Reviews: Mac and Amiga Software, Rane MPE 14 MIDI EQ ■ Make Hard Disks *Work!*

Electronic Musician

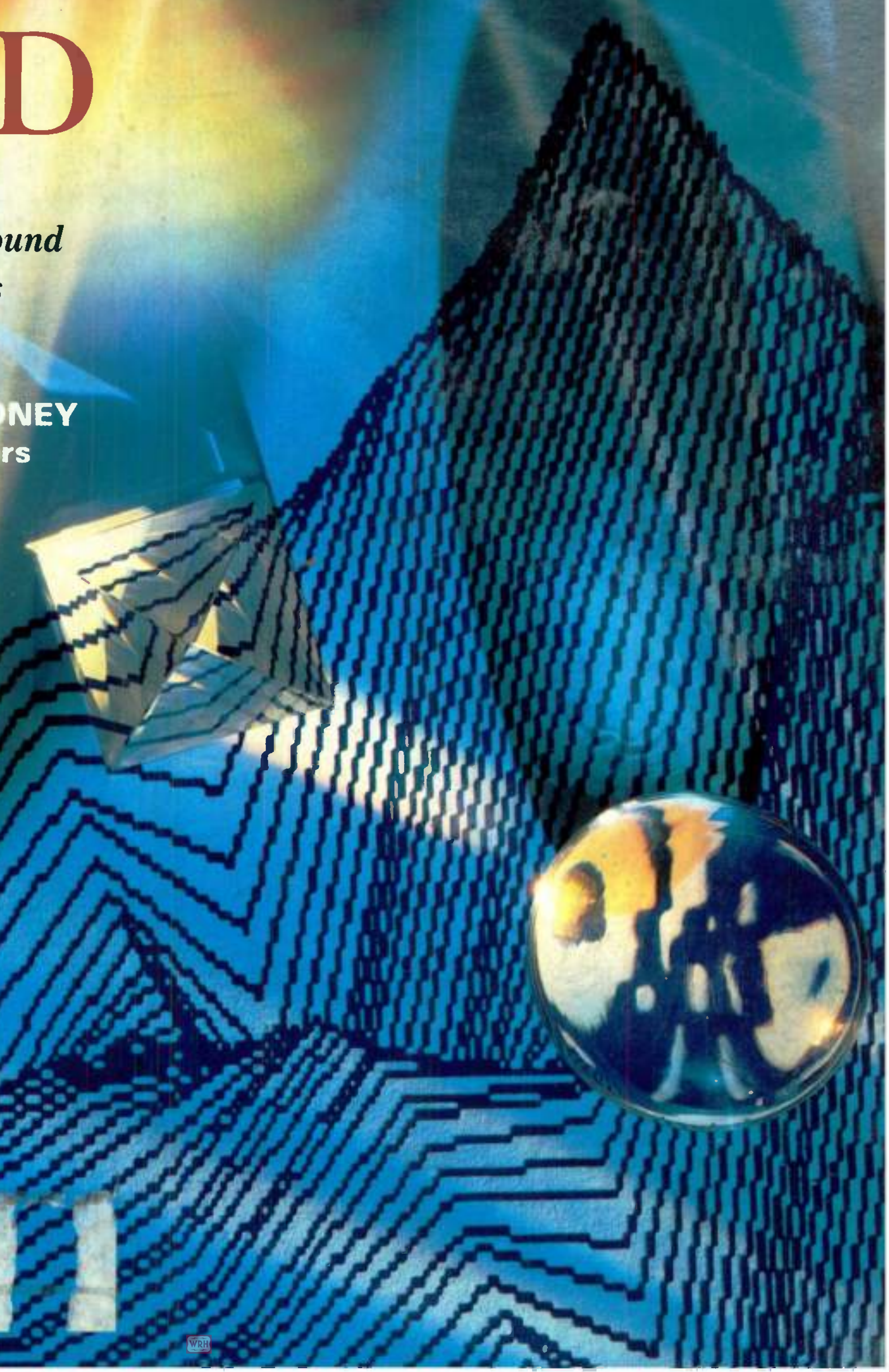
U.S. $3.50 Canada $4.50
February 1990

All About SOUND

The Essentials of Sound
Sound as Music, Music as Sound
New Methods of Synthesis

MORE INPUTS FOR LESS MONEY
Build Your Own Line Mixers

MIXERS DEMYSTIFIED
Choosing the Right One for Your Studio



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Electronic Musician

AN ACT III PUBLICATION

FEBRUARY 1990 VOL. 6, NO. 2

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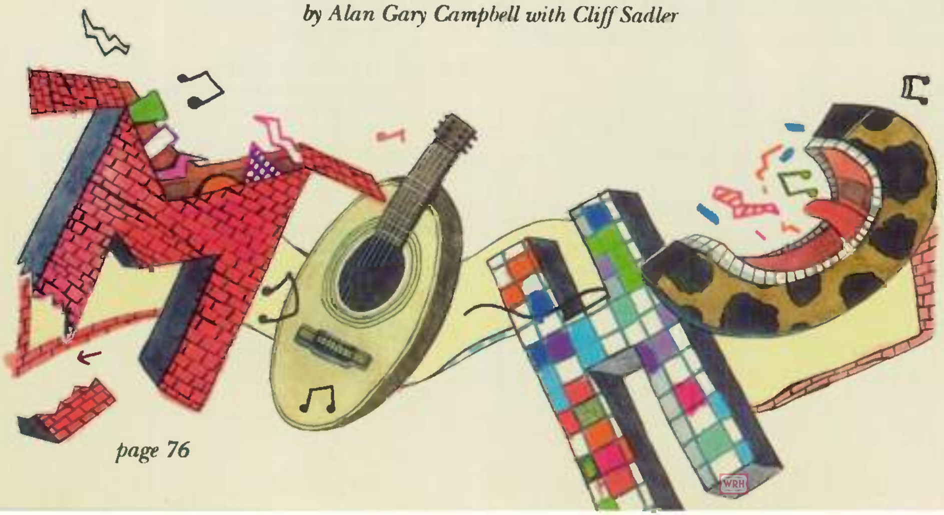

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Cover: Photograph by Pierre-Yves Goavec.

Electronic Musician® is published at 6400 Hollis St. #12, Emeryville, CA 94608, and is ©1990 by NBB Acquisitions, Inc. This is Volume 6, Number 2, February 1990. **Electronic Musician** (ISSN: 0884-4720) is published monthly. Second Class postage paid at Oakland, CA, and additional mailing offices.

The Wonder of Sound

Taking a step back and looking at sound from a new perspective could lead to an important reassessment of your music.

Though we all express it in different ways, I believe the desire to think and to learn is an essential part of human nature. Particularly if you hold any notion of yourself as an artist (we are musicians, after all), the urge to grow and expand your *self* is a strong one.

With that in mind, I offer you this thought to consider: The path to learning and growth begins with a very simple but critical step—a sense of wonder.

It's not easy to take this step, particularly because it seems like so much of what we do and learn about in our society is based on straight, objective facts. But the need to think deeply about a subject and consider its ramifications and connections to other ideas and concepts is essential to development as a human being and as an artist.

The process is not as removed from your current experiences as you may think. The sense of awe and wonder with which children greet new topics or new experiences, for example, is fascinating. If you've ever seen the satisfied glow on a child's face when he or she first walks into a amusement park, sees a museum's dinosaur exhibit, or plays a synthesizer, you'll no doubt agree that it's one of those incredibly special states of mind of which it seems harder and harder to partake as we grow older.

As musical artists, we're challenged to rediscover that sense of wonder and somehow share it through our works. To put it mildly, this is easier said than done. Particularly as electronic musicians who put together complex music using complex, confusing instruments and tools, it's easy to get overwhelmed by the minutiae of technical details and forget what we're really trying to do: create satisfying music with some pretty amazing instruments.

The first step toward sharing the wonder is perceiving the wonder inherent in the elements with which you work. As with any artistic discipline, if you're the least bit interested in improving your skills, you need to understand and appreciate the fundamental building blocks of the subject matter—in this case, sound.

Sound is the core around which our musical and technical efforts are based, and through a more detailed examination of it we can achieve a deeper understanding of the relationship between the tools we use and the music we make. Have you ever considered, for example, that a chord could be thought of as a timbre, or that a synthesized timbre could be a chord? The connection between music and sound is of tremendous importance for musical growth.

In addition to having an impact on our current musical concepts, I believe new methods of thinking about sound will also have a huge influence on musicians in the '90s. As we start to deal with integrated digital systems, more complex synthesis methods, and more sound-based forms of music, a thorough knowledge of sound and all its wonders will be invaluable.

This month's issue of **EM** is focused on steering you in the right direction. An article on harmonics, the basic elements of sound, provides the necessary background information for an exploration of new methods of creating sounds and for a philosophical article on the nature of sound, music, and technology.

The time you spend—as a composer, arranger, and/or performer—wondering about sound will reward you through profound reexaminations of your compositional methods or subtle changes in existing works. In either case, you'll be growing as an electronic musician, and isn't that what we're all trying to achieve?

Bob O'Donnell

Electronic Musician

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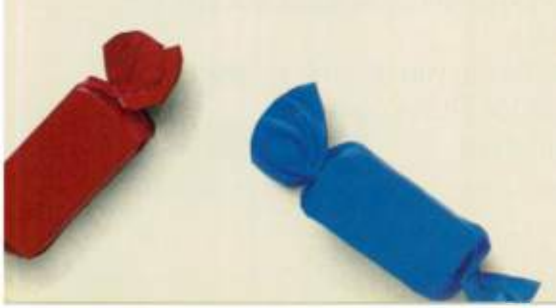

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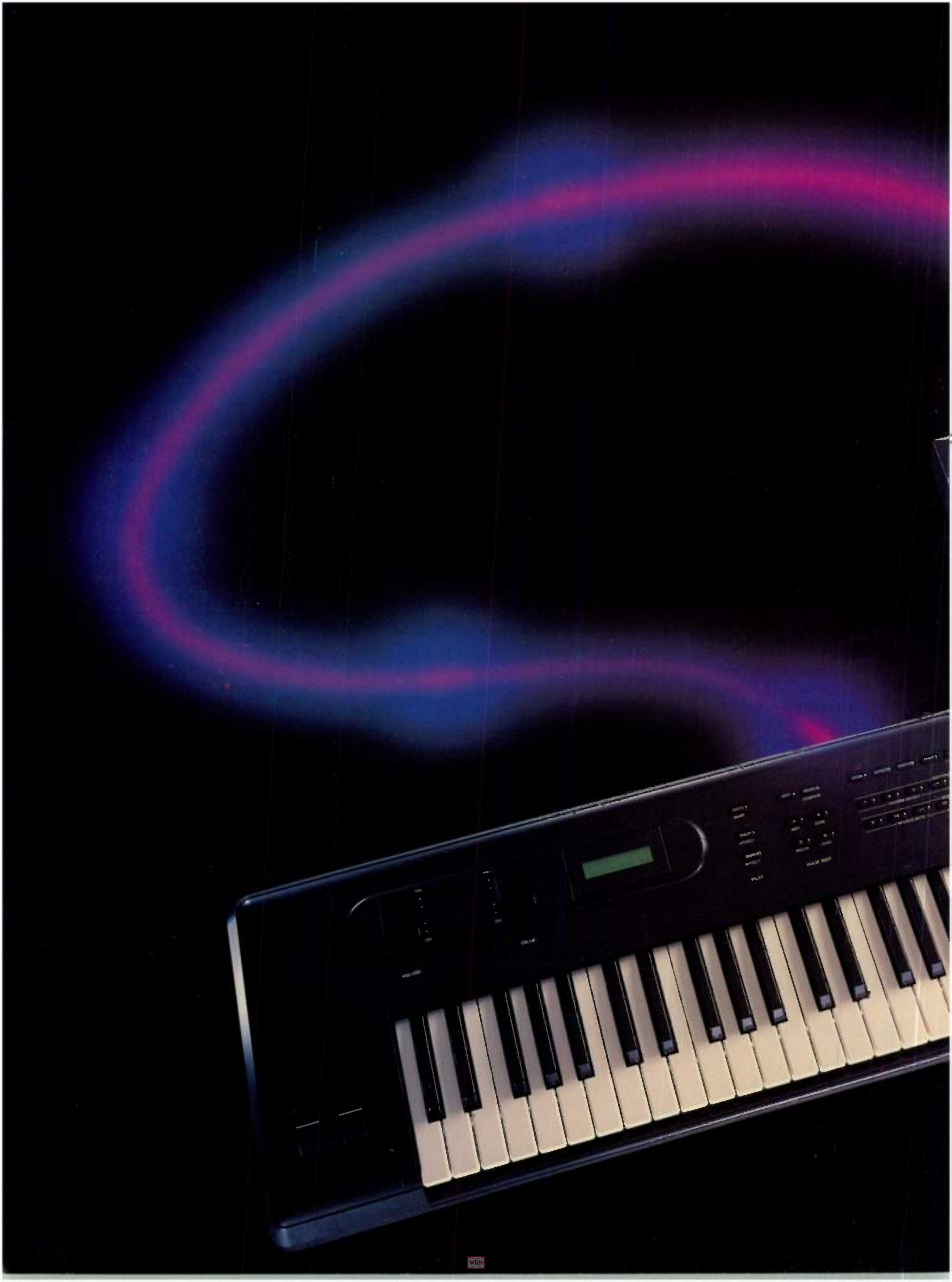

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Sometimes, the whole is greater than the sum of the parts. Consider the K4 and Q80. When combined, they offer not only more power and performance than any stand-alone workstation, but at a price substantially less than most.

