.

As mentioned, J1 is the control input you would use when you wish the HFVCO to track at a precise 1V per octave rate. There may be other times, however, when you will want to modulate the oscillator's center frequency in an exponential fashion. Use J2 in this case. Attenuator R12 allows you to set the amount of frequency modulation.

For best results, the exponential current source must operate in the proper range. R15 sets the reference current which guarantees that the output current corresponding to an input voltage of 0V is 10μ. This is about the lowest value with which we can expect good results.

A control signal applied to J3 is added to (or subtracted from) the reference current, and this leads to linear frequency modulation. Attenuator R13 lets you dial in the amount of modulation desired, while C3 blocks any undesireable DC offsets which would knock the HFVCO into an unuseable region. By the way, this input is the one you would normally use to create vibrato or trill effects.

A few more fine points need to be considered and we'll have the exponential current source under wraps. As mentioned earlier, trimpot R1 sets a precise 1V per octave response (with respect to the input at J1). Unfortunately, this response can droop at the high end, due to the effect of bulk resistance of Q2's base-emitter junction. Trimpot R9, in conjunction with R4 and D1, lets us feed in a correction voltage to counteract this effect. After setting R1 to give a good response in the lower octaves, R9 is then trimmed to give a precise response over the higher octaves as well.

As mentioned above, the reference current source is configured around bi-FET op amp A2. This op amp is one-half of the LF353, and is internally compensated. However, the presence of Q1 in the feedback loop distorts this compensation, and so external capacitor C2 is added.

Finally, notice R5. This "goof-proofing" resistor limits the amount of current flow out of Q2, regardless of any settings of the various control voltage inputs.

That takes care of the exponential current source. After converting the input voltage to an appropriate exponential current, it's easy to then add on the oscillator. Take a look at **Fig. 1** again. One chip and one capacitor; can anything be simpler? Several years ago, it would have been very tedious to come up with an equivalent for this, but thanks to the work of Solid State Micro Technology for Music (the SSM2031's designers), the job is now a breeze.

IC1, the SSM2031, is a high frequency oscillator which has a number of interesting features. First, it has been designed to accept an exponential control current via pin 4. Second, there are complementary outputs available at pins 6 and 7, and these are TTL-compatible. Best of all, the SSM2031 is fast, and is capable of operating up to 10 MHz. The basic operating frequency is set by a capacitor strung across pins 2 and 3. In the circuit of **Fig. 1**, a value of 47 pF is used to set the high end oscillation at about 500 kHz.

So, exponential control is as simple as running the control current from Q2's collector to pin 4 of IC1! Some important details about the control input of the SSM2031 are that pin 4 is a virtual ground that responds to negative current, and that the chip works best with control currents below 5 mA.

Fig. 2 *High Frequency VCO: Parts List.*

Resistors

R1	100 Ohm trim pot
R2	390 Ohm
R3	2k thermistor
R4, R5	10k
R6-R8	100k
R9	100k trim pot
R10-R13	100k linear pot
R14, R15	1M5
R16	3M3

Capacitors

C1	47p
C2	100p
C3	220n

Semiconductors

D1	1N4148 diode or equivalent
Q1/Q2	AD818 NPN matched pair
IC1	SSM2031
IC2	LF353 dual op amp

Miscellaneous

J1 - J3 ¼-inch phone jack, heat sink grease, sockets, wire, solder, front panel, hardware, etc.

Note:

To modify this circuit for use with a bipolar 12V supply, replace R14 and R15 with 1M2 resistors.

And that's all there is to it! Making the High Frequency VCO is easy, too. Your first goal will be to find the parts. R3, the 2k thermistor is manufactured by Tel Labs. The SSM2031 is made by Solid State Micro Technology for Music, Inc. which is willing to supply it in small quantities to hobbyists living outside of California. The AD818 is manufactured by Analog Devices (see DataBank for addresses). All of the other parts are easily found from a variety of mail order houses. **Fig. 2** shows the complete parts list for this project.

I handwired my HFVCO on a Radio Shack breadboard (part # 276-154). This method works fine, as long as you are very careful to keep things as neat as possible. Remember, we're talking about some very small currents and some very high frequencies, so neatness really counts. No rat's nest wiring need apply!

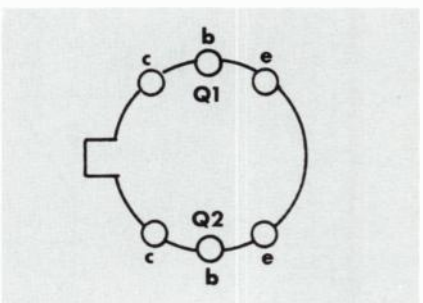

Fig 3. *Pinout of the AD818 (bottom view).*

Pay strict attention to the orientation to the AD818 matched pair (**Fig. 3** shows the pinout). When mounting this part, leave room so that you can position thermistor R3 right on top of it. Before completing the connection of the AD818 or R3, smear some heatsink grease between the two components to insure good thermal tracking.

CALIBRATION

Exactly how you calibrate the device depends somewhat on the circuitry you're driving, but here's one way that works quite well. Since the HFVCO is operating up in the supersonic range, divide the output down using some cascaded frequency dividers (such as the 4024 CMOS binary divider chip), thus making the output audible. Next, turn the high-frequency trim completely down. Patch in your keyboard to the 1V per octave input at J1, and while playing first a low C and then the C one octave above that, adjust trimmer R1 until you hear the appropriate interval. Now turn coarse tuning control R11 up somewhat, thus putting the output into a higher register. While playing octaves again, adjust high frequency

trimmer R9 for the proper interval. Repeat this entire process several times, re-adjusting first one trimpot and then the other, until you reach the desired accuracy. On my unit, I was able to obtain a seven-octave range with no noticeable deviation from true scale, and several more octaves with a minimal amount of detuning.

MODIFICATIONS

This is clearly a building block circuit; it is not really a complete module in itself. Exactly how you use it, of course, depends on the application you have in mind—and there are all sorts of modifications you can enact to adapt the circuit to your own uses. Here are a few tips to guide you. As shown in **Fig. 1**, the HFVCO will work with a standard bipolar 15V power supply. When I built my unit for use in a combo organ (to drive some top octave dividers), I needed to run it from a bipolar 12V supply. It's easy

"...there are all sorts of modifications you can enact to adapt the circuit to your own uses"

to adapt the circuit to this lower value; simply change R14 and R15 to 1M2 resistors, and away you go.

Another change, which might make your wallet happy, involves the thermistors. Thermistors are most commonly found in 1k and 2k values. The one specified here is the 2k variety, but what if you only have a 1k type in your parts box? Well, it can be used for R3, too, but you will have to decrease resistors R6 through R8 to 50k and knock R16 down to 1M5 to compensate. Of course, this change will violate our standard of 100k input impedance, but then again, this might not matter depending on your application.

Many modules don't really need the high precision this circuit has to offer, and if that's the case, here's an easy way to save money. Instead of using a matched pair for Q1 and Q2, simply use to 2N3904 transistors epoxied together. This will still give a certain degree of temperature compensation for a very modest investment. If you feel that the intended environment for your circuit is fairly stable, you could even replace thermistor R3 with an ordinary 2k resistor (use metal film, if possible). It all depends on your needs; use the circuit of **Fig. 1** for best results over all temperatures, or make the above substitutions for an inexpensive version which still works fairly well, but without the ultimate in thermal characteristics.

As promised, next time in "Practical Circuitry," we'll see an interesting application for the High Frequency VCO. Be sure to stop back then!

Acknowledgement: Thanks to Dan Parks, president of Solid State Micro Technology for Music, for being so helpful in keeping me posted on new chip developments and to Ron Dow (also of SSM) for making a number of technical suggestions on the best way to use the SSM2031. EM

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[2] Terry Mikulic, "Exponential Converters," *Electronotes*, Volume 5, Number 37, March 30, 1974, pp. 2-4.

[3] Bernie Hutchins, "High End Compensation of Exponential Converters," *Electronotes Mid Month Letter*, Number 15, March 20, 1978, pp. 1,2 (includes a discussion of Rossum's circuit).

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The X15's output level is just fine—if you like to feed it into a guitar amp! But if you prefer to work in a pro or semi-pro studio environment and have been yearning for a hotter output, have we got a mod for you . . .

Increase Your X15 Output Level

BY CRAIG O'DONNELL

Over the past several years, I've been having a great deal of fun with my Fostex X15. With fellow Chicago Noise Monster member Sean Alias, I've compiled hours of low-cost industrial tech music created entirely on the X15. Along the way, I've learned a lot about getting good sound from the little guy.