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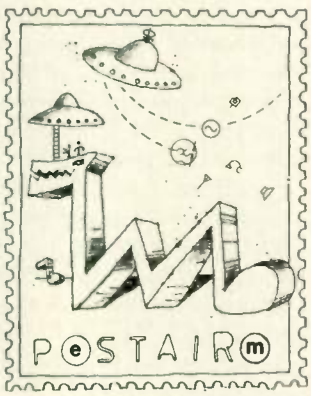
A few corrections and clarifications lead off this month's collection of musings from our readers.



WOMEN, BE WISE

As an active consumer of electronic software and hardware, I have been appalled by some of the tasteless and demeaning advertisements used to promote the sale of computers, software, and electronic equipment. After reading Marsha Vdovin's November 1989 guest editorial, I took a careful look at the ads in your November publication. I was saddened to find only two ads (Fostex and Audio Institute of America) depicting women as serious users of electronic equipment.

Most ads showed a product by itself, or they depicted men as dignified patrons of the advertisers' products. In one of the few ads, with the exception of those mentioned, where a woman was used, a woman vocalist is seen bumping her sexy rear end up against a guitar player. There is an ad with a drawing of a grandmother taking orders for professional equipment. Another ad shows a woman as a magical being playing three keyboards; it certainly gives the impression that only a woman who's a sorceress, no less, can use computerized hardware. These ads fail to represent women as important consumers of electronic equipment. There are ads appearing in other publications that are far worse. One such ad depicts a woman as half machine and half sexpot. Any company that portrays a woman in such a fashion does not deserve patronage, no matter how good the product.

Perhaps if more companies utilized women respectfully and showed women as serious users of their products, more women would be inspired to break into this exciting musical field. It would also be helpful if **EM** featured and interviewed women musicians on a regular basis. Women are waiting to be reckoned with as a potential consumer force.

Teresa Rivera
California

Teresa—Thanks for the input; we knew there had to be more women out there who feel the way Marsha does. As for your suggestion on interviewing female musicians, check out last month's issue and Deborah Parisi's article "Into the 21st Century: New Ways of Making Music," which includes a section on Fiorella Terenzi. It's at least a start.—Bob O'D

SETTING THE RECORD STRAIGHT

With regard to E. Tomlins's letter (November 1989 "Letters") questioning **EM**'s editorial integrity in having me review the Yamaha V50, allow me to clear up two misconceptions. First of all, the Center for Electronic Music (CEM) is not a "synthesizer school," as Mr. Tomlins states. We are a nonprofit organization that offers a variety of services to the public at large as well as programs that use music technology to work with disadvantaged youngsters and with disabled individuals. Second, we are very proud of the fact that we receive equipment support from a very broad segment of the industry (currently over 65 manufacturers and counting). The fact that we receive this kind of support is a wonderful reflection upon the charitable spirit in this industry but in no way compromises our objectivity and neutrality. I try to bring the same objectivity to my writing, and I can assure Mr. Tomlins that neither Yamaha, nor any other manufacturer, has ever tried to exert any influence over what I write.

Thanks for giving me the opportunity to clear up these points.

Howard Massey
The Center for
Electronic Music
New York

THE POPPYCOCK PATROL

I have a few comments on the December 1989 **EM**. In my article on wireless systems applications, the second sentence under the heading "The Wireless Band" did not read correctly, probably because my wording did not make my intent sufficiently clear to the editor. It should read, "Now the spectre of interference between your own systems looms along with the problem of interference with local broadcasts." It is conceivable that the sentence may be accurate as printed, i.e., I would think it possible that there could be summing or interaction between the spectra of two kinds of interference; however, this was not what I meant to say.

There is an important clarification to be made in the sidebar on SMPTE/EBU time code that appeared with the article "The Industrial Video." Drop-frame code does not run at a rate of 29.97 frames per second (fps), it runs at 30 fps. Drop-frame code was created to compensate for the cumulative error that occurs as a result of the difference between the 30 fps time code frame rate and the approximately 29.97 fps frame rate of NTSC color video, which is used for color TV in the U.S. (The 59.94 Hz

DAVID POVILAITIS

field rate of color video is what is really at issue, and there are two fields for each frame.) Certain frame addresses are omitted on a formulaic basis to resolve this discrepancy; the time code rate is never altered. This essentially boils down to, "The time code you are seeing is true; only the names have been changed to protect its integrity." This distinction is a source of constant confusion and has significant ramifications in practice.

In the inconceivably picayune department, VITC is actually a Recommended Practice of SMPTE (RP108), not a part of the SMPTE/EBU time code standard (ANSI V98.12M). Also, in the sidebar "Down to the Wire," which appeared with the article on power amplifiers, Mr. Kumin states that, "Audio frequencies tend to travel on the surface of a conductor, which is called 'the skin effect.'" In fact, skin effect is an electromagnetic phenomenon that increases with frequency; low frequencies do not tend to travel solely on the surface. This is one of the reasons some specialty cables use two or three different gauges of wire within the same cable—the same division-of-labor concept as multiway speaker systems.

Larry (the O) Oppenheimer
Toys in the Attic
California

MARKETING YOUR PRODUCT IDEA

I enjoy reading *Electronic Musician,* but I was surprised and disappointed by your response to one of the "generic" letters in your September issue. In that letter, "Wanda B. Rich" felt she had a great new product idea and asked for advice on how to see it become a real product. Your response was basically that the big boys are controlling the game, and they're not gonna let you play, so forget about it and go back to flipping burgers.

If her question had been how to get an original song recorded, or how to make her own record, I doubt your answer would have been so negative. In fact, the situations are very similar, and fortunately, the situation is not as black as you painted it. If an idea is a good one (in the music area or not), there are ways to see it become a product, even if the desire is to sell the idea to someone else that brings it to the market. As with a song or recording, there are many different ways to go about it, and it will require some hard work. Unfortunately, it's also possible to get burned (or burned out). That's actually what makes it interesting.

I've seen some of both the good sides and bad sides of getting product ideas into the market, and I'm willing to give advice to any reader who is serious about an idea they have. Write me at: 919 E. Hillsdale Blvd., Suite 300, Foster City, CA 94404.

Paul F. Titchener
California

Paul—You bring up some good points, and in retrospect, the response to Wanda was probably a little bleaker than it needed to be. However, just as it's unrealistic for people to think that because they've written a great song they'll get a record contract, so should they realize that a great product idea won't necessarily result in a commercial product, particularly in the case of hardware. If you create some unbelievable software on your own, you may have better luck turning it into a successful commercial product, but even there the pitfalls are many. The point is, you need to have lots of patience and tread carefully.—Bob O'D

HEAR HERE

I couldn't help writing after reading Kerry Livgren's December letter about the toll of high dBs on our ability to hear. A study of the ear's anatomy makes it easy to conclude that the organ was there in order to perceive the breaking of a twig of an approaching enemy or a distant cry for help. It is a precise and delicate sensory tool. The prospect of losing Kerry Livgrens or Pete Townshends to hearing loss troubles me greatly. Kerry, please accept my best wishes. I can assure you that your unselfish warning to fellow musicians is one I'll take to heart.

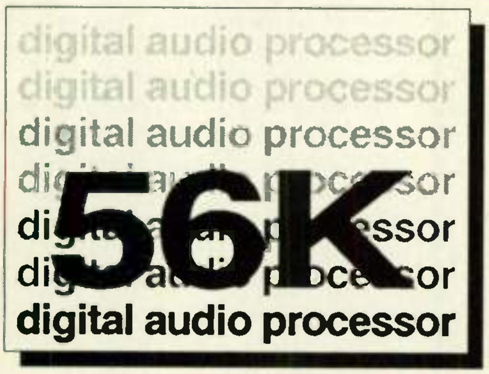
Jeff Borkowski
New York

THE GOOD OLD DIYS

We've all come a long way from when we started with you a bunch of years ago. But I still need help arranging my chicken-wire-and-gum studio. I'd love to see some of the projects you published in the earlier days. Lately things have been real technical. That's good, but I'm getting a little bored. I need to build something. And even if I

Chilled Turtle Surprise!

Winter... Cold waves, frozen seagulls, snow in our sandbox, and some great news from Turtle Beach Softworks. The boys at the beach have hung up the surf boards, lit a fire in the fireplace, and cooked up 2 hot new products to get you through the cold months.



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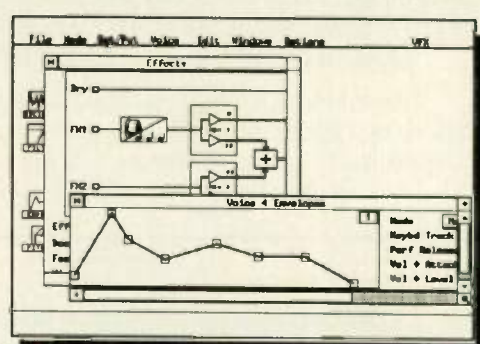
Is Oview/VFX really this cool or are we just rattling your shell? See for yourself. Call or write us for a **free demo disk**. Please include your graphics adapter type and preferred disk size.

Proteus owners: check out Oview/Proteus!



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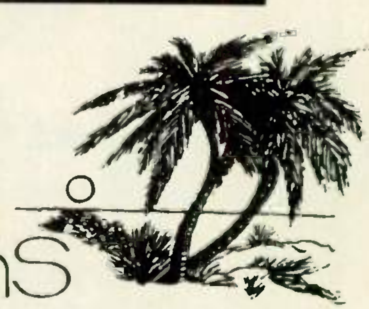

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don't build, it's fun reading about the ideas and techniques that go into it. You've had some great ones.

Scott Harriman
California

Scott—We're trying to get Craig and Vanessa free time to create some really special projects, including do-it-yourself designs. The other editors are picking up a lot of tasks Craig and Vanessa usually handle—such as answering the majority of the letters—so they can create the kind of articles and DIY projects you want, including Craig's long-promised mixer project. Patience will be rewarded.

We are preparing an updated back-issue listing if you're interested in finding out what other DIY projects we ran in the past. This is a good chance to remind everybody that the "FYI: For Your Information" page has important info about doing business with **EM** *and the Mix Bookshelf, including how to obtain back issues; read it before you write or call.*

—Steve O

OPERATION HELP

Exponential VCO: I'm looking for specs and suppliers for an exponential VCO, a CEM-3340, or any other. John Foster, Apt. 5, 2016 Edinburgh Dr., Burlington, Ontario, Canada L7R 2C9; tel. (416) 639-2361.

Wanted: Orchestron optical sound disks; info on obtaining AD7574 and AD7528 chips for an ailing Decillionix DX-1 card (Decillionix appears to be out of business); info on obtaining software to allow the Mountain Computer Music System (Soundchaser/Syntauri) for Apple II to be driven from a MIDI card instead of a keyboard—I know this exists. Also wish to correspond with users of the Greengate DS3 sampler, especially 8-track and crossfade looping software users. Tim Fluharty, 22722 Denker Ave., Torrance, CA 90501; tel. (213) 533-4121.

Preamp: Please recommend a suitable, not-too-expensive preamp/impedance-matching transformer (or kit) that will cut down the noise and boost the output level of my Rhodes Stage 88 Mark 1 to match the input level expected by my effects. Randy Piscione, 475 The West Mall, Suite 210, Etobicoke, Ontario, Canada M9C 4Z3; tel. (416) 626-6411.

ERROR LOG

December 1989, "100 Great Products for Under $100," p.48: An old number was given for Triangle Audio. The correct number is (301) 526-6224. ■


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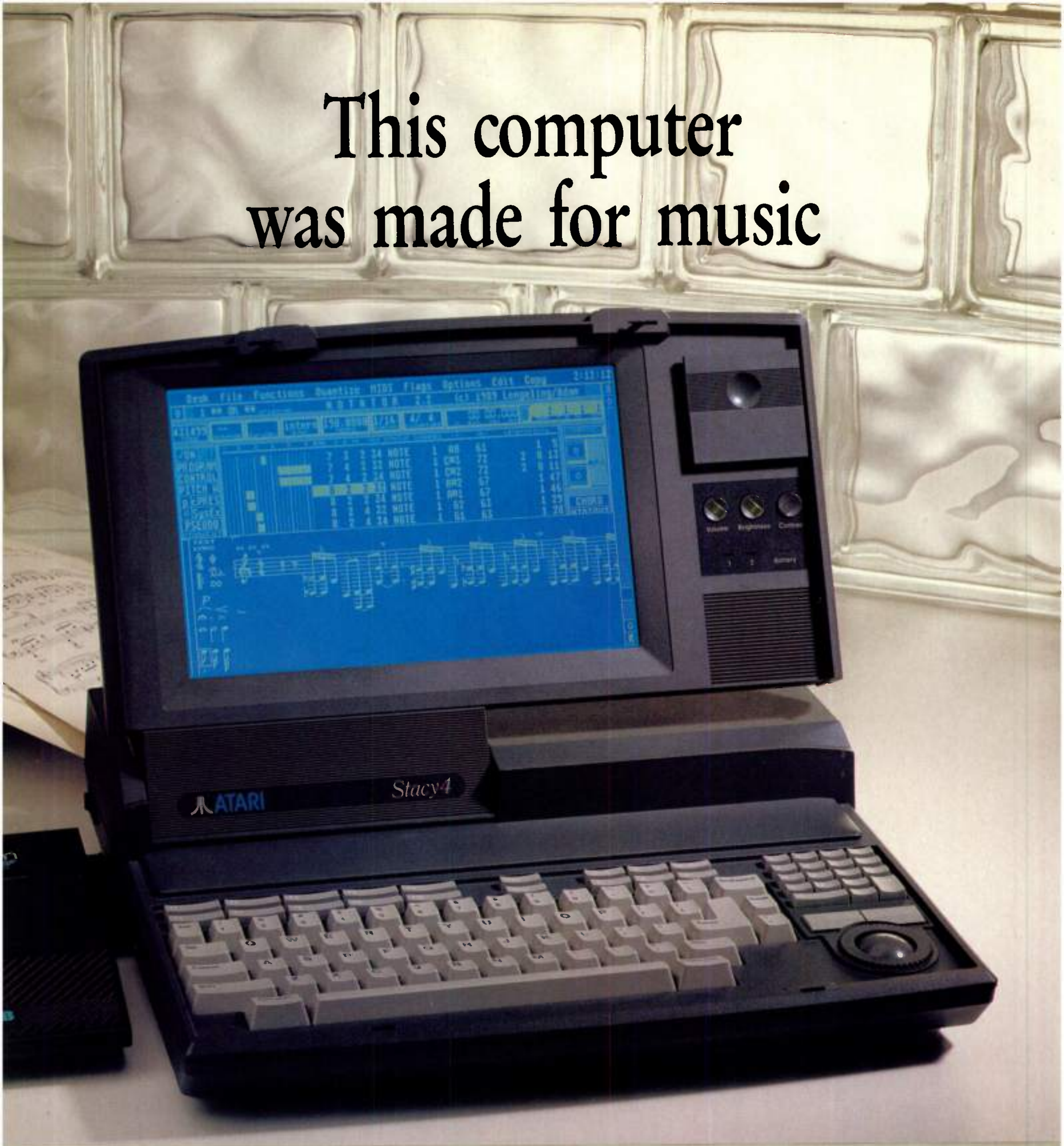

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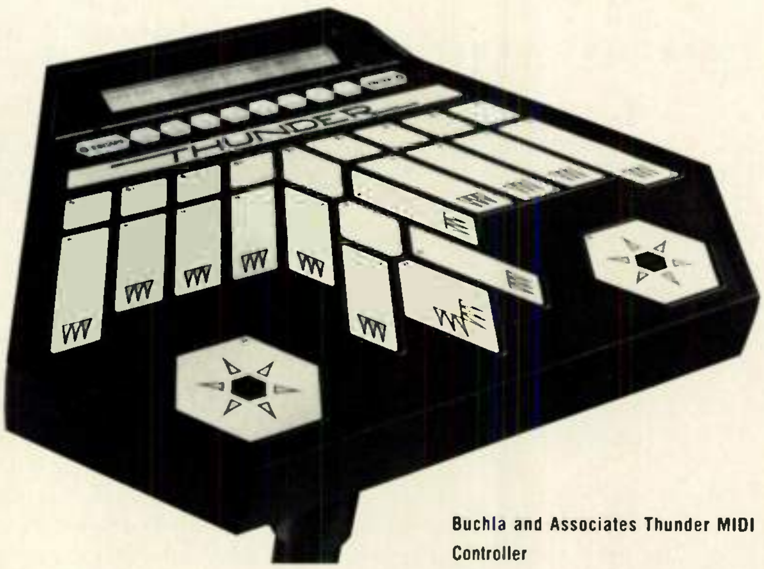
408-745-2367

For Valentine's Day, give an alternative MIDI controller, a feature-packed Mac MIDI interface, or a power supply.



Buchla and Associates Thunder MIDI Controller

MIDI

Polyphonics presents the IMS-200 ($239.95), a 3-in, 9-out, 1U rack-mount MIDI switcher that allows selection/disabling of MIDI sources via toggle switch, with LED status indicators. An autoswitch scans the first two inputs and routes MIDI data from the active master to the designated output.

Polyphonics Inc.
71 Fairway Dr.
N. Kingstown, RI 02852
tel. (401) 295-7659

Synthesizer pioneer Don Buchla, long noted for his innovative synthesizer controllers, has designed Thunder, ($1,990) an alternative to conventional MIDI controllers. Based on sealed-membrane technology, Thunder's playing surface is an array of keys organized to complement the human hand. The keys can sense velocity, location, and pressure and can be individually programmed. Editing is facilitated with software-labeled keys, dedicated editing controls, a sophisticated operating language that allows users to program complex relationships between input gestures and musical responses, and a backlit display. Setups can be stored internally or on plug-in cards. Thunder can address all MIDI controller types, route multiple MIDI program changes, and capture, store, and transmit sysex, and lots more.

Buchla and Associates
PO Box 10205
Berkeley, CA 94709

Mark of the Unicorn has introduced the *MIDI Time Piece* ($495), a combination MIDI merger/splitter, SMPTE converter, and 8 × 8 MIDI interface for the Macintosh, controlled from a desk accessory. The eight independent MIDI inputs and outputs, including one input and one output on the front panel, give you 128 MIDI channels, running from either the printer port or the modem port (selectable), with a passive thru jack that connects to the printer or modem. The MIDI Time Piece's Network port permits two units—including interfaces from other manufacturers—to be chained from each Mac serial port, allowing up to 512 MIDI channels. MIDI data can be rechannelized or merged, channel data muted, and MIDI in/out assignments switched. All four SMPTE formats, MIDI time code, and Direct Time Lock are supported. In addition to the 1 MHz asynchronous data-transfer rate, a higher transfer rate is offered. The next version of Mark of the Unicorn's *Performer* will fully support all MIDI Time Piece features.

Mark of the Unicorn
222 Third St.
Cambridge, MA 02142
tel. (617) 576-2760

AMPLIFIERS

ADA offers the T100S ($949), an all-tube, stereo power amplifier that delivers 50 watts per channel into 8 ohms. The 2U, 27-pound unit has a switch-selectable output impedance of 4, 8, or 16 ohms for each channel, and the channels have independent input-level attenuators. An "active high-voltage regulation" feature eliminates power-supply-induced 60 Hz hum and adds definition to sharp attacks and transients. The power supply is programmable to preserve and achieve optimal performance with various types of 6CA7 (EL-34) power-output tubes. Inputs and outputs are ¼-inch phone jacks.

ADA Amplification Systems
7303-D Edgewater Dr.
Oakland, CA 94621
tel. (415) 632-1323

SIGNAL PROCESSORS

Boss introduced the BE-5B multi-effects processor for bass guitar ($395). The 10-band graphic EQ (±15 dB, 31.5 Hz to 16 kHz), digital delay/chorus/flanger (2 ms to 500 ms delay, 15 kHz bandwidth), limiter, enhancer, and overdrive can be used simultaneously and modified from the front panel. An additional effects loop, headphone jack, and tuner out jack are included.

RolandCorp US
7200 Dominion Cir.
Los Angeles, CA 90040
tel. (213) 685-5141

SYNTHS AND MODS

The Xpander Xpansion is a retrofit for the Oberheim Xpander ($179.95, six inputs) and Matrix-12 ($279.95, twelve inputs) that allows external audio signals to be processed by the synth's filters and VCAs. The mod is nondestructive to the synth's circuitry and only takes one hour to install.

Oddernmart
1387 Baker St.
San Francisco, CA 94115
tel. (415) 346-8129

The unique sound of the PPG Wave 2.2 and 2.3 synths returns with the MicroWave ($1,995) from Waldorf Electronics, distributed by Steinberg/Jones. The 2U rack-mount, 8-voice multitimbral, wavetable synth has 64 Single sounds and 64 Multis. Additional sounds and Multis can be accessed via RAM card. The unit provides 32 preconfigured wavetables, twelve user-definable wavetables, and twelve that can be stored to a RAM card. Each wavetable contains 65 different waveforms. Other features include stereo outputs, four independent mono outputs, flexible modulation routing, modulation macros, and global editing, augmented by eight independent edit buffers that allow you to make changes to multiple sounds without storing them first.

Steinberg/Jones is also distributing the Musitronics M•EX ($425), a multimode expansion board for the Roland D-50/D-550 synthesizers that adds 8-voice multitimbral capabilities and increases memory storage to 128 patches, with an option for adding a third bank (for a total of 192 patches). M•EX also adds ten new PCM, looped waveforms and extends master keyboard functions such as the ability to simultaneously transmit on two MIDI channels. The D-50's shift key becomes a "panic button," sending an "all notes off" command to all eight voices.

Musitronics M•EX

Steinberg/Jones
17700 Raymer St., Suite 1002
Northridge, CA 91325
tel. (818) 993-4091

SOFTWARE

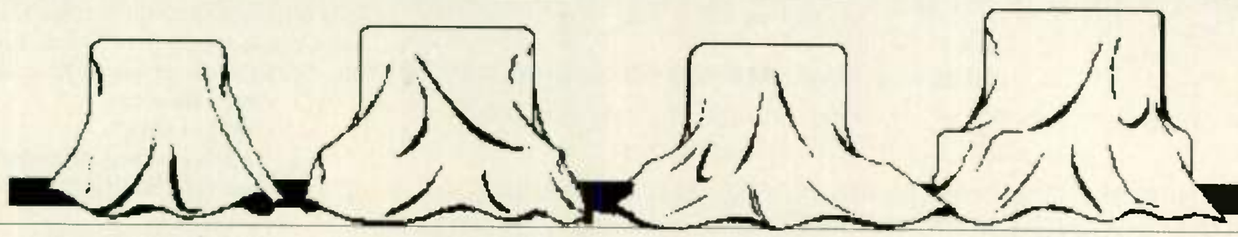
Following up on the success of Vision, Opcode introduced EZVision (under $200), an entry-level, MIDI Manager-compatible, MIDI sequencer for the Macintosh. Although it doesn't have Vision's SMPTE sync, extensive quantize control, and integrated list event editing, EZVision offers most of the larger program's essential features, including a Mixer window and other graphic editing capabilities.

Opcode Systems
3641 Haven Dr., Suite A
Menlo Park, CA 94025-1010
tel. (415) 321-8977

NameThatChord ($59.50 with transposition option, $49.50 without option) is a text font of chord-name symbols for the Macintosh, available with both laser and bitmap fonts. You

BOSS BE-5B Bass Multiple Effects

SOUND QUEST SYNERGY PREMIERE



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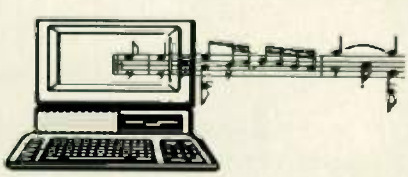

Sound Quest Inc.
1573 Eglinton Avenue West, Ste. 200
Toronto, ON, Canada
M6E 2G9
Telephone: 1-800-387-8720 1-416-256-0466

can type chord-name symbols just like you type text. Also available: FretFinder ($49.50), a text font for the Mac that enables you to type guitar tablature, complete with the chord name characters.

Note Ware Co.
PO Box 9953
Marina del Rey, CA 90295
tel. (213) 822-1300

SAMPLERS

The Cheetah SX16 stereo, 16-bit sampler ($1,399) supports up to 48 kHz sampling, 8-voice polyphony, eight outputs, velocity and aftertouch sensitivity, and a 3.5-inch disk drive, all in a 1U rack-mount package. Its 512 KB RAM is expandable to 2 MB. The unit can load Akai S900 samples, providing an instant sound library. Several options, including a joystick and a CRT card that allows editing on a regular TV screen, are available.