For instance, input sources should be the highest possible electronic quality. Whether miked or direct, it's best if they've been limited or compressed a little. For giggles we took the X15 into a pro 16-track studio one night. What a difference! We bypassed the internal mixer and $50 Roland 4-to-1 mixer we usually use. The console's clean high-gain circuitry, the studio's good mics, and the clean monitor mix really made a difference.

In the Monsters' studio we'd found that the monitor section of the X15 is *really* low in output—around -20 dB, to be precise. Our home stereo equipment had trouble giving us a decently loud monitor for overdubbing, playback, and general ear-blasting.

One cheap and dirty way around this is to use the Headphone output as a feed into the Aux on the stereo; use the phone volume control as a preamp level.

If you're track-bouncing to another cassette deck, you might well be using the internal mixer, EQ, and faders in REMIX to go 4-to-2 or 4-to-1. Every bit of output helps you get a clean, noise-free signal.

***Craig O'Donnell** has written about musical electronics and computers for* Illinois Entertainer, Popular Computing, *and* Sound Choice, *as well as served as technical editor for* Option *magazine. A multi-instrumentalist and member of the group* Scientific Americans, *he loves surf music, cheap drum machines, and analog synthesizers.*

Dedicated X15 fans may have noticed that the four individual track outputs are louder than the stereo (or Monitor) outputs. I noticed it too and it began to bug me. How come my $80 Panasonic cassette deck has a hotter output than my $400 semi-pro X15?

A call to Budd Johnson at Fostex did the trick. One of the Japanese engineers, Shinji Sugivra, kindly provided a schematic and a pair of resistors for a simple mod that increases the output of the stereo sends. Changing just two 4.7k resistors to 47k increases the output 10 dB, to -10. Of course, it also voids your warranty, but that's life—just remember that no one but you is liable for any damage as a result of your attempting to do this mod.

"Our home stereo equipment had trouble giving us a decently loud monitor (from the X15) for overdubbing, playback, and general ear-blasting"

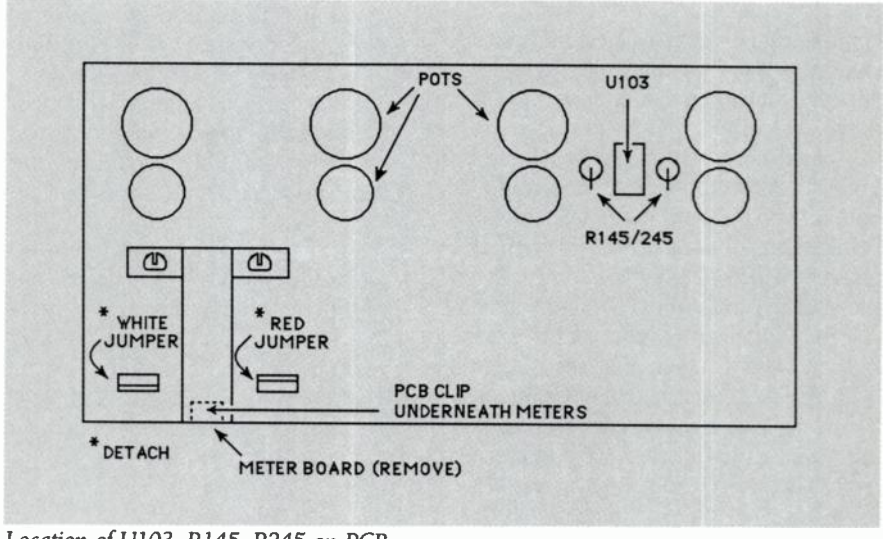
Feedback resistors R145 and R245 between pins 1-2 and 6-7 of IC U103 are replaced with 1% 47k resistors (**Fig. 1**). The mod is easy. Just be sure to go slowly and be careful—Fostex packed a lot of stuff into a tiny box, and we only want to mess with the right things.

First, take a Phillips screwdriver and remove the five bottom screws. Flip the unit over and remove all knobs (Mon/Mix, Pan, EQ, Fader). Slowly work the top off by lifting at the back first and pulling forward.

Remove the meter board (two screws) and pull up gently. Remove the two switch knobs for Mic/Line/Remix. Detach the red and white jumpers; you will now have the printed circuit board staring you in the eyeballs. Locate U103—it's clearly marked with R145 and R245 on either side. Look to the upper right between a pair of pots. Use sharp clippers to remove the two resistors.

Now look at the front where the meter board was; you'll see a plastic clip. Push it and gently lift the circuit board. The board will twist and flop, thus allowing access to the traces. Use a solder-sucker or braid and clean the four holes. Push the new resistors in—gently—and solder each at one end. Using needle-nose pliers, bend the remaining leads 180 degrees, insert them into the circuit board, and solder. This completes the mod.

Slide the circuit board under the two rear mounting bosses and push under the front clip. Re-attach the meter board



Location of U103, R145, R245 on PCB

with the two screws. Replace the switch knobs for Mic/Line/Remix select switch, and re-attach the jumpers.

Carefully replace the top, which will take a few minutes of patient joggling. Replace all knobs, flip the X15 over and put the five screws back in.

Careful! *Do not* begin fussing with other circuit boards in a fit of curiosity. This can disturb the mechanical transport lockout switches, and you don't want to do that. Re-aligning everything can be a big pain—or impossible—and your X15 will spend some time in the shop.

Now you'll have more headroom, especially coming off the Remix setting. The meters are unscaled for the output now (they run way into the red, though without audible problems), but this isn't the big problem you might imagine. The meters are still scaled for the input levels, which is what you really need anyway. Use your *ears* to judge the output level coming from tape; if you hear distortion, it's coming from the playback circuits and not the tape—so back off on the levels. (I am assuming that you recorded the tracks undistorted.) Chances are you're mixing or copying onto another deck with meters anyway—use them. If you're in Remix mode for track bouncing, you'll be going to one of the tape tracks on the X15. Again, the meter indication is no problem; because the meter readings will vary so much depending on the sounds you work with, a hard and fast rule is close to meaningless. Always set levels to be as hot as possible by *tape sound*, not by meter.

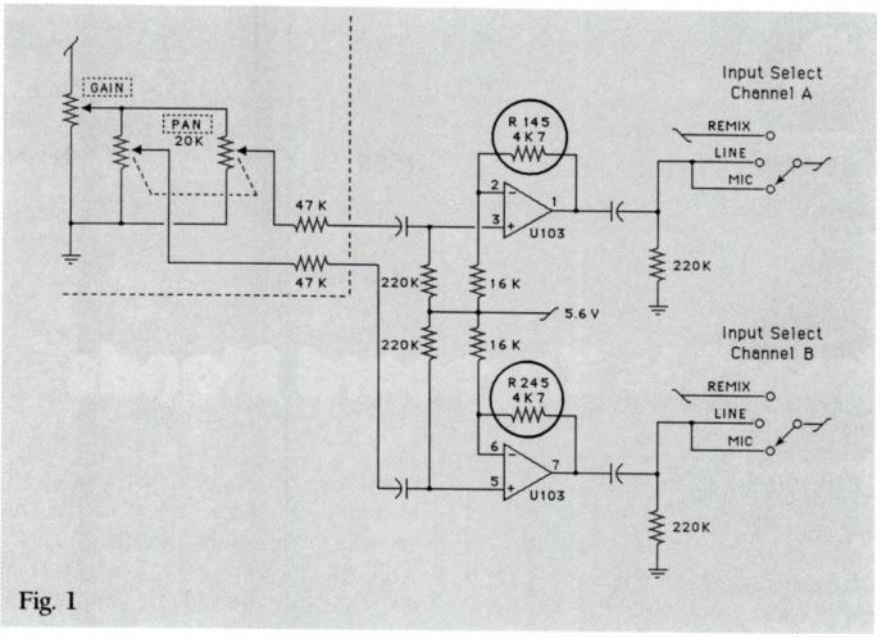


Fig. 1

As with any budget item, you'll have to play around a lot more to find the best results. But this simple mod gives you more output (less noise) on mixdown/bouncing with the internal mixer. It gives you a hotter monitor send (what a relief!). In fact, it's an enhancement I'd like to see on future production models. **EM**

EM Wants Your Problems . . . And Your Solutions

Electronic Musician is calling to all readers for information to be included in an upcoming MIX Publications book on interfacing musical instruments. Any and all questions, problems, solutions, answers, or data pertaining to hooking these instruments together are needed. This would include information about control schemes, such as voltage control and MIDI; timing schemes, such as gate-and-trigger, sync clocks, SMPTE, sync-to-tape; computer interfaces; and electrical considerations, such as connecting "pro" audio gear (as opposed to "semi-pro" or "creative" audio gear) to musical instruments. Data on specific instruments is most helpful, particularly older, out-of-date instruments or extremely new systems. A few examples: processor delay times for individual instruments and how you deal with them, details of the computer interface found on the Yamaha CP-35 electronic piano, a list of inputs and outputs found on the old Oberheim 4-voice synthesizer, etc.