Jessico
11230 Grandview Ave.
Wheaton, MD 20902
tel. (301) 949-9314

PUBLICATIONS

Akai EWV 2000 owners can subscribe to *The Flit Bun* ($15), a users' newsletter that includes reviews, setups, and applications hints. The editor offers free subscriptions in return for article contributions.

Jim West
6234 Silverleaf
Baton Rouge, LA 70812

American Design Components has a free catalog of electronic parts, including components such as ICs, crystals, fans, power supplies, and connectors, and computer-related products such as disk drives, monitors, and add-on boards.

American Design Components
815 Fairview Ave.
Fairview, NJ 07022
tel. (800) 776-3700
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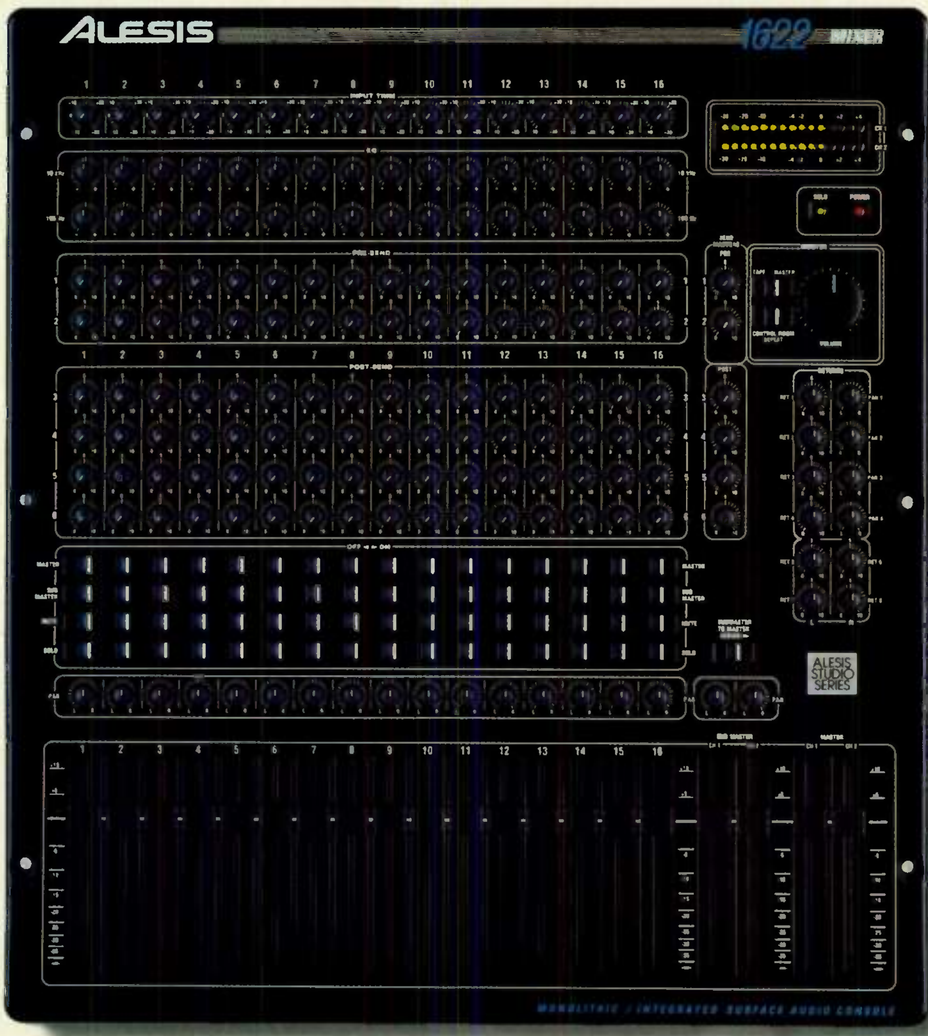
EVENT TIMING

Jeanius Electronics' Russian Dragon ($350 tabletop, $495 rackmount) measures the timing accuracy of

16 Channels 6 Sends 8 Receives $799



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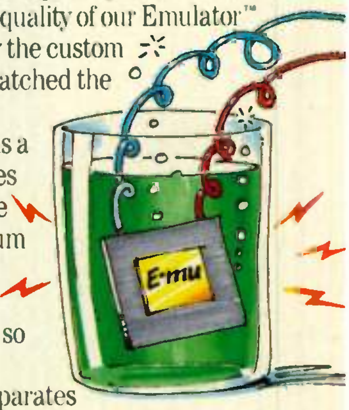

For you, this means a keyboard that combines extraordinarily life-like sampling. And Spectrum Interpolation Digital Synthesis.

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And it can grow to gigantic proportions. You can increase memory up to 8 Megabytes. Install a 40MB internal hard drive. Or add a host of Mac-compatible peripherals through its SCSI port. Like the new 600MB erasable optical drive or our own RM45 rack-mount 45MB cartridge hard drive.

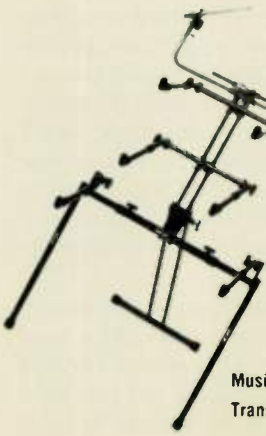
Built in the U.S., Emax II is taking over your neighborhood E-mu dealer.

Stop by. And discover the new keyboard that's so good, it's scary.

E-mu Systems, Inc.
applied magic for the arts

two events that were intended to occur simultaneously, displaying which is ahead (rushin') or behind (draggin') and by how much. Applications include checking how closely a drummer is playing to a click track, investigating delays in a MIDI setup, or monitoring how tightly a percussion overdub is performed. The LED display detects timing errors of 80 microseconds to a half-second.

Jeanius Electronics
2815 Swandale Dr.
San Antonio, TX 78230
tel. (512) 525-0719



Music Industries
Transformer QL-699 Stand

POWER

Kleen-Line AC power regulator/conditioners (starting at $400) feature heavy-load startup capability, 7-stage input spike/surge suppression, wide-band input interference filtering, and sine wave output (including a patented isolator output design). Regulated output is 117 VAC ±4% for a 90 to 140 VAC, 60 Hz input range. Standard and isolator models rated at 250, 500, 1,000, 1,500, and 2,000 watts are offered.

Electronic Specialists, Inc.
171 South Main St.
Natick, MA 01760
tel. (508) 655-1532
or (800) 225-4876

Juice Goose's Twelve PAQ ($249) provides low-voltage current into six independent power outputs, each of which supply 9 VAC, 18 VAC, 9 VDC, and 12 VDC (selectable), connected via a Juice Goose output cable. The unit supports a long list of manufacturers including Peavey, Lexicon, Alesis, ART, Roland, DOD, and many more. Each output is isolated from the other and from the current input, and the unit includes line filtration and spike protection. Six 120 VAC convenience outlets are included for a total of twelve outputs.

Juice Goose
7320 Ashcroft #302
Houston, TX 77081
tel. (713) 772-1404

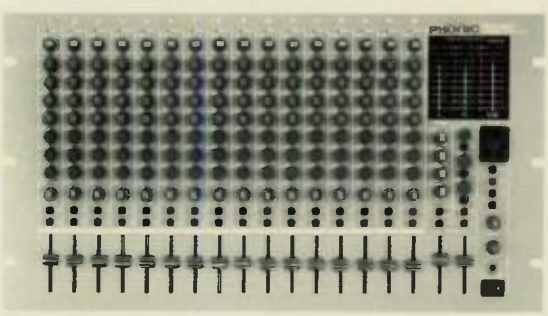
STANDS

Music Industries added two heavy-duty Transformer keyboard stands to their Quik-Lok series. The QL-699 ($209.95) is designed for playing while seated and allows plenty of free leg room. The QL-690 ($159.95) is designed for playing while standing. Both stands can be collapsed quickly and come with two adjustable tiers. Two optional, nonadjustable tiers (three varieties, $25 to $30) may be added.

Music Industries
99 Tulip Ave.
Floral Park, NY 11001
tel. (800) 431-6699
or (516) 352-4110

The Sound Office ($199 for basic RAMstand) provides a modular system of equipment stands for the studio. Made of heavy wall steel tubing with hardwood veneer shelves and tables, the system is designed for quick assembly and breakdown. To provide flexibility the speaker stands pivot and spin on articulating arms, and the computer desk swings and turns. The many options allow custom setups.

Sound Office
1020 S. Main St.
Moscow, ID 83843
tel. (208) 882-6549



Phonic PMX-1600 Mixer

MIXERS

Hosa Technology is marketing a line of gear by Phonic, including the Model PMX-1600 ($1,790), a 16-channel, rack-mountable, stereo mixer with 3-band, semiparametric channel EQ; three effects buses with four stereo returns; 21 back panel insert points; prefader monitor bus; LED peak indicators on both channels and main outputs; and XLR inputs on the first eight channels. The Model BKX-8800 ($480), an 8 × 2 mixer with two effects buses and 2-band channel EQ, is also available.

Hosa Technology, Inc.
13042 Moore St.
Cerritos, CA 90701
tel. (213) 926-0808

REV UP

Opcode Systems (tel. [415] 369-8131) released Version 3.0 of *CUE* for the Macintosh ($595, upgrades $49.95; free upgrades for purchases after October 1, 1989). New features include a Custom Menu window that lists your most-used menu commands and offers point-and-click access to them; sending cue point timings to the Tesla Video Streamer Box for superimposing color streamers and punches on video (with colors assignable from the Mac); transfer of tempo and meters, via MIDI files, to Opcode's *Vision;* and a real-time cue-point logging feature. A color version is available for the Mac II; upgrades are an additional $50. Opcode has also lowered the price of its Timecode Machine SMPTE-to-MIDI converter to $199 and is making SMPTE-striping software available for the Timecode Machine and Atari ST as well as the Macintosh...Altech Systems (tel. [318] 226-1702) is shipping MIDIBASIC Version 3.0 ($100, upgrades $25), which adds time-stamping functions that allow Microsoft *QuickBasic* programs to do time-sensitive tasks such as sequencing or handling "slow" MIDI dumps like those from a Roland D-50 or Korg DDD-5. Two sequencers are included as examples. One MB RAM is required to construct applications. ■

Hard-Won Lessons About Hard Disks

Hard disks are great—until they crash, or fragment so badly they crash your computer. Here's a survival guide on living in the hard disk age.

By Craig Anderton

Hard disks are becoming common not just in computer setups, but as add-ons for samplers and digital recording systems. While hard disks are not cheap (around $10 to $20 per megabyte of memory), they're much faster than floppy disks and hold a lot more data. Because they hold so much data, though, you need to manage that data with care—which is what this article is all about.

BACKING UP IS HARD TO DO

...but necessary. Hard disks aren't as failure-prone as they used to be, but someday your hard disk *will* fail. If you've backed up your disk, you'll suffer a minor inconvenience; otherwise, it's disaster time.

One way to back up a computer's hard disk is with a commercial backup program. An *incremental* program creates a set of floppies, backing up only those files that have been changed since the last backup, which takes less time than saving the entire hard disk. (Unfortunately, I know of no incremental backup programs for samplers, so all you can do is scrupulously back up your files to floppies or other storage media.) When—I didn't say "if"—your hard disk crashes, you copy the backup set of floppies over to a new (or reformatted, as discussed later) hard disk, and you're set—unless you had *installed* copy-protected software on your hard disk prior to the crash.

THE PERILS OF COPY-PROTECTED SOFTWARE

With many copy-protected disks, you need to insert the master disk for ID verification on boot-up. Since this is an annoying waste of time, software companies often allow you to *install* one or two copies of the master disk on a hard disk, letting you run the program directly from the hard disk without inserting the master disk. If you later need to remove the program from the hard disk, you can *de-install* the program back to the master disk, recouping your installation option. Unfortunately, if the hard disk crashes with an installed program, you lose your installation. This is why most programs offer two installations.

Installed, copy-protected programs will transfer to backup floppies, but unfortunately, will not necessarily transfer back to the hard disk again in the case of a crash—a major drag. If you install copy-protected software, make sure your backup program offers the option to back up only data files, since backing up installed software creates more problems than it solves.

Copy-protected programs that are not installed, but simply copied to, a hard disk (and therefore require a key disk insertion at boot-up) are immune from this potential problem and can be safely backed up.

ALTERNATE BACKUP METHODS

Floppies aren't the only way to back up your data. If your system supports multiple hard disks, you can back up one hard disk to another. This is fast, but assumes that the two hard disks won't fail at the same time (unlikely, but possible). Hard disk contents can also be transferred to tape cartridge backup units quickly and conveniently, although this is not an inexpensive way to go, and tape drives can't be used as pseudo-hard disks should the need arise. A complete discussion of tape backup is beyond the scope of this article, but new technologies offer the promise of inexpensive, reliable tape data storage (see sidebar, "High-Tech Tape").