Also of use is information pertaining to instrument pickup systems and their applications, with specific data again being most useful. We cannot guarantee an answer to every question we receive, but we will attempt to answer as many as possible. Please send to the attention of Larry Oppenheimer, MIX Publications, Inc., 2608 Ninth Street, Berkeley, CA 94710.

They're thin and spindly, and hardly look like they could hold up a couple of hefty keyboards . . . but looks can be deceiving. Find out how one musician saw just what she needed in something invisible.

Invisible Keyboard Stands

BY VANESSA ELSE

At first glance, one might imagine that the Invisible™ Keyboard Stands could not possibly hold any significant weight or survive a dramatic performance with you bashing away at your keys, but such a preconceived notion would be misinformed. I can understand why one might think that way—as a child I used to be somewhat anxious when I crossed a bridge, and thought of the tons of trucks and cars being supported by a thin concrete structure suspended above a body of water; I expected to go crashing to my death at any moment. But weird phobias are just not logical. When you think about it, good engineering made both the bridges and these keyboard stands functional, sleek, practical and simply beautiful. What the Invisible stands lack in bulk (which no roving musician needs anyway) is no reflection on their strength.

On stage, the stands are anything but obvious and meld into their surroundings. You can stand behind them and face an audience (if that is your preferred direction) and the stands won't block your view. The stands' visual aesthetics are precisely matched by their ease of assembly and operation.

To assemble the stands, you initially bolt shelf brackets (**Fig. 1**, A) onto the top tubes. Bracket size depends on the depth (front to back) of the instrument they're going to hold. Each bracket requires only one nut for installation since the nut screws directly onto the bottom end of each shelf bracket, which is threaded like a screw. Once bolted in place, the shelf brackets won't need to be moved again unless you want to change the arrangement of your keyboard stack; this means negligible wear on the threads.

These shelf brackets are attached to two top tubes which insert snugly into the two triangular bases (**Fig. 1**, B) like an extension on a vacuum cleaner hose. The posts may be moved up or down within the base (to vary the shelf height) with impossible-to-break height adjuster clips (**Fig. 1**, C).

A thin, steel rod which requires no connecting hardware forms a cross brace at the middle of the keyboard stand (**Fig. 1**, D). Beneath this, two pieces of flexible cable form an "X" between the bases (**Fig. 1**, E). This flexible cable attaches to loops welded to the center tube of the base via keychain ringlets—you know, the kind that you have to pry open with a screwdriver and wind a key around and around until you get it to the center where it never comes loose (unless you deliberately and painstakingly repeat this process in reverse). Once attached, these cables don't ever need to be unattached because when the stands are broken down, the cables collapse like string.

It seems that their function in the design is to maintain a fixed maximum width between the very bottom of the stands along with some complicated physical forces, caused by criss-crossed tension, pulling inward and upward. Anyway, not being a bridge designer, I merely observed that this bottom web pulls taut when the turnbuckle (with attached wire; **Fig. 1**, F) connecting the tops of the tubes is tightened. The bottom quadrilateral is pulled tight with the cross brace acting as a fulcrum.

Now that you have a sturdy three-

Vanessa Else, bored with a life of "sex, drugs, and rock & roll," decided to do something socially positive and joined **EM** *as assistant editor. Presently she plays a variety of musical instruments in* Transmitter, *and in her spare time practices Aikido.*

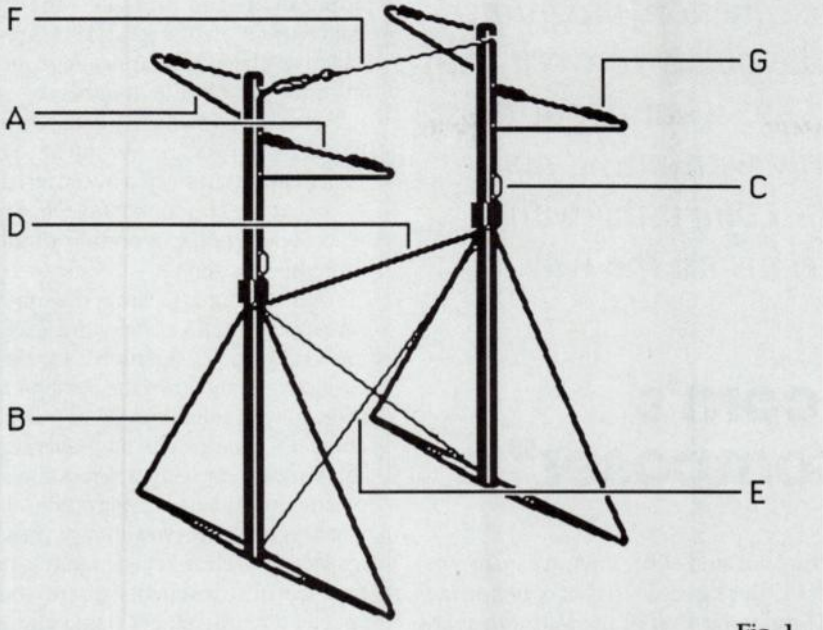


Fig. 1

dimensional object, placing keyboards on it only adds to the stability by exerting a downward force. This lowers the center of gravity; the structure, with its weight underside, keeps the stand from moving sideways when subjected to a reasonable push. Each keyboard is held stationary atop the shelf brackets by four rubber gaskets that slip onto the brackets (**Fig. 1**, G).

The setup of E-mu Emulator II and Oberheim OB-8 that I've been using lately has the OB-8 sitting on the bottom-front tier and the E-II on the upper-back tier. As a result, the OB-8 wiggles a little when I play it, but definitely no more than any other keyboard stand I've tested. Not only that, the E-II doesn't bounce at all because I've situated its keys directly over the stand's center of gravity. This works well because the E-II is a touch sensitive keyboard and no matter how much muscle is used to strike the keys, they're rock solid; they hardly even move!

As you might have gathered, there isn't a lot of connecting hardware with which to "fiddle about." In fact, a stand that holds two keyboards is essentially a total of four nuts, four shelf wires, two bases with attached cables, two upper posts and one turnbuckle with attached cable. That's it! This is why assembly and disassembly are such a breeze.

When an entire stand is disassembled, it is 2½-feet wide, 3¹/₃-feet high and only 4-inches deep. Disassembly is simple: unscrew and remove the turnbuckle, pull out the two upper posts, and remove the cross brace. That's all folks—you then put all the pieces in the bag provided and toss it in your van. However, note that the bag is not a very sturdy one—if you did any heavy duty touring you would need better protection. To set up the stand, all you do is hit the reverse key on your body sequencer and slip on the cross brace, insert the two upper posts, hook the turnbuckle on to one of the posts, and tighten.

Setting them up and taking them down was so much fun (until the novelty wore off) that for a while I kept assembling and reassembling them while marvelling at the ingenuity. Then I would show the process to any visiting friend, but only those that had ever played live understood my excitement.

On the negative side of things, I noticed that the black finish (which is slightly more expensive than the chrome) would probably have a lot of nicks in the paint with even moderate use unless you're very careful—which I don't always have the time to be. Since the silver metal is not simply painted onto the welded steel, it won't wear as quickly. However, I wanted black so it would not reflect light . . . ever reminded that everything's a compromise. I suppose the turnbuckle might show some wear after *a lot* of use, but even so, this item can be found in any hardware store. Two other characteristics which someone might find objectionable is that the keyboards can't be slanted to easily see display readouts, and you can only practically fit two keyboards per stand (though lateral shelf wires, discussed next, can hold a few small units per tier). For this reason, fitting several synthesizers vertically into a small studio or music store display are not really appropriate applications for these stands—but hey, that's because they're optimized for travelling musicians anyway.

"(with) a touch sensitive keyboard . . . no matter how much muscle is used to strike the keys, they're rock solid"

Extra Cost Options

Lateral shelf wires ($24; note that all prices quoted are list prices, and are subject to change without notice) are available for items that are too small to span the width of the stand. I used these on one tier of a small stand to hold an E-mu Drumulator and Electro-Harmonix Instant Replay with drum pad. Each one of the set of two wires lies across the shelf brackets and is secured by snapping clips over rubber gaskets (identical to the rubber gaskets that keep the instruments from sliding around on the shelf wires; **Fig. 1**, G). The space between the lateral shelf wires, which you would set according to the depth of the small instrument, is completely adjustable from zero to 12 inches (long shelf brackets) or seven inches (short shelf brackets). Note that instruments deeper than the length of the shelf brackets will comfortably sit atop them.