Most hard disks have finite storage, and when filled, you'll need to add another hard disk. However, *removable* hard disks are starting to appear at semi-reasonable prices and are ideal for backup. Not only do they offer unlimited storage (simply pop in a new disk cartridge to store more data), but you can carry the data around with you if you're security-conscious. Aside from

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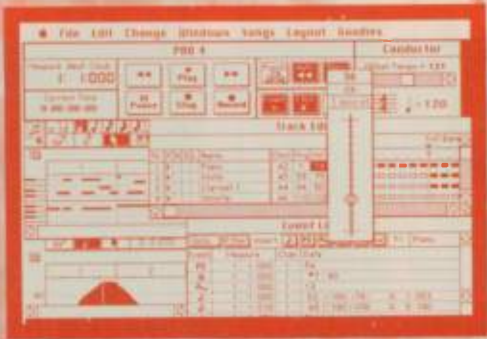

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After ten years of listening to the needs of musicians, Passport continues to offer the highest standards in MIDI technology for personal computers.

Passport products are available at fine music, computer and software stores worldwide. For more information visit a Passport dealer near you or call us for a free catalog.

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Half Moon Bay, California 94019 USA
(415) 726-0280

expense, the only tradeoff is slightly slower access time.

Also consider off-site backup for important files; a fire, for example, could wipe out your backups *and* originals. A safe deposit box is a good option, but there are other possibilities. Many telecommunications services have personal "workspaces" where you can save files. While storage time costs money, this is my preferred backup method when I'm on the road with a laptop and don't have access to other storage media.

UNITED WE STAND, FRAGMENTED WE CRASH...

Understanding fragmentation is the key to hard disk management, so before continuing let's address this in depth.

When you copy a file over to a hard disk, it will look for a contiguous block of memory on which to write the data. However, as you save, remove, and modify files, it becomes harder to find contiguous blocks of memory.

For example, suppose you have a Mac and just bought a hard disk. You copy your system file over to the hard disk, and that takes up a block of memory. Then you copy over some applications, and these take up the next available block of memory. If you then go back and add some desk accessories and fonts to the system, they cannot be written in a block adjacent to the system, because it will already be occupied by other applications. So the rest of the system goes somewhere else on the disk.

As you continue to save, delete, and modify files, these are "fragmented" to different places on the disk. This slows down access time because the hard disk has to pick up pieces of a file from all over the disk instead of from one contiguous block of data; with severely fragmented system files, crashes and other problems may result.

AVOIDING AND REPAIRING FRAGMENTATION

The basic rule is to copy files that won't change (applications, operating system, etc.) over to the hard disk first, then copy over files that will change often. If you don't mess with the applications and system, they'll continue to occupy the same contiguous blocks of memory they did when you first copied them over. With samplers, copy over factory sounds first, then sounds under development.

There are two main ways to clean up a fragmented hard disk:

1. Commercial defragmentation programs. These rearrange data into contiguous memory segments, but there are several cautions. First, back up your disk before running a defragmentation program as it may not work flawlessly. Second, defragmenting a severely fragmented disk with a lot of data takes time. Third, defragmenting copy-protected software will cause *major* problems. *Al-*

HIGH-TECH TAPE

If you're working with a relatively small hard disk (under 40 MB) and aren't constantly filling and/or changing its contents, you generally don't need to worry about a tape backup system. But if you're doing hard disk recording onto huge drives, a tape backup system is essential so that you can free up the drive(s) for other projects.

Like hard drives, tape drives come in both stand-alone and internal formats, and many use SCSI to communicate with the connected hard drive and computer/instrument. Unlike hard drives, however, tape drives do not offer random access to the data—instead, it's recorded linearly like an audio tape deck.

Systems based on cassette data tapes, 8mm videotape, and DAT (the most recent development) are all currently available, as are many other formats, with each one having different benefits, storage capabilities, and costs. To give you an idea, a typical DAT drive (which is *not* the same thing as a DAT recorder, though there are companies working on a conversion box that would allow DAT recorders to work as a tape backup system) can store up to 1.2 gigabytes, or 1,200 MB, of data. The type of system you'll need depends on how much money you're willing to spend, how quickly you'll need to make backups (some systems require several hours for large hard disks), and the kind of hardware you currently own. *(Thanks to Oz Barron of Sytron Corp. and Suz Howells of Digidesign for technical assistance.)* —*Bob O'Donnell* *ways, without fail, de-install any copy-protected programs before running defragmentation software.* However, there is a workaround; use *partitioning software* (such as Symantec's SUM II for the Mac) to divide a single hard disk into several virtual disks, each of which behaves like a separate hard drive. Keep all your copy-protected software in one logical drive, and only defragment the other partitioned drives.

2. The second method involves reformatting (also called initialization—the process of erasing all files on the disk and preparing the disk to accept new data), copying over system and applications, then copying your backup floppies over to the hard disk. Copied files will all be written in contiguous blocks. Copying can be a time-consuming procedure, but is less so if you have backup software. Copy-protected applications that were de-installed prior to reformatting should be copied over individually from the master disks.

Fragmentation can be quite a problem with hard disk-based samplers since constant tweaking of samples promotes excessive fragmentation. To combat this, I keep an up-to-date list of the samples on disk, and periodically reformat the hard disk and recopy files back on to it from floppies. The improved loading time and reliability is well worth this effort.

HOW MUCH HARD DISK DO YOU NEED?

Maybe not as much as hard disk manufacturers would like you to believe. Most people load up a new hard disk with data, applications, and so on. While convenient, this invites fragmentation and, due to the sheer volume of data that needs to be backed up, often inhibits people from backing up as often as they should.

A different way to look at your hard disk is as a partner to floppies, not a replacement. Copy all your applications and system files to the hard disk, but leave your data files on floppies. Transfer any needed data files over to the hard disk at the beginning of a session for quick access, and before shutting down, save the files back to floppy. This solves several problems:

1. Fragmentation is minimized since the files that change the most live on floppies, not the hard disk.

2. Backing up is less of a problem; should the hard disk fail, all your data files are safely on floppy. These are always harder to re-create than system and application files.

3. You don't need lots of hard disk capacity, which saves money. An inexpensive 20-megabyte hard disk can hold a lot of applications—certainly just about every music program you would ever need.

However, large storage capacities are essential if you work a lot with samples or graphics/animation files, which tend to take up so much space that loading them from floppies all the time could get to be a drag. Samplers also like large hard disks; just remember to back up your samples on floppy, and periodically reformat and recopy, and all should be well. A final note: Larger drives also have slightly faster access times.

MISCELLANEOUS TIPS

Most articles on backing up exhort you to buy really expensive, top-quality floppies on the premise that your data is invaluable. But since name brand disks can cost up to twice as much as generic ones, I buy twice as many generic disks and make two backups. This makes me feel more secure than having a single, high-quality backup.

A couple of other fine points: If power is interrupted as you save to, or read from, a hard disk, irreversible damage might occur. If you don't have an uninterruptible power supply, and a storm is on the horizon, forego the hard disk and use floppies. Also, never jar the hard disk while it's in use.

When you purchase a hard disk for musical applications, look for the following features:

1. At least a one-year warranty. If a company doesn't believe in its hard disks, why should you?

2. Quiet operation. Of course, quiet hard disks are best suited to studio applications.

3. Free software. Hard disks often come bundled with free software. My hard disk came with encryption, backup, and print spooling programs, most of which I would have had to buy eventually anyway.

4. Sampler compatibility. If a manufacturer recommends particular hard disks for use with their samplers, believe those recommendations. Some hard disks that work fine with computers don't work as well with particular samplers, for reasons far too esoteric to go into here.

5. Dual SCSI connectors. This is important if you want to use your hard disk as part of a daisy-chained SCSI system. If the hard disk includes a terminator, it's best if you have the option to disable this if needed.

6. Make sure the hard disk "parks" its head when not in use. If you plan to move a hard disk—even if it's just across the room—head-parking is important. Some models park automatically upon shutdown; others require a command to park, and some older drives don't park their heads at all.

Hard disks can really make life in the studio go faster and easier. Follow the above tips, and you'll swear by your hard disk, not at it.

Acknowledgment: Thanks to hard disk user par excellence Richard Vanderlippe (customer support, Passport Designs) for fact-checking and supplying additional valuable information.

Craig Anderton *shares a relatively small (but very interesting) planet located in a relatively obscure part of the Milky Way with over five billion other humanoid bipeds.*

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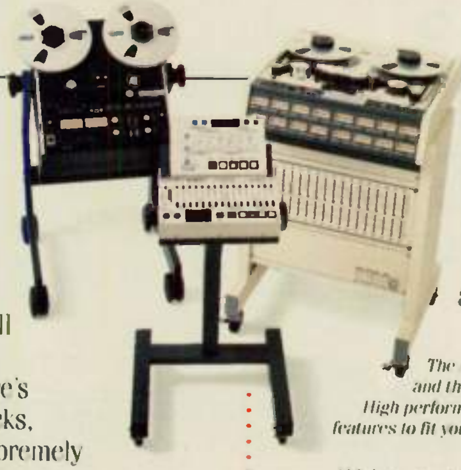
The MX-80, (and its little brother, the one-inch, 16-track MX-70) share a constant-tension, servo-controlled transport, plus the Otari 37-pin standard parallel interface for SMPTE synchronizers, editors, and controllers.



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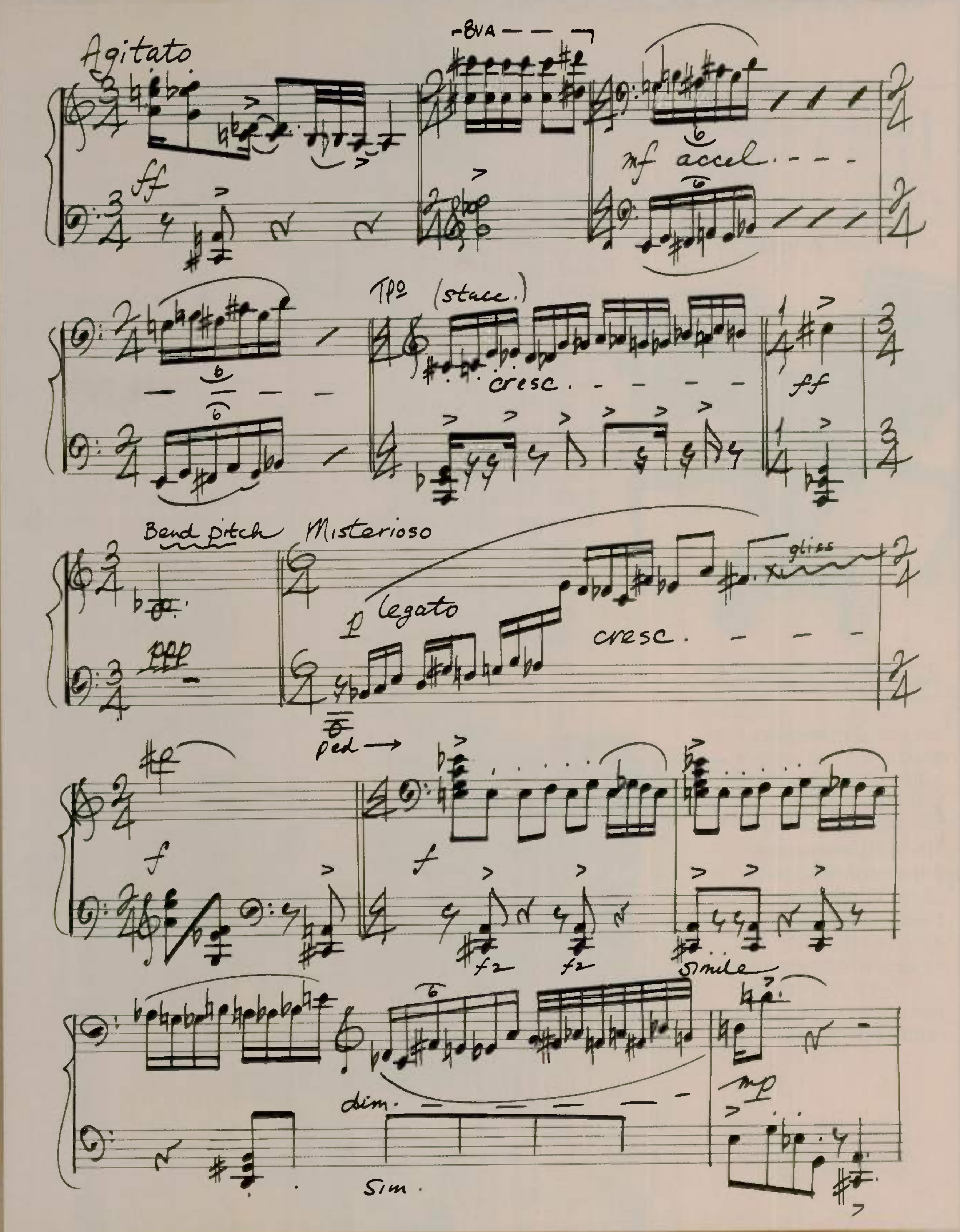


Ditto.

If you want to read the other side first, go right ahead. You're exactly the kind of person we're looking for, so we're more than happy to wait for you.

Now that you're back, get ready to find out all about the new Rhodes MK-80. Or as we like to think of it, your next keyboard.

What makes us so confident that you're going to want this keyboard? Two things, really. The fact that our ingenious stretch tuning method accurately duplicates the inharmonicity of an acoustical piano. And the equally remarkable fact that our Advanced Structured Adaptive synthesis actually allows you to modify the harmonic content of your sounds.

Either one of those is quite a technological breakthrough. Together, they're only slightly less impressive than cold-fusion.

The MK-80 comes standard with perfected digital versions of the sounds Harold Rhodes himself pursued as "ideal": The classic tone, with its thick sustain and sharp attack; a modified sound with a higher harmonic content; a blended sound that's a combination of the first two; and a contemporary sound with bell-like qualities of synthesizer-based Rhodes sounds. Add to those four other extraordinary sounds — concert grand piano, electric grand piano, clavi, and vibraphone — and you've got an instrument that's ideal for performing.

As extraordinary as these sounds are, however, you may only want to use them as starting points. Which is fine. The MK-80 is equipped with chorus, phaser, and tremolo, as well as a three-band equalizer with parametric mid-range for advanced tonal adjustments. You can even edit the harmonics of the tones using a Macro Edit function on the ASA Operator level. And an Auto-Bend parameter allows you to apply a velocity-sensitive pitch envelope to your sounds.

What this means is that you can create all the legendary Rhodes sounds of the past 20 years, as well as new sounds you never imagined possible.

Then, once you've sculpted the sound you want, you can save it, along with 55 of its comrades, in the user memory. With variations on everything from Macro Edit, parameter settings and effects on/off switching to MIDI messages.

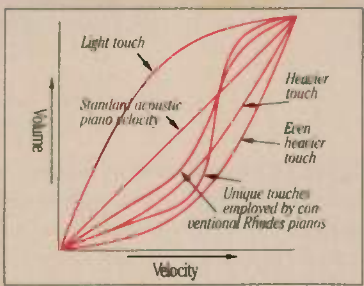
As if that weren't enough, the MK-80 is also a formidable MIDI controller.

There's plenty more. For example, we didn't even begin to tell you about the smaller, yet equally impressive, MK-60. (Just to whet your appetite: We packed many of the same features into a 64-note keyboard that uses an octave shift to play the complete 88-note range.)

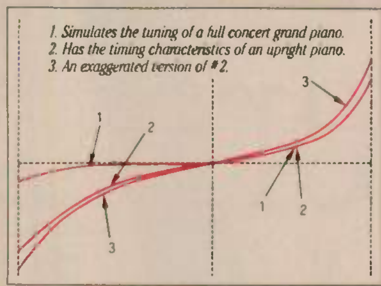
But since you've read this far, it's fair to assume that you're interested in seeing the new Rhodes firsthand. In fact, you're no doubt already wondering who in their right mind is going to take your old keyboard off your hands.

Probably someone who wasn't lucky enough to read this ad.

Or good enough.



It's one thing for a keyboard to have a nice feel. It's quite another for a keyboard to have six, all of which can be stored as parameters in the user memory.



These are the stretched tuning curves that give the new Rhodes its unique sound. If you want to find out how they work, read the ad.

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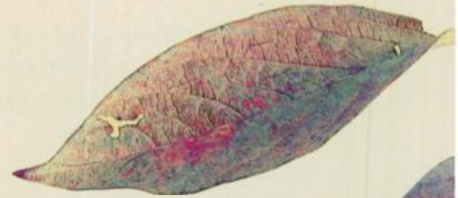

Mixers

Basic Studio Series, Part 4

By Craig Anderton

Few products are as intimidating as mixers. Not easily defined, a mixer can be as simple as a little 8-input, powered mixer for a club band or as complex as an 80-input, automated console the size of a small apartment. Prices can range from $100 to well over $100,000. What's more, different studios have different needs. While it's just as important to have the *right* mixer for the job, regardless of whether you have a 4-track cassette or 48-track digital recorder, the functions each person needs will be radically different.

It's even more difficult to come up with meaningful mixer specs. For example, mic preamps add noise. So do you measure noise with all the mic preamps running at full blast with all the levels wide open, or with all the faders set to off, which will yield a quiet mixer spec but bear no relationship to how the unit will perform in the real world? Then there's the issue of sound character. Equalizers that sound "tinny" to one engineer may sound just fine to another, and the true "golden ear" types can differentiate between different types of mic preamp technology, although proving that one technology is "better" than another is more difficult than just noting that a difference exists.

Given all these considerations, the brutal fact of buying a mixer is that you have to work with a mixer a bit to get a feel for how it performs and sounds. All I can do in this article is explain the purpose of various features so that you can narrow the range of candidates down to something reasonable. For example, if you run a MIDI studio and need a lot of inputs, the number of inputs might be more important than the purity of the mic preamps.

Let's assume you're mostly interested in a studio mixer rather than a P.A. mixer, and that if you're in the market for a $100,000 console, you already know what you're doing (even then, some of the following will still be of interest). Let's start with the basics and work up from there.

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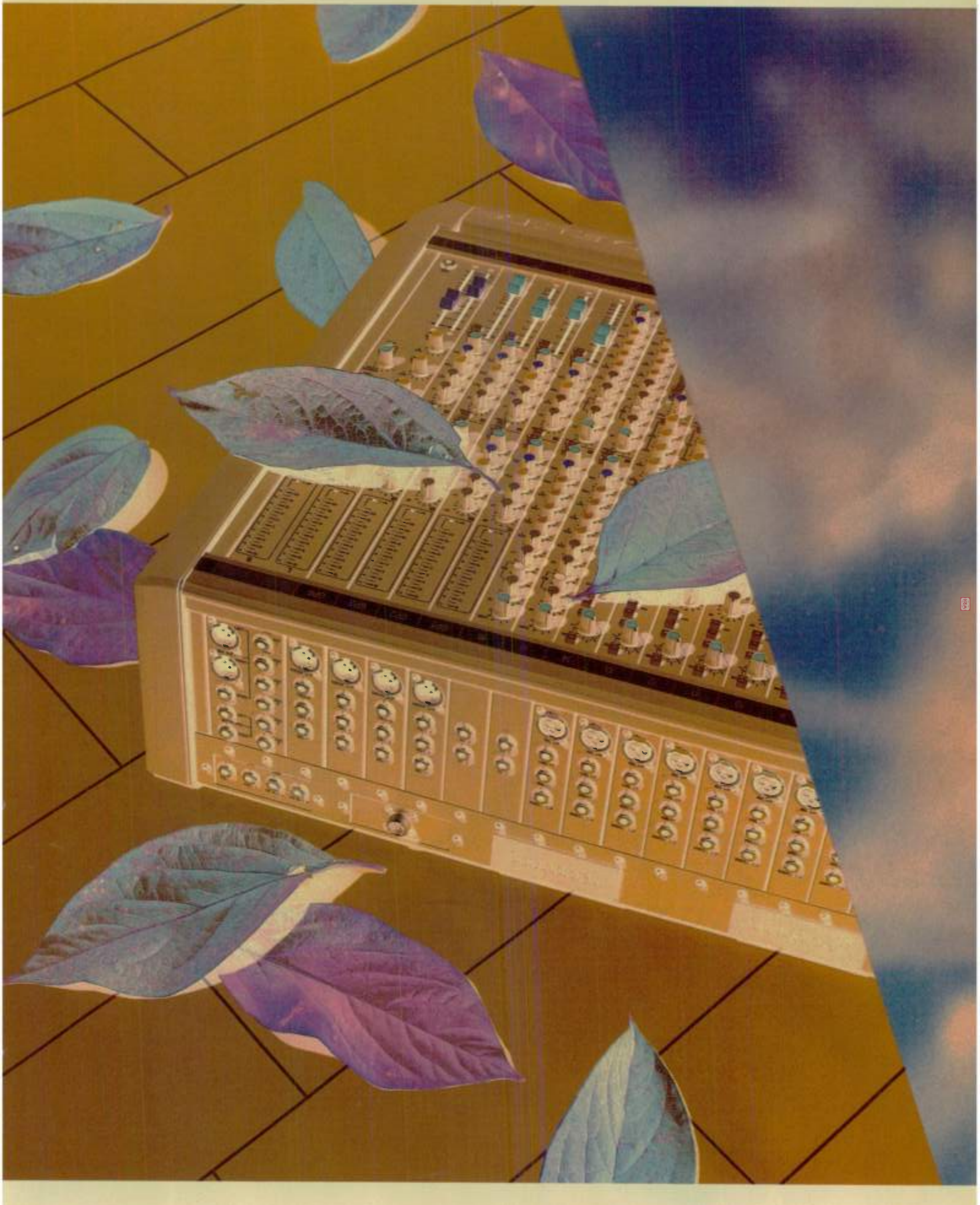

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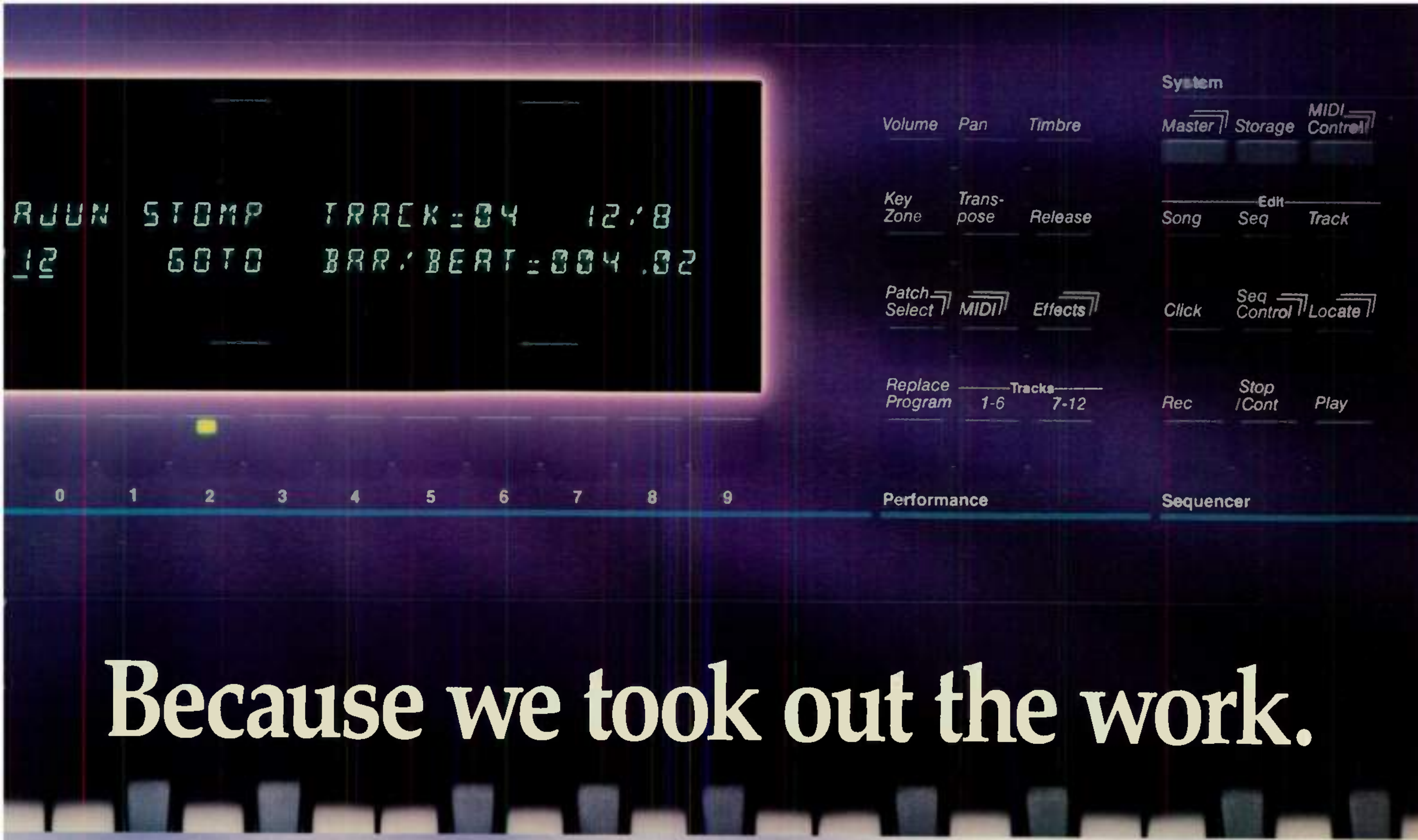

Because we took out the work.

at the keyboard. You can stay focused on your music as it happens.