Megabrackets ($20) are 16-inch shelf brackets for extra-wide keyboards, mixers, etc. Also, Tee Brackets ($28) are extra-sturdy, 18-inch brackets that will hold a very large keyboard, amplifier, PA cabinet or any very heavy equipment which needs to be centered over the stand. I tell ya, these things are versatile.

A 24-inch wide, two-keyboard stand ($170) includes two sets of long shelf brackets, one set of short shelf brackets and a carry bag. An 18-inch wide, two-keyboard stand ($170—including the same hardware as the 24-inch stand) is for smaller instruments and can also be used as a third tier if placed behind the 24-inch stand. Prices quoted are for nickel finish—add $10 per stand if you want a flat black finish. For more information write: Invisible Products Corp., Box 230, Cambridge, MA 02139. Tel. 617/354-7329.

So far, I've been totally pleased with the Invisible Stands' performance; I only wish mine was as perfect. EM

There are a lot of sequencer programs that also include the ability to work with music notation . . . but what about those composers who deal primarily with music notation? If you're more at home composing with notation than keyboards, help is on the way.

Mark of the Unicorn's "Professional Composer"

BY GEARY YELTON

The ideal music processor is not difficult to imagine. It should be a lightweight, portable computer with a built-in monitor and disk drive. The monitor screen should resemble a sheet of manuscript paper, preferably producing black print on a white background. Transposing parts and changing rhythms should be simple enough to invite experimentation. When your work is complete, it should print out as many copies as you need, exactly as it appears on screen, with the highest degree of resolution possible. Of course, you should also be able to hear what you're writing as it's being written. Such an instrument would be a boon to composers, arrangers, orchestra leaders, band directors, choir conductors, musicologists, students, teachers, and music copyists the world over. Ladies and gentlemen, meet Professional Composer.

Ads for Professional Composer promise to relieve the drudgery of writing out music manuscripts by hand; virtually every standard musical symbol (including lyrics) is capable of being integrated into an attractive hard copy of your final score. Using a mouse makes it all so simple, they say, even a musician that all of the claims are true . . . that's good, because at $495 list, this program isn't cheap.

Professional Composer requires a 512K "Fat Mac" and either an Apple Imagewriter or, for best results, LaserWriter printer. Through no fault of the Mac, the sound generation flexibility is not outstanding at present; there are only three playback tempos, and the tone colors of a scant handful of digital voices are similarly etched in stone. But it's still a powerful tool, and the Macintosh is an ideal computer for such a program.

COMPOSING WITH THE PROGRAM

When you insert the program disk, an image soon appears on the monitor screen which represents a desktop. Three or more icons sit on the desktop: one represents the Professional Composer application itself, one a folder containing the operating system files, and the others are composition files. Point the cursor at the application icon and click the mouse button twice; a few seconds later, a prompt called a "dialog box" appears, requesting information on the number of staves needed. Single, multiple (up to 40), piano, or piano-with-vocal staves can be selected. These can be connected by braces, brackets, or bar lines on command. Once you've made your choice, a portion of a blank score fills the screen.

A menu bar appears at the top of the screen; the menu choices are File, Edit, Basics, Symbols, Variations, Extras, and Groupings. Pull down the Symbols menu (**Fig. 1**) with the mouse, and you have a choice of nine palette bars offering over 100 symbols. You can pick just one or two palettes, or you may display them all. The Symbols menu lets you change clefs; Basics lets you select key signatures, meter, or a list of instruments where you can specify any instrument name and range (**Fig. 2**). The same dialog box also lets you assign one of six tone colors for playback.

After making your various selections, you're ready to enter notes, rests, and bar lines. There are four methods at your disposal, using one or two hands. If you select "Notes," "Rests," and "Barlines" from the Symbols menu, palette bars appear on the screen with the appropriate symbols. Using the mouse, position the insertion point on the staff line or space where you wish to place the note, then click the note on the palette bar with the correct duration. Your choices run from breve to 128th note, plus dots and accidentals. Rests and bar lines are similarly inserted. As you write an entire measure, spacing

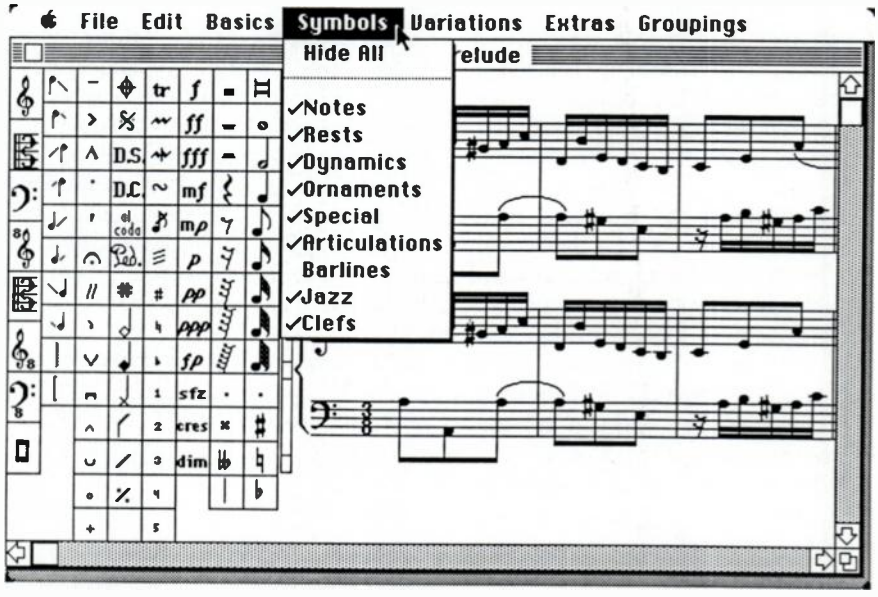


Fig. 1 More than 100 musical symbols may be called up with the Symbols menu.

*After a stint as proofreader for a large printing firm, **Geary Yelton** embarked on a career as a professional musician and part-time writer (he is the author of* The Rock Synthesizer Manual*). In addition to playing in numerous musical groups, he has composed and recorded electronic music for film and video and performed one-man synthesizer concerts.*

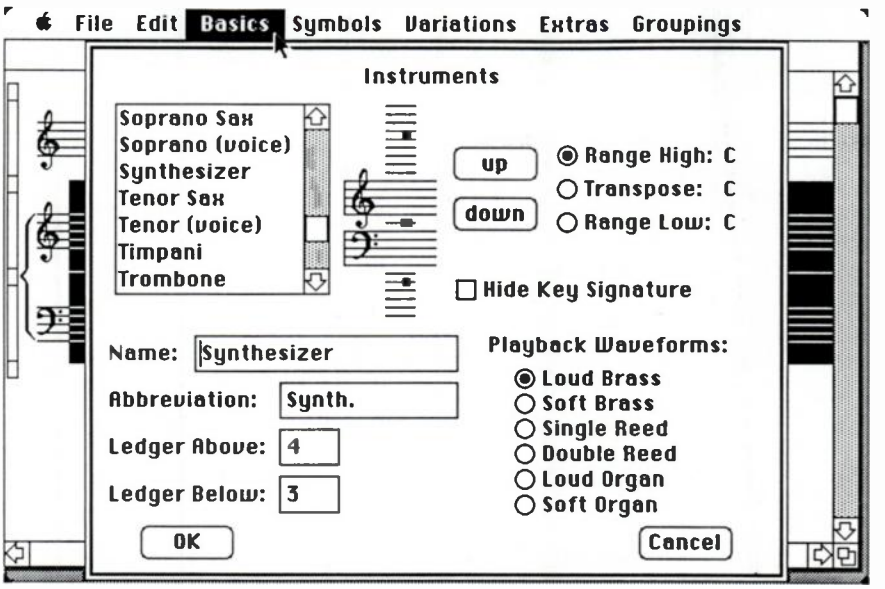


Fig. 2 The Basics menu reveals a list of instrumental assignments, or you may define them yourself.

is automatic; if you make a mistake, just backspace and try again.

The second technique of entering music, using the Mac keyboard in conjunction with the mouse, is a bit faster. A group of 13 keys select note durations, accidentals, dots, rests, and a bar line. To place a symbol on the staff, hold down the proper key as you click the insertion point with the mouse.

You can also enter note data using the Mac keyboard exclusively. Certain keys move the insertion point forward or back, up and down by lines and spaces, or from one staff to another. Another method lets you actually type in your music, using a group of seven keys to represent note names; simply hold the appropriate duration key as you press the desired pitch key.