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MIXER BASICS

A mixer combines many input signals electrically into a smaller number of outputs and lets you balance the relative levels of these input signals. For example, suppose you have a mastering tape deck with two inputs (left and right) and have eight channels to plug into that deck. A mixer can blend these eight channels into a stereo output that can plug directly into the mastering deck's stereo inputs.

Although mixers are common in keyboard setups, guitar racks, P.A. systems, and so on, it is in the recording studio that the mixer has reached its zenith. This all-important component plays the role of a traffic director, sound shaper, and control center. It's the audio interface between all your devices and a converter for some (for example, taking the low-level output from a mic and boosting it).

CHARACTERIZING MIXERS

You may have heard a mixer referred to as an "8-in, 1-out" (8 × 1) mixer, or a "24-in, 16-out" (24 × 16) mixer, and so on. The first number refers to the number of input channels available. For example, a 16-in mixer has sixteen different input channels, so you can plug in up to sixteen signal sources—microphones, instruments, tape tracks, etc. A 64-in mixer accepts 64 signal sources.

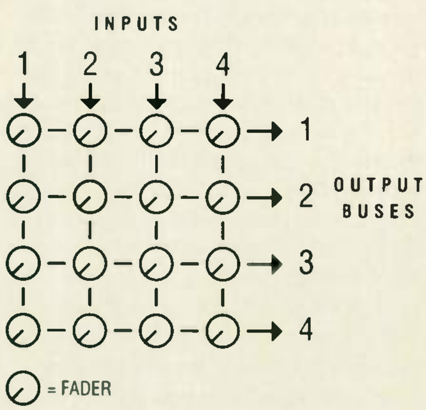


FIG. 1: Fader layout

The second number refers to the number of output buses available. What's a bus? Fig. 1 shows the input signal path as a vertical, downward flow and the output signal path (bus) as a horizontal flow from left to right. At the junction of the input and output lines, you'll find a control that sets the level (usually called a *fader*). This may be either a standard, rotary motion control or, more likely, a linear motion (slider) control.

With some mixers, you'll also be able to mix the output buses down to a smaller number of additional output buses (often called *subgroups*), and this is expressed with a third number. For example, a 16 × 8 × 2 mixer would have sixteen inputs and eight output buses, and those eight could be further mixed down into two buses (stereo).

Fig. 1 also shows four outputs. Each represents the output of a 4-in, 1-out "mini-mixer" within the console. For example, bus 1 mixes inputs 1 through 4, as does bus 2, bus 3, and bus 4. With this arrangement, we can take any input signal and, by properly setting the level controls, mix any desired amount of that signal into any of the four output buses. These buses will be assigned to some specific destination: headphone cue amp, main stereo mastering deck, reverb input, and the like.

The reason for having separate mixes on separate buses is that the mix will depend on the destination. A singer listening to the headphone-amp cue bus might not want to hear a lot of voice, while those monitoring in the control room might want the voice extremely loud—or perhaps even soloed—in order to catch any problems. Similarly, the mix feeding a reverb unit will probably be different from the main stereo mix intended for a master tape.

In practice, most mixers do not have separate level controls that feed input signals to each bus. Instead, there will be some combination of switches, level controls, and *panpots* (short for panoramic potentiometers); the latter place the signal on two or more buses in varying proportions. The most common use for panpots is feeding to the stereo output bus, where the panpot sends the signal to the right channel, left channel, or anywhere in between. By adjusting the level control and panpot together, the signal may be placed anywhere in the stereo field at any desired level.

There may be subgroups that let you control the level of several channels at once. A typical application would be if you were mixing a brass section, had each instrument assigned to its own channel, and wanted to bring the overall level of the entire section up or down. Moving one fader to change the brass section's level certainly beats moving each instrument's fader individually. This is, of course, also very useful for drums and background vocal groups.

Multiple output buses are useful when recording several tracks, in real time, to a multitrack recorder, especially if multiple signals are mixed to particular buses and these then feed the deck. However, it is not necessary to have one output bus for each tape track. In fact, it often makes sense to patch instruments from the input module directly to the recorder, thus bypassing the bus electronics to obtain a cleaner sound.

MIXER INPUT MODULES

The simple mixer in Fig. 1 does nothing but mix. Real-world consoles include a series of modules, capable of conditioning or modifying a sound, between the input and the actual mixing circuit. Thus, before the various input signals are mixed together, they can have their level, tone quality, or some other parameter changed via a signal-processing module in the input section.

Manufacturers do not offer identical input modules from one mixer to the next; in fact, what gives any one manufacturer's model a competitive edge over any other manufacturer's model will often be the input module design. As a result, it's crucial to understand the features found in, and differences between, input modules if you're going to make an intelligent buying decision.

FOR THE BEGINNER

What's in a Name?

A mixer goes by many names. A recording *console* is thought of as a fairly large mixer, like the kind in a major recording studio. The term mixing *board* seems to find favor in P.A. systems and smaller studios. In Britain, consoles are called mixing *desks*. Small, (generally) rack-mount mixers designed for utility applications and optimized for mixing fairly high-level signals are called *line mixers*. *Keyboard mixers* are variations on line mixers and are used in multikeyboard setups. Finally, *that *#$%$^%#% mixer!!* is what you call one that starts to malfunction at the beginning (or worse, in the middle) of a crucial session.

INPUTS

Most pro mixers accommodate balanced-line, XLR connectors (as used with many microphones) and balanced or unbalanced, ¼-inch phone jack connectors. Budget mixers and smaller mixers that need to save space will often include RCA phono jacks or RCA phono and ¼-inch, unbalanced phone jacks.

If you plan to do a lot of recording with microphones, XLR connectors—which are the most common form of connector for pro-level mics—are essential. If you mostly plan to go direct into the board, then XLR connectors and mic preamps assume far less importance. You can buy separate mic preamps with XLR inputs and patch them into your board as needed.

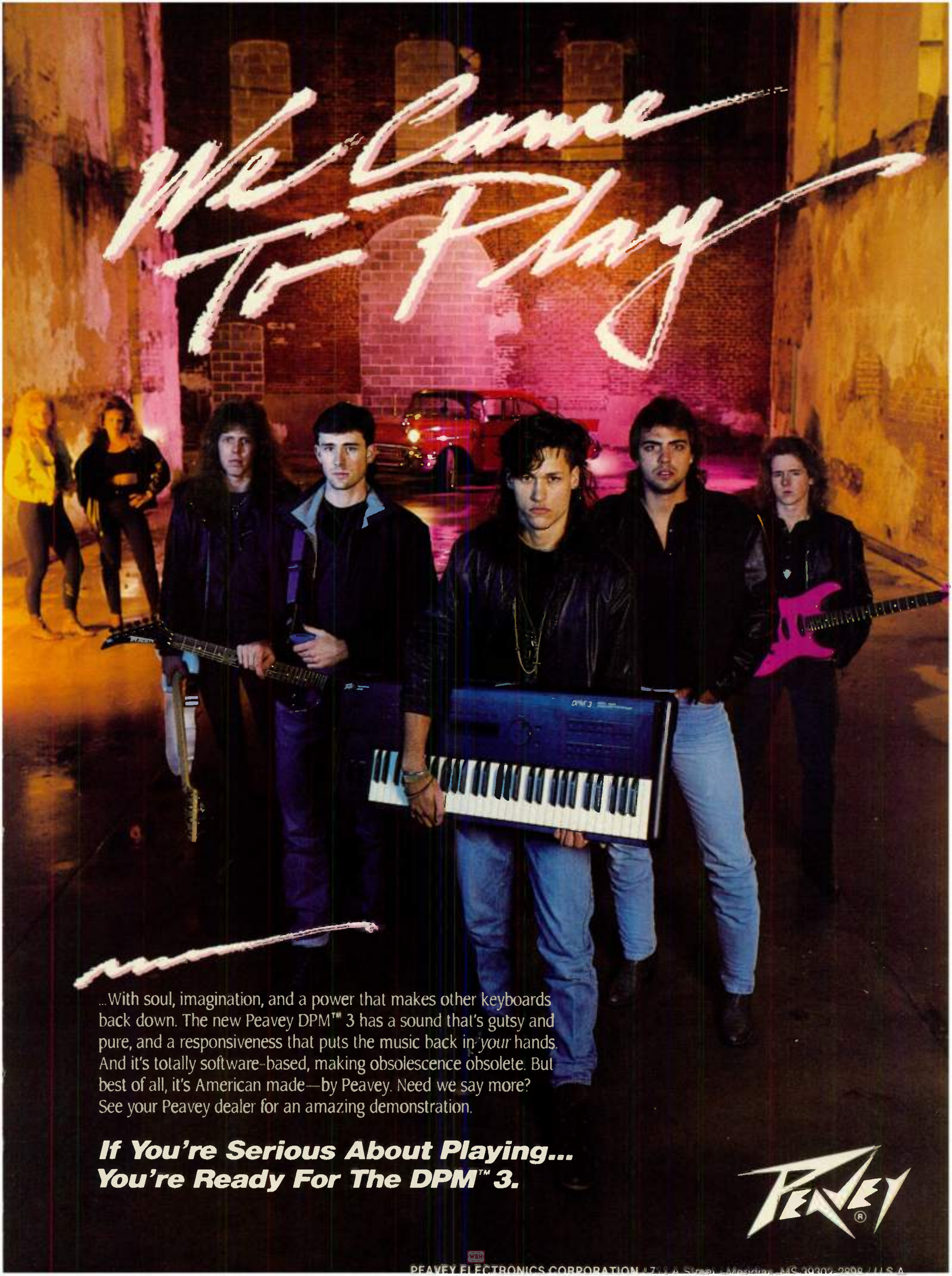
You also may find some *insert* jacks somewhere in the same vicinity as the inputs. These let you patch in a signal processor—equalizer, delay line, etc.—right into the input module signal path.

A recent and welcome trend is the inclusion of stereo inputs. These serve two functions. With instruments having stereo outputs, both channels run independently through the same module, allowing them to take advantage of some common input module circuitry, such as the channel fader. Alternatively, you can feed twice as many mono inputs into an equivalent amount of mixer rear panel space (although both may end up being controlled by the same fader). As more and more synthesizers include multiple outputs, users are crying for more inputs; stereo inputs help address this problem. With stereo input modules, you may also find a stereo image (width) control. This lets you narrow or widen the stereo image from full stereo to mono.

How many inputs are necessary? *Always count on needing more than you think you'll need.* Your system will expand over time, but your mixer will stay the same; plan ahead. If you don't have the space or budget for a lot of inputs, then consider the concept of *distributed mixing*. For example, you can dedicate a small mixer to submixing eight discrete drum machine inputs to stereo, then just feed that stereo line to your main mixer. Or, you could submix the outputs from a multikeyboard setup, and run them into the main mixer as a stereo pair.

Another consideration is an input-select switch. Smaller mixers will often include a switch that lets you select between a line or mic signal and a tape-track output. When recording, you set these switches to select input signals; when mixing or overdubbing, you monitor tape tracks as needed. Big-time consoles usually feature *split* configurations, where the input modules and tape-monitor modules are physically separate. This minimizes repatching and speeds up overall operation but increases the mixer's size and price tag.

Sophisticated consoles offer equally sophisticated routing options, such as buttons that route the input-module signal (after any input-module signal processing) to individual tape tracks, subgrouping options that allow input-module signals to appear magically, mixed to a bus that returns to a specific fader (e.g., routing three inputs to bus 7 will let fader 7 control the subgroup level), a display to show what inputs are selected, and the like. Subgrouping may also allow for easy track-bouncing if you can group a bunch of inputs together and send them directly to tape rather than have to assign them to the stereo bus and then route that to tape. Most of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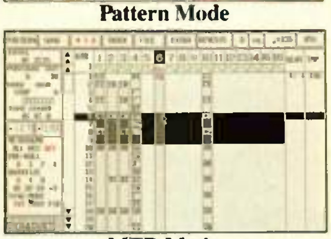
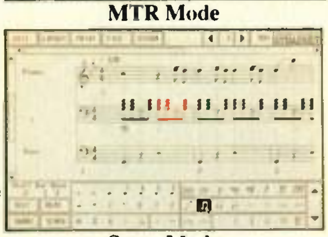


MTR Mode



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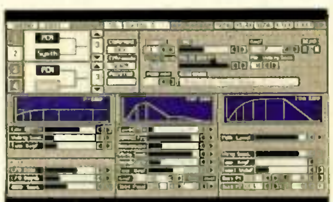

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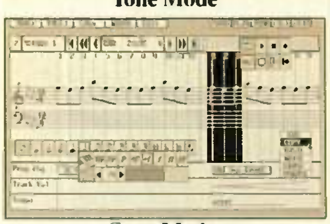

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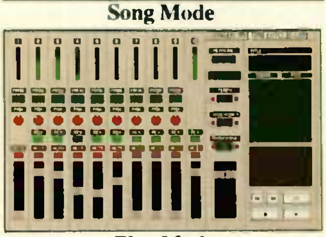

Play Mode

DYNAWARE™

these features come under the time- or labor-saving category; pushing patch cords around will often accomplish the same tasks on lower-priced mixers.