In addition to symbols representing what is played, Professional Composer lets you notate how it's played. An "Ornaments" palette lets you indicate tremolo, trills, or mordents, and specify fingering. The "Dynamics" palette lets you indicate how loudly sections are to be played, from triple piano to triple forte, and notate crescendos and decrescendos. Other palettes offer fermata, pauses, referential marks, pedal marks, harmonic and percussion notes, chord strokes, accents, and dozens of other symbols. Clearly, the program's authors have accounted for many notational possibilities.

Up to four staves appear on the screen simultaneously. You can scroll through a composition horizontally, or scroll it from the topmost staff to the bottom. Measures can be numbered automatically, and the

"If a part is written for a particular instrument, you can even ask the program to check that a part is within its range"

Mac can be instructed to display any measure by number or rehearsal mark. Individual notes or measures can be "selected" by dragging the cursor across them as you hold down the mouse button. The selected portions are highlighted in white-on-black on the monitor screen. An entire staff is selected by clicking in a "select box" to the left of the staff. Selecting is necessary to use the Groupings menu (Fig. 3). If there's a group of notes with flags that you wish to beam together, just select those notes and choose "Beam." If you make a mistake, choose "Beam" again to toggle the beam off.

Any kind of triplets or tuplets can be created by the same basic method. Professional Composer is outstanding in this area; I know of no existing music software as flexible when it comes to tuplets. Ties and slurs, as well as stem direction, can also be selected.

When a portion of the score is to be repeated, that part is placed within repeat signs. When each repetition requires a different ending, you can select the measure that ends the repeated portion and assign a number to the ending. Unfortunately, repeat signs and multiple endings are ignored on playback.

Lyrics and Other Text

Lyrics, chord names, and other text can be added to score, both above and below each staff. Space can be easily added between staves to accommodate lyrics. As with other Macintosh software, you can choose text style, including various fonts and typesizes, with boldface, italics, or outlined letters. Aligning lyrics with the proper notes requires a bit of practice, however, since text is written in a small box which you position with the mouse. You can also type in information you wish to appear at the top of the first

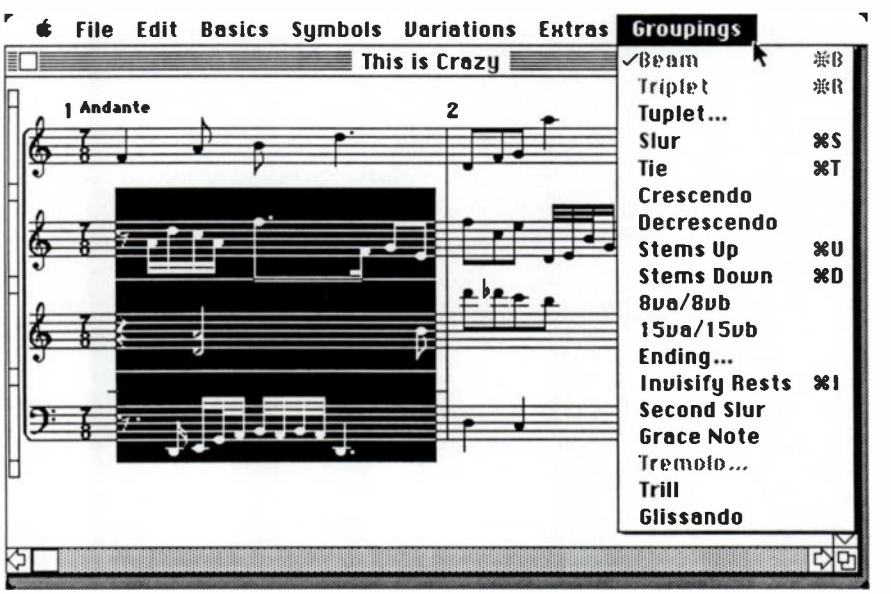


Fig. 3 *Any number of notes or measures may be altered with the Groupings menu.*

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page, such as song title, composer, date, and so on. Headers and/or footers on every page are also possible.

EDITING

Editing your work is where the Mac really shines, and Professional Composer takes full advantage of its capabilities. Using the mouse, you can easily cut or copy any portion of your score and place it in a temporary "clipboard." You can then "paste" it anywhere else in either that score or another. These cut and paste procedures allow extensive experimentation and save a lot of time when transcribing repetitious passages.

"...cut and paste procedures allow extensive experimentation and save a lot of time"

You can add or delete notes, measures, or entire staves anywhere in your score at any time. Individual notes, groups of notes, parts, or whole songs can be transposed to any key or any interval, chromatic or diatonic. The rhythmic duration of any note, rest, measure, part, or a whole score can be halved or doubled as you like, and then automatically rebarred to accommodate the meter on request. Another menu selection automatically checks for rhythmic errors, ensuring that everything "adds up." If a part is written for a particular instrument, you can even ask the program to check that a part is within its range. A group of staves can be merged into one staff for piano reductions and the like, and later unmerged for separating individual melodic lines.

"Another menu selection automatically checks for rhythmic errors, ensuring that everything 'adds up'"

PRINTING

Professional Composer's primary purpose in life is music transcription, and it does this job very well. Teamed with an Imagewriter dot matrix printer, it produces manuscripts that rival any produced by a personal computer program; with a LaserWriter, you can print out scores with a degree of resolution surpassed only by professional music typesetting.

When you choose the Print command, the screen displays the first full page in a reduced form. Every page can be previewed before printing. You can choose the paper size, then print individual parts, pages, or complete scores in either medium- or high-resolution. The print size can even be reduced by half, but this operation eats up a big chunk of disk space.

If you'd like to augment your manuscript with additional graphics or esoteric notation, you may save individual pages of your score as MacPaint documents for further enhancement. (MacPaint is Apple's easy-to-use graphics program upon which most other graphics programs for personal computers seem to be based.)

PLAYBACK

Professional Composer's only disappointing feature is its ability to playback sound only via the Mac's on-board sound generators, which are heard either through the Mac's 2-inch speaker or (by means of a mini-jack on the back) through a larger sound system. Most Mac music software sounds better. Luckily, as of this writing, Mark of the Unicorn will soon be releasing Professional Performer, which in conjunction with any of the "standard" MIDI interfaces (MIDI Conductor et al) will allow you to playback music composed on Professional Composer through a MIDI system (as well as allow for transcribing from any MIDI-equipped keyboard).

The Mac has four voices that can play simultaneously. If you have more than four parts, you can select which four can be heard. Otherwise, the top four staves are played by default. Unfortunately, each staff is assigned only one voice. If you place chords on a staff, only one note per chord will be heard. To hear complete chords, parallel notes must be placed in multiple staves. You then have the option of merging staves to resemble chord notation.

The timbre of individual voices can be changed by selecting a staff and calling up the "Instruments" dialog box. There are loud and soft choices for brass and organ, plus single and double reed. (Don't expect these voices to sound like the instruments for which they're named, but more like a cheap combo organ.) The default waveform is loud brass. Playback tempo can be one of only three speeds. If you indicate tempo changes on the manu-

—page 67

The sequencer software deluge continues with Moog's latest offering. But be forewarned—this sequencer is quite different from the rest. As to whether it takes the right approach for you . . . well, read the review and decide for yourself.

Moog Song Producer

BY JAMES CHANDLER JR.

The Moog Song Producer ($395 list) is a MIDI "package" for Commodore 64 and 128 computers that includes a MIDI interface, five software programs on two disks, and over 300 pages of documentation. First we'll cover the interface, then the software.

SONG PRODUCER INTERFACE

The typical C-64 MIDI interface is a rinky little cartridge with a couple of DIN sockets. Except for basic applications, a MIDI thru box must be used with such an interface in order to provide enough "drive" for multiple synthesizers and to eliminate MIDI "data distortion" (see Alan Campbell's "Making MIDI Work for You" in the June 1985 issue of *EM*). For advanced applications, MIDI filters (to remove excess data that can "clog" the MIDI stream), tape synchronizers, etc. are also required. Soon you have a table full of dumb little $100 boxes, each powered by its own battery eliminator. And an accidental jerk on a MIDI cable can send mechanical shocks directly into the computer via the interface cartridge.

The Song Producer Interface, housed in a rugged 14.5 x 7.5 x 1.5 inch metal chassis, alleviates many of these common C-64 MIDI problems. It connects to the computer with a ribbon cable, so minor tugs on a MIDI cable won't bend the computer's circuit board. Connectors include MIDI In, MIDI Thru, and four independently controllable MIDI Outs. There are also 14 phone jacks comprising eight drum trigger outs, two footswitch inputs (function specified by software), clock in and out (for synchronization to logic circuits or tape), and clock disable in and out (for use with equipment such as Roland drum machines).

The supplied 6-inch ribbon cable allows the Song Producer interface to be placed under the computer monitor; the ¼-inch jacks are difficult to access under and behind the monitor, but in a studio they would probably be connected through a patch bay anyway. Since I keep my disk drives under the monitor, the only other convenient place I could find for the interface was upside down under my C-64. Although the jacks were easy to access in this position, it would be advisable to use Moog's optional 17.5-inch ribbon connector. The 17.5-inch ribbon cable allows the Song Producer interface to fit perfectly atop the portable Executive 64, and permits many placement options for the C-64.

The Song Producer's four independently addressable MIDI outputs eliminate the need for MIDI thru boxes or MIDI filters except for very large synthesizer arrays. Clock in and out works with tape, so Song Producer users probably won't need to buy a tape synchronizer. Owners of older, non-MIDI drum units can make good use of the eight drum trigger outs, as can Simmons drum owners or musicians with modular synthesizers. I modified my Korg KR-55 drum unit to accept computer trigger outputs by wiring a multipin connector to the trigger inputs of the drum generators. This technique should work with most older drum machines.

Moog states that PAiA Digital-to-Analog converter modules can be used with the Song Producer to control synthesizers that respond to one volt per octave control voltage signals rather than MIDI data. One PAiA module could control four synthesizers. Modified Song Producer software is available from Moog to service these additional music channels.

In this reviewer's tests of the hardware, everything worked (however, PAiA DACs were not tested). All tape synchronizers, including the Song Producer, come with the advice "Some experimentation with recording and playback levels may be necessary." In my case, this took most of an afternoon. I found it helpful to record the clock on two tape tracks (or record the clock sequence twice, end to end) as insurance against tape glitches.

Song Producer was designed with an eye toward curing problems related to incompatibility between brands of MIDI synthesizers and incompatibility of non-MIDI equipment. This design approach has only one major drawback, that of software incompatibility. Other C-64 MIDI interfaces, though lacking many of the Song Producer features, are somewhat interchangeable. A large software base for Passport-type interfaces is evolving, and these programs won't work on the Song Producer unless modified by each software supplier. Since software suppliers also usually sell interfaces, they may not see an advantage in porting their programs to the Song Producer. It may be possible to modify the Song Producer to switch to a Passport-emulating mode, thereby eliminating this problem entirely.

On the other hand, working software for several MIDI applications is currently available, most of it already bundled with the interface. The typical user, already having two sequencers supplied with this interface, may not feel a burning need to spend hundreds of dollars for extra sequencers.

SOFTWARE

The software roster includes Songstepper (sequencer program), MIDI Drum Songstepper (sequencer program), Dr. T's Keyboard Controlled Sequencer (sequencer program), MIDI Command (real time keyboard management program) and Sync Command (multiple output clock generator for synchronizing tape, rhythm units, and sequencers). Since all the programs bundled with the Song Producer have long manuals describing gobs of features, we will cover the essence of each program rather than discuss minute operational details.

SONGSTEPPER AND MIDI DRUM SONGSTEPPER

Both programs are identical, with the exception of drum driver features. Song-

James Chandler Jr., though educated in psychology, has been a professional musician for 15 years, doing gigs ranging from honky-tonk solo piano to funk-rock bands and electronic lounge duos. Other activities include electronic repair and design, computer programming, writing, and piano tuning.

stepper will drive eight drum devices from the drum trigger outputs on the interface, but MIDI DRUM Songstepper will additionally drive 16 percussion voices via MIDI.

Songstepper allows 120 independent sequences. Songs are scored from two drum tracks and eight monophonic music tracks. Transposition, voice, channel, and MIDI port can be changed on the fly from within each music track. Editing functions facilitate forming block chords out of multiple monophonic sequences. Drum sequences are entered via drum grid screens that display two measures per screen, so one can see a drum sequence's rhythmic structure at a glance. Music sequences can be created in step mode accompanied by on-screen musical notation, or recorded in real time.

Music sequences don't record velocity, pitch bend, or aftertouch. However, this is consistent with a package featuring on-screen music notation, since traditional music notation does not provide for these parameters either.

Songstepper's organization may appear sensible to non-keyboardists, but I was initially appalled at monophonic-only recording. A jazz player I know complained, "I like the drum grids, but you've got to use up all the music tracks just to play a chord!" After using the program, which appears bug-free, I like it better. I recorded a four-part canon. Using the many features of the score, I was able to select voicings and sequence combinations, to expand four short sequences into a fairly interesting three-minute arrangement. By reviewing the melody lines in various octaves, tone colors and combinations, I discovered there was more substance to the musical themes than I had first realized. The ability to quickly audition many arrangement options can yield a much better song than by simply hacking parts to tape.

Songstepper can also be a learning tool. One can play a part, then see the notes and rhythms on the screen. This program could make a good addition to college electronic music departments; Songstepper, controlling multi-timbral (MIDI mono mode) instruments such as the Casio CZ-101 or SCI Six Trak, could supply more useable musical power than a roomful of old modular gear. In single or duo live performance, all you really need is bass, drums, and a few fill parts. Songstepper is easy to program for this kind of function, and disk access is short enough for live use. The Song Producer package may also be attractive to drummers who want onstage sequencing of Simmons drums and/or drum machines.

Moog is working on a Commodore-128 version of Songstepper. This update is planned to retain the present features of the package, and add polyphonic recording capabilities.

Dr. T's Keyboard Controlled Sequencer

This program is identical to the version available for other interfaces except it is ported for the Moog interface. If you are interested in a detailed review, refer to Charles Williamson's article in the February issue of *Electronic Musician*. The only comments I would add is that the Dr. T sequencer writes identical data to all four MIDI Outs, performing as if it were connected to a single-output interface and a four-output MIDI thru box. Memorymoog Plus, Prophet-5, and other omni-only synthesizers will still require MIDI filters until a planned software update becomes available.

A musician I know has been using the Dr. T sequencer for bass, drums, and backup parts in a solo act. Tempo and synthesizer setups are preset for each song, so all she must do onstage is load and play. The software allows her to give fat renditions of Madonna and other MTV-type material. She tested the Song Producer onstage for a week; all her material, composed on another interface, functioned flawlessly on the Moog. Though initially frustrated by this program, she now loves it.

MIDI Command

MIDI COMMAND is a real time synthesizer management program. It performs functions similar to those available from top-of-the-line keyboard controllers or dedicated computers such as the Yamaha QX-1. A master keyboard controls four or more synthesizers connected to the Song Producer MIDI outputs. Pitch bend, velocity, and all other performance features are supported.

The computer holds 100 pages (a page is one video screen) of control information, which can be saved to disk. Each page allows each MIDI output to be programmed to respond to a zone of the control keyboard, with each zone having specifiable top and bottom end points.

Sequencer Software Basics

Computers are totally useless without software. For readers who have not had hands-on experience with MIDI computers, here are a few general observations about MIDI sequencer software:

Multi-track tape recorder emulations are the easiest programs to learn. A song is made up of parallel tracks; more advanced recorder emulations allow several sequences made up of parallel tracks. The multi-track sequences are serially arranged (scored) to form a song. There are also score-oriented programs which allow many discrete sequences of variable length. Sequences are scored horizontally and vertically for virtually unlimited control of a song's arrangement and orchestration.

On a small computer such as the C-64, one can easily run out of memory using a tape recorder emulation. Disk access time for loading a "full" song can be one minute or more, and one disk will only store five or six such songs. For serious linear recording on a C-64, one must either assign drums and perhaps some other instruments to external sequencers, or use several tracks of sync-to-tape.

Score-oriented sequencer programs are memory efficient and minimize disk access time, since a pattern which is played 30 times in a song need only be recorded once. A complete song constructed in this fashion can load from disk in 15 seconds or less. This piecemeal compositional method is useful in modern popular music genres and classical baroque styles. In jazz or classical beyond the baroque, scored sequences are less useful, because the music is so subtle it rarely repeats exactly.

Interestingly, every one of the several musicians I know with MIDI computer systems experienced dissatisfaction and frustration while learning to use the system. Music is such a complicated art that a good sequencer program *must* have numerous commands and functions, which correspondingly means that you must spend some time getting to learn the sequencer's subtleties. Incidentally, I think sequencer programs probably reflect the program designer's compositional technique; learning to make someone else's ideal sequencer work for you can be time-consuming.

Thus, as you play on a master keyboard, playing in certain zones will play certain slave keyboards—bass on the lowest octave, perhaps, with some other sounds occurring at other places on the keyboard. Zones can also be overlapped for layering effects, enabled, or disabled. A transposition feature on each output allows one to play high synth parts from low keyboard zones, or vice-versa.

Each page holds eight *quadsets*. A quadset is a line containing voice numbers for the four outputs. One can step forward or backward through quadsets (voice changes on all instruments) or pages (new keyboard zones, plus eight more quadsets) using the computer function keys or footswitches. Twenty-six chains of pages can be programmed, allowing footswitch control of incredibly detailed sequences of instant program changes and keyboard controller re-definition. For concert-grade performances of high-tech material, all this power looks good. I remember when I had to tweak knobs on three mono synths between each song...I thought I'd gone to heaven when programmable synths hit the market.

Sync Command

This program provides nine clock outputs for sync-to-tape applications. The pulses-per-quarter note of each output can be specified, and clocks can be individually advanced or retarded to overcome response delays of some devices. I haven't had a chance to test this program, but since all the other programs work as advertised, I have no reason to suspect this one. Musicians owning many incompatible sequencers may find this program valuable.

Third Party Software

Patch librarian programs for Yamaha and Casio are available from Dr. T. Moog sells Setplayer, which holds eight to 15 Songstepper songs in memory and plays a set unencumbered by disk access waits. Updates and versions for the C-128 will supposedly be available in the future.

Conclusions

This package works: IBM MIDI systems outclass it, but money buys performance. It looks especially useful to users of non-MIDI drum machines, omni-only synthesizers, and CV synthesizers—in other words, those who might have a lot of existing gear around and don't want to re-invest in a totally new setup. While Song Producer may not be for everybody, for some musicians it will be just what the doctor ordered. EM

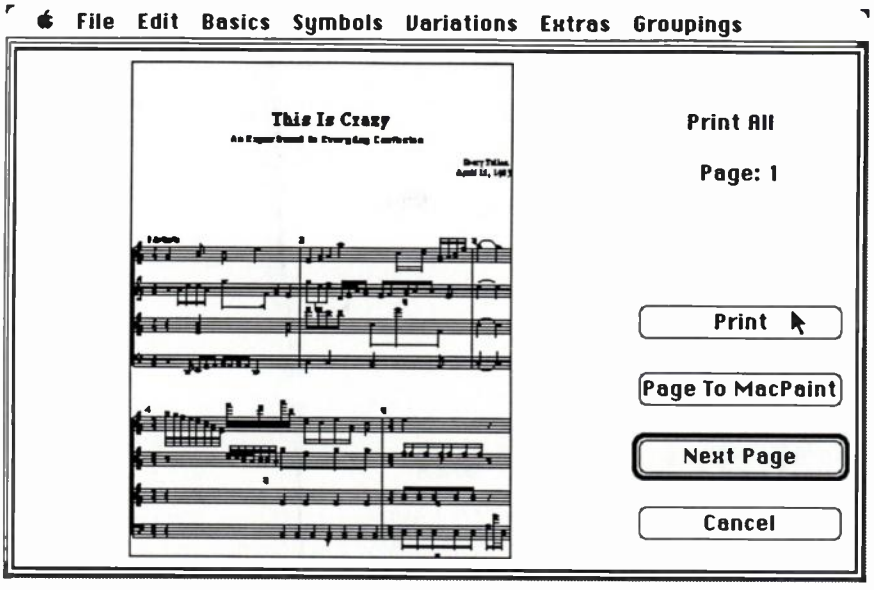


Fig. 4 *The entire score may be previewed before printing.*

—from page 64, Professional Composer

script, they have no effect on how fast the music is played by the computer. Likewise, dynamic markings, glissandos, vibratos, and the like are ignored on playback.

Conclusion

The original release of Professional Composer had a few problems, but they've apparently been ironed out completely. It's a shame you still can't hear chords as they're normally written—perhaps that problem will be solved in the future. The application, along with the system folder, takes up almost the entire 400K microfloppy, leaving little disk space for files and printing operations. This situation makes a second disk drive absolutely necessary.

Professional Composer is very attractively packaged. The documentation is adequate, though the opening tutorial may be confusing at first. Most of the text is easy to read and understand, but it could be a little more comprehensive. Some features are omitted from the documentation altogether, and others are mentioned only briefly.

Mark of the Unicorn has done an admirable job of adhering to the features of the Macintosh interface, with a few deviations. Pull-down menus and palette bars abound. Most operations can be performed with the mouse, and many of those can also be handled by command-key combinations. One unwelcome variation is the undo command. Most Macintosh applications allow you to "take back" your most recent operation by selecting "Undo." Professional Composer's "Undo" command only affects backspace, cut, and erase commands. If you transpose a part and then change your mind, you must re-transpose it back to its original key. When you try to undo any command other than backspace, cut, or erase, whatever was last deleted will appear at the insertion point.

For those who are extremely adept at transcribing music by hand, Professional Composer may not save a lot of time. On the other hand, unless you're extremely neat, the computer-generated manuscript will probably look better. If your transcribing skills are no better than good, Professional Composer can be a real time-saver and can have a tremendous influence on the way you compose. It's likely that some people will buy a Macintosh just to use this software. The ability to print multiple copies of individual parts by hand takes a long time, but it's a snap with Professional Composer. Just about every musical symbol you'll ever need is at your beck and call, but if you want to add your own specialized symbols, you can always augment your score in MacPaint. The folks at Mark of the Unicorn have been extremely thorough.

Once the MIDI software becomes available (probably by the time you read this), it won't matter that Professional Composer doesn't take full advantage of the Macintosh's timbre-producing flexibility, because sounds will come from MIDIed musical instruments. If you're an electronic musician, you may want to wait and check out the complete package. But if you spend many hours of your life transcribing music by hand and can pay the price, you owe it to yourself to check out the Apple Macintosh and Professional Composer. EM

Tired of cassette interfaces that take forever to save and load programs, or RAM cartridges that cost an arm and a leg? Saving patches on diskette is the way to go . . . but it takes the right Patch Librarian.

Opcode's MIDIMAC Patch Librarian

BY GEARY YELTON

Do you like the idea of being able to carry hundreds of synthesizer patches in your shirt pocket on a little 3.5-inch disk? With an Apple Macintosh, MIDI interface, and Opcode Systems' MIDIMAC Patch Librarian, you can permanently store and transfer synthesizer patches back and forth from synthesizer to computer with the greatest of ease and speed. (No more waiting for cassettes to load and dump patch programs!) The MIDIMAC Librarian can rearrange up to four banks of patches at the same time, then print out charts of which patches occupy which memory locations. You can even play your synthesizer from the Mac keyboard. Opcode makes Patch Librarians for the Yamaha DX7, TX7, TX816, DX5, DX21, RX11, and RX15, the Casio CZ-101, -1000, and -5000, Oberheim's OB-8, Xpander, and Matrix-12, Roland's Juno-106 and JX-8P, and the Chroma Polaris, with more on the horizon. One Patch Librarian may be combined with others so that you can transfer information to and from several instruments with the same program. For this review, I tested the Polaris librarian.

MIDIMAC software is compatible with MIDI interfaces from virtually any manufacturer. Likewise, Opcode's interface is compatible with virtually any MIDI software for the Mac. When you call up Patch Librarian, you can choose an interface frequency of 500 kHz, 1 MHz, or 2 MHz; you can also indicate whether the interface is connected to the Mac's modem port, the printer port, or both. This dialog box always appears when you open the application, and remembers the settings from the last time the program was used.

Geary Yelton is a professional musician and part-time writer (he is author of the recently-updated Rock Synthesizer Manual, *published by Rock Tech Productions).*

When you okay your choices, a second dialog box reminds you to be sure the Polaris is listening on the specified MIDI channel.

PATCH BANKS

For the Polaris, a patch bank appears as a Macintosh "window" with four *columns* (A, B, C, and D, each with a row of *fields* (patches) numbered one through 12. These columns correspond to four of the 11 Polaris banks of 12 patches each. The additional columns (E-K) can be accessed via scrolling, as only four columns can be seen at a time. When you click the mouse while the cursor is on a field, it becomes the selected patch. You can also step sequentially through a list of patches, or select the patch immediately to the right of the currently-selected patch. The selected patch's name appears in an edit box at the top of the window. "Enter Names," a choice in the Edit menu, lets you type in patch names of up to 16 characters. Because the Polaris doesn't memorize patch names, this information is not transferred to or from the Polaris; but some synthesizers supported by MIDIMAC Patch Librarians can store and display patch names.

The Polaris is always ready to send and receive patch information to and from the Librarian. One hundred thirty-two programs can be transferred from synth to Mac, or Mac to synth, in seconds. Also, any patch currently in the synthesizer's edit buffer (a "holding zone" that contains a single patch) can be sent to the Mac, and any individual patch in the Mac can be transferred to the Polaris' edit buffer. Once in the buffer, the Polaris "Store Program" function stores the patch wherever you like.

The "Send on Select" function also sends single patches to the synthesizer. When selected, pointing to a bank patch automatically sends it to the Polaris' edit buffer. This feature is great for auditioning sounds before storing them, or for bypassing the synthesizer's memory entirely by relying on the Mac for patch retrieval. You can zip through a list of piano sounds, for instance, listening to each one, then store the one that's right for the job.

THE MIDI MENU

A MIDI menu selects transmission channels, which provides an unexpected bonus for Polaris users. The Polaris can play eight different patches at once, each of them polyphonic (subject to its six-voice limitation). In other words, each of eight channels transmitted by a sequencer or MIDI controller can control a different patch, although no more than six voices will sound at once. Oddly enough, none of the Polaris advertising mentions that it can be fully multi-timbral under MIDI control; the manual tells you how to enable the extra channels, but not why. I discovered this well-kept secret in the Polaris MIDI document, so this just goes to show that it pays to read all the documentation you can get for your instrument. To create eight "virtual instruments" (i.e. be multi-timbral), you must transmit patch information on eight different MIDI channels. This is easily done using the Patch Librarian's MIDI menu: Assign one patch to channel 1, another to channel 2, and so on. MIDI channel selection also lets you transmit patch information to particular synthesizers without confusing other instruments in the same network. If 16 channels aren't enough, you can use two interface boxes (connected to the modem and printer ports) and switch between them.

PATCH LIBRARIES

In addition to banks, the Patch Librarian displays another type of window called a *library*. This is a list of practically any number of patches arranged alphabetically rather than in banks. Up to 21 patch names appear in a library window simultaneously; a scroll bar moves through the rest of the list. The Polaris library included with the disk contains most of the patches from Volumes 1 and 2 of the Fender-programmed Polaris data cassettes, plus a scant handful of good-to-mediocre original patches.

Rather than sending patch banks to the synth, a libary can only send individual patches by the "Send on Select" option. Selected sounds can be heard without ever touching the Polaris itself—"Test Sounds" lets you play the synthesizer from the Mac keyboard. Twenty-one letter keys correspond to three octaves of musical keys with no sharps or flats, and nine

number keys preset the Note On velocities. This feature is especially useful when working with synthesizer expanders that have no keyboard.

Copying and Pasting

Patches can't be sent directly from the synth to the library, but are instead copied from bank windows. Up to four windows of banks and/or libraries may be on screen at the same time. Patches are freely copied from any field and pasted into another using the usual Macintosh copy and paste techniques. Any number of patches may be selected in a single window; for example, you could select a dozen patches anywhere in a bank and paste them into a library. New banks may be assembled by pasting patches from libraries and/or from other banks.

When you paste patches into a library, they're alphabetized automatically. If the library already contains patches with names identical to the ones being pasted, a dialog box appears, asking if you'd like to replace all of them, replace them individually, or rename each new patch. Adhering to the Macintosh user interface, any edit function can be "taken back" with the Undo command.

Patches can be removed from a library, but unfortunately, not from banks. Opcode's documentation says that this is because empty patches are not useful in banks, but with the Polaris, leaving empty slots frees up memory which can be used by the on-board sequencer.

Printing

The Print command prints the contents of the currently-selected bank or library on a Mac-compatible printer. A bank prints out in rows and columns, and a library prints out as an alphabetized list on a single page. Printed banks are extremely handy for reference when the current bank is not on screen, when the Mac is busy with other musical duties (like sequencing), or when the computer is at home, sitting on your desk, and you're somewhere far away.

Combining Applications

Two or more MIDIMAC Patch Librarians may be merged into one program, capable of sending patches to and from different kinds of synthesizers. You must have Patch Librarians for each synth you intend to use, and they must be the same level of revision. By the way, Opcode offers free upgrades by modem, or for a small charge by mail. . . pretty civilized of them, I'd say.

According to the documentation, there are two ways to combine Librarians. The first method requires opening the Librarian and selecting "Install" from the File menu. The documentation says this method is foolproof, but when I tried it, I just swapped disks back and forth endlessly with no results. Fortunately, the second method was successful. Other Patch Librarian disks include an installation utility which you copy to the Polaris disk, then open; a few seconds later, it's installed. You can then open bank or library windows for either synth on the same screen. For some reason, the Polaris disk didn't include an application for installing it to other Librarians.

The well-written Librarian documentation says that you can use the Patch Librarian with the MIDIMAC Sequencer using Apple's Switcher, but when I tried this, the system bombed whenever I tried to open a Polaris bank. I'm sure this bug will be corrected by the time you read about it. When I tried switching the Sequencer with the DX/TX Patch Librarian, however, everything worked just fine.

Conclusions

I'm quite impressed with all the MIDIMAC products I've seen, and Patch Librarian is no exception. When I first received it, though, it didn't work very well. Patch transfers were successful only about one time in ten, if at all. A call to Opcode revealed the source of the problem. The Polaris Librarian is designed for use with later Polaris revisions. Fender's latest ROM upgrade was Rev. 9, and my year-old synthesizer contained Rev. 3. When I replaced the EPROMs, everything worked almost perfectly.

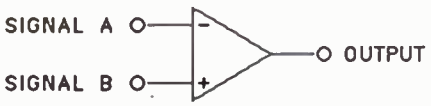
It's very useful to be able to transfer patches back and forth instantly from synth to computer, just by pointing the mouse and clicking. It's also convenient to have a choice of MIDI interfaces—buying a different interface device for every piece of MIDI software is expensive and unnecessary. If you have more than one supported synthesizer, the ability to combine Patch Librarians is no less than wonderful.

You can make back-ups of the MIDIMAC software, so don't worry about being stuck on stage or in the studio with no way to retrieve all those valuable sounds you struggled to collect and perfect. Can you imagine being in a performance situation with your one and only patch disk and it doesn't work? Nightmare City. As of this writing, Opcode's DX/TX Librarian includes over 400 patch programs, many of them quite excellent. Despite a few very minor flaws, at $50 the MIDIMAC Patch Libraran from Opcode Systems is an all-around good buy. EM

Analog Devices, P.O. Box 280, Norwood, MA 02062.

Balanced line: A balanced line comprises of three wires, consisting of two hot conductors and one ground; the two signals sent on each conductor are identical except for being 180 degrees out-of-phase with respect to each other. Balanced line connections minimize the effects of common-mode noise (hum, RFI, motor "hash," etc.) induced into both hot conductors in long cable runs by virtue of terminating in a *differential amplifier*. The differential amplifier is configured so that at its inputs, the out-of-phase audio signal (which has noise characteristics similar to the in-phase signal) arrives at the inverting input of the amp and is inverted. This inverts the audio signal so it is now in-phase with the other audio signal, while the noise signal now becomes out-of-phase with the other signal. When combined, the noise cancels out and the audio signals add together, thus producing a strong output with minimum common-mode noise. Audio balanced line systems usually use XLR or stereo ¼-inch phone connectors.

Differential amplifier:

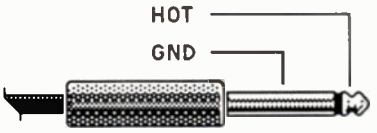


International Parts Specification Standard: This standard avoids the unnecessary repetition of zeroes, decimal points, and stating Ohms or Farads where it is implicitly understood. It is widely accepted in the international community and EM would like to bring it home. Following are some examples.

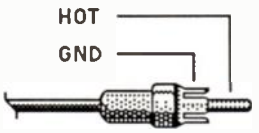
USA	Int'l
1k	1k
1.5k	1k5
2.2M	2M2
1μF	1μ
0.01μF	10n
3300pF	3n3
0.00228μF	2n2
10pF	10p

where
k = 10^3 Ohms
M = 10^6 Ohms
μ = 10^{-6} Farads
n = 10^{-9} Farads
p = 10^{-12} Farads

Mono ¼-inch phone jack:

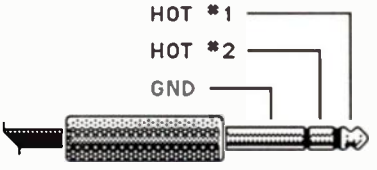


RCA phono jack:



Solid State Micro Technology for Music, Inc., 2076 B Walsh Avenue, Santa Clara, CA 95050.

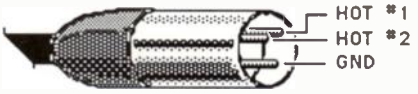
Stereo ¼-inch phone jack:



Tel Labs, P.O. Box 375, Londonderry, NH 03053

Unbalanced line: Unlike three-line balanced line systems (see above), two-line unbalanced systems consist solely of one hot conductor and ground. Unbalanced line exhibits no noise cancelling properties like the balanced line systems, but is simpler and less expensive to implement, thus making it preferable for some applications. Unbalanced lines generally use mono ¼-inch phone connectors and RCA phono connectors.

XLR jack:



prophet 2000

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