POLARITY (PHASE) SWITCH

XLR wiring is easy to goof up, and making a mistake usually reverses a microphone's polarity from what it should be. (By *polarity* I mean that the crest of an audio waveform produces a positive voltage at the microphone output, and the trough of an audio waveform produces a negative voltage output. If the opposite is true, then the polarity is said to be reversed. This is also called a *phase reversal*, hence the term "phase switch"; strictly speaking, though, phase-shift is a more complex phenomenon, and most engineers use the term polarity to indicate whether positive audio wave motion produces a positive or negative voltage output from a microphone or other transducer.) Odds are one of the mics in a mic collection has reversed polarity, which you'll discover as soon as you mic an instrument with two mics and find that mixing them together actually makes the sound *thinner*, not fatter. Flip the polarity switch, and if the sound gets fuller, you'll be real happy your console has this feature.

> *Designing a mic preamp that delivers lots of gain with very little noise is difficult and expensive.*

MIC PREAMP

Mics put out very weak signals, so they require preamplification before their level will match that of an amplifier or tape recorder input. Unfortunately, designing a preamp that delivers lots of gain with very little noise is difficult and expensive; it's easy to obtain satisfactory performance, but truly excellent performance is another story.

The most important mic preamp spec is its signal-to-noise (S/N) ratio; the higher the number, the better. An S/N ratio of -90 dB or better is very good; -100 dB or better is superb. With budget mixers, figures of -70 dB and -80 dB are more common. However, this is much lower than the noise of most associated budget equipment, so the mic preamp noise is usually lost among the tape hiss, signal processors, mastering deck hiss, etc.

Another way to measure noise is called *equivalent input noise* (EIN). This measures the noise at the output of a system (usually at the mixer output) and subtracts the amount of gain to determine how much input noise would have been required to yield a given amount of output noise. Any resistance at the preamp input contributes noise, and at room temperature with a 150-ohm termination, you're not going to get an EIN spec much better than -125 dB or so. (Beware of shorted-input specs; they're not "real-world".) The closer the EIN comes to -125, the better.

Some mic preamps use transformers to convert balanced-line signals into higher-level signals as well as provide appropriate impedance matching, whereas others use transformerless active circuitry. There are many tradeoffs here, and sometimes the controversy can be as passionate as "tubes vs. transistors." A transformer adds no noise and can offer higher rejection of common mode signals such as dimmer hash and noise, but may color the sound (especially if you combine an inexpensive transformer with a set of "golden ears"). An active circuit, no matter how well-designed, will add some noise but will often sound more "transparent" compared to a transformer of equivalent cost. There are many types of active circuits: IC amps, hybrid amps, and discrete amps each have their own merits and demerits (although to restore some perspective, no radio program director will ever say, "Well, I was going to put your record in heavy rotation, but you didn't use discrete differential mic preamps…").

To further complicate matters, some *prefer* the coloration that transformers add, feeling that it makes the sound "warmer." This is a very subjective kind of call, which is why I designed my own mixer without mic transformers; I can patch in the type of preamp that seems most appropriate for a given job. Sometimes I use transformers and sometimes active circuits.

You should also find some kind of clipping indicator and a *low-frequency rolloff filter*. The latter is typically a very sharp filter that cuts off around 40 to 80 Hz to help eliminate mic "popping," boominess in certain instruments, room rumble, 60 Hz hum, and the like. An equivalent *highpass filter* that cuts off sharply at around 10 kHz or so can be used to cut down a bit on hiss or sibilance. Of course, these duplicate equalizer functions to a certain extent, but you may want to use the equalizer to do something more glamorous than just get rid of hum or hiss.

You may also find a switch to enable or disable phantom power to the mic input jack (useless if none of your mics requires phantom power) and a pad switch to attenuate high-output signals before they encounter the preamp. Why attenuate a signal going to something that's supposed to amplify that signal? Two reasons. First, a very high-level signal might overload the preamp. Second, it's difficult to design a high-gain preamp that handles unity gain elegantly. It makes more sense to run the preamp with a moderate amount of gain and use a pad to cut down the signal.

EQUALIZATION

An equalizer is a sophisticated tone-control circuit. When mixing, equalization (EQ) lets you make sure that instruments don't conflict with each other from the standpoint of frequency response. For example, having a vocalist accompanied by a guitar can lead to problems since both are midrange instruments; cutting the guitar's midrange allows more room for the vocals. This is a very simplistic example; in a real-world mix, the frequency spectrum is quite dense, and properly equalizing everything is a fine art.

Equalization is also important because the transducers used to record musical instruments (i.e., microphones, guitar pickups, contact mics, and the like) very seldom have a flat frequency response. The equalizer can make this frequency response "equal" (hence the name), or "flat," throughout the audio spectrum. However, I suspect that even if all transducers were flat, people would still be just as enamored of EQ. One guitar, processed through twenty different equalization settings, can sound like twenty different guitars.

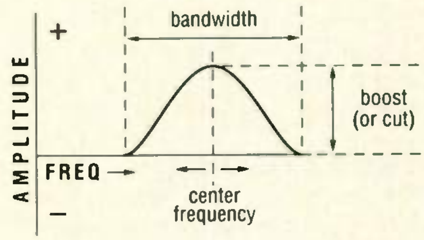

FIG. 2: Parametric equalization

The ideal equalizer would be able to put any kind of boost, or any kind of cut, anywhere in the audio spectrum, adding no noise and affecting the natural sound of the instrument as little as possible (no distortion, no strange colorations, and so on). This is a tall order.

The simplest type of equalizer simply boosts or cuts at two or three fixed points in the audio spectrum (say, 100 Hz, 1 kHz, and 10 kHz, ±12 to 15 dB). The high and low ends will generally be *shelving* equalizers that increase or decrease the level above and below the high- and low-frequency points, respectively. The midrange will boost and cut at that specific frequency only. These are not terribly useful except to add some bass, treble, and midrange (although this is sometimes all you really need, anyway). However, if you want to boost the midrange on six channels, with the example above they'll all boost at 1 kHz, which will create quite a buildup of energy at one specific frequency band and tend to make the sound worse, not better.

A *quasi-parametric*, which has a variable frequency where the boost or cut occurs, is much more desirable. The reason for the "quasi" prefix is that there will not be a *bandwidth* control; in other words, you cannot determine the range, in octaves, over which the boost or cut will occur. This is a job for the "true" parametric equalizer (**Fig. 2**). While very versatile, it takes restraint and experience on the part of the parametric operator to get natural-sounding effects, since there are so many options available. Some people will use EQ just because it's there; these folks can be really dangerous with a parametric in their hands.

A large console typically has three or four bands of parametric EQ, or perhaps two bands of parametric EQ and two shelving bands at the high and low

continued on page 82

"In blindfold listening tests with the best software sequencers, the Alesis MMT-8 won hands down for the best feel."

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On Sound

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AS MUSICIANS, what we're ultimately trying to do is create an interesting group of sounds. In musical terms, that's an obvious statement, but let's look at it from a more technical standpoint: When we compose, arrange and play music, what we're striving for is an ear-catching waveform.

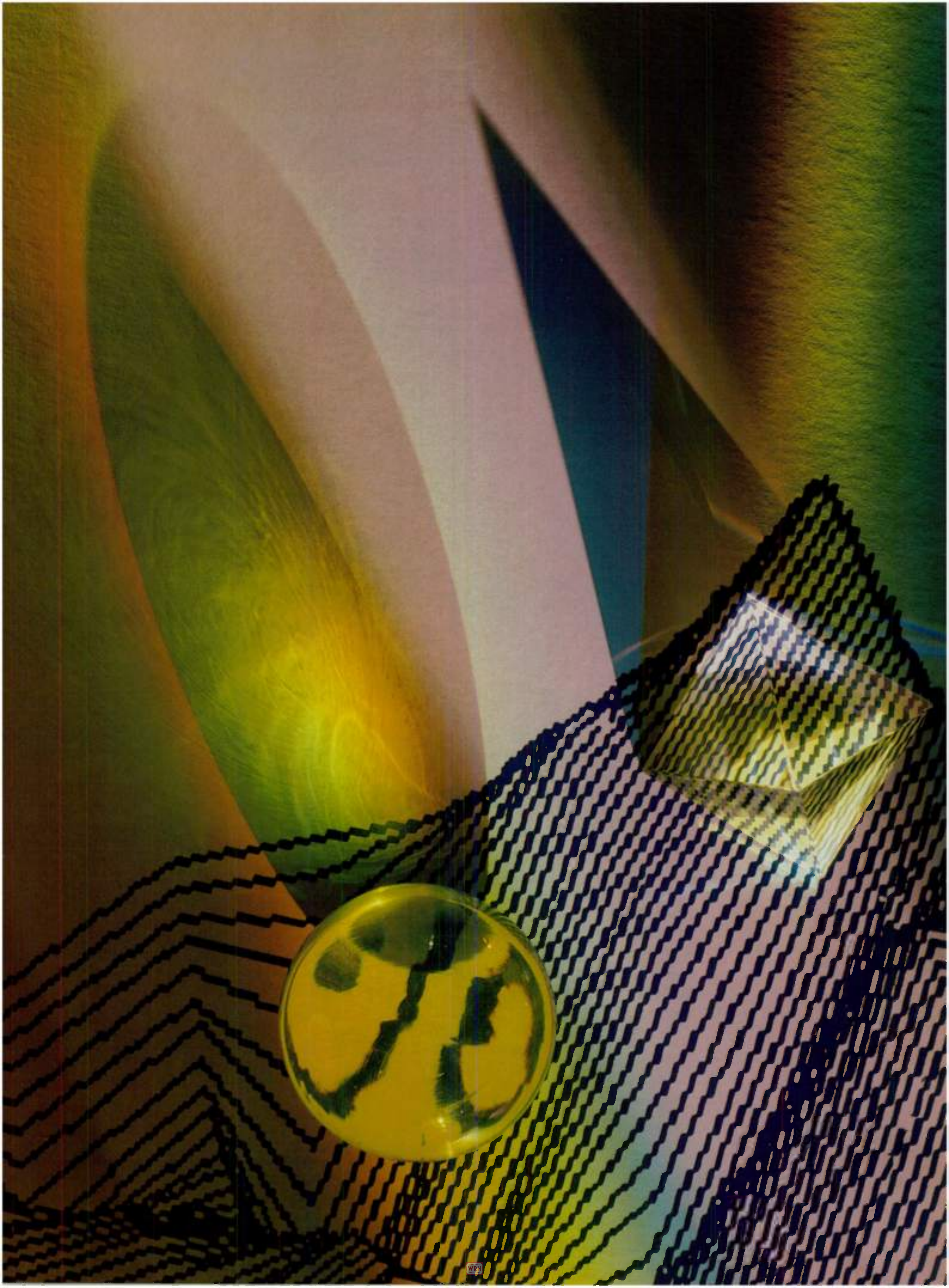
Musicians of every age have dealt with sound as a musical parameter, but thanks to recent technological developments, the idea has been brought to the fore. We now have tools that can generate and shape sound in just about any way imaginable, and that has forced us to think about sound in new ways.

We can understand this in terms of other art forms as well. As the artist has paint, the sculptor clay, the writer words, so the musician has sound. Yet how much time do we spend thinking about our raw materials? By omitting this step and hence, missing the sense of wonder that often accompanies a thoughtful reflection on essential elements of any art form, our ability to grow artistically from a solid understanding of the fundamentals is limited.

Sound, not composition per se, may be the desired end result of our musical labors; by concentrating on notes, we just might be missing the forest for the trees. Similarly, by not recognizing the raw materials, it's often difficult to see the degree of connectedness that exists between all the various pieces of equipment we use and the music itself. How can we be effective musicians if we don't understand the relationship between synthesizers, signal processors, waveforms, timbre, orchestration, and the actual music we compose?

PIERRE-YVES GOAVEC

By Bob O'Donnell, Craig Anderton, and Jim Conger



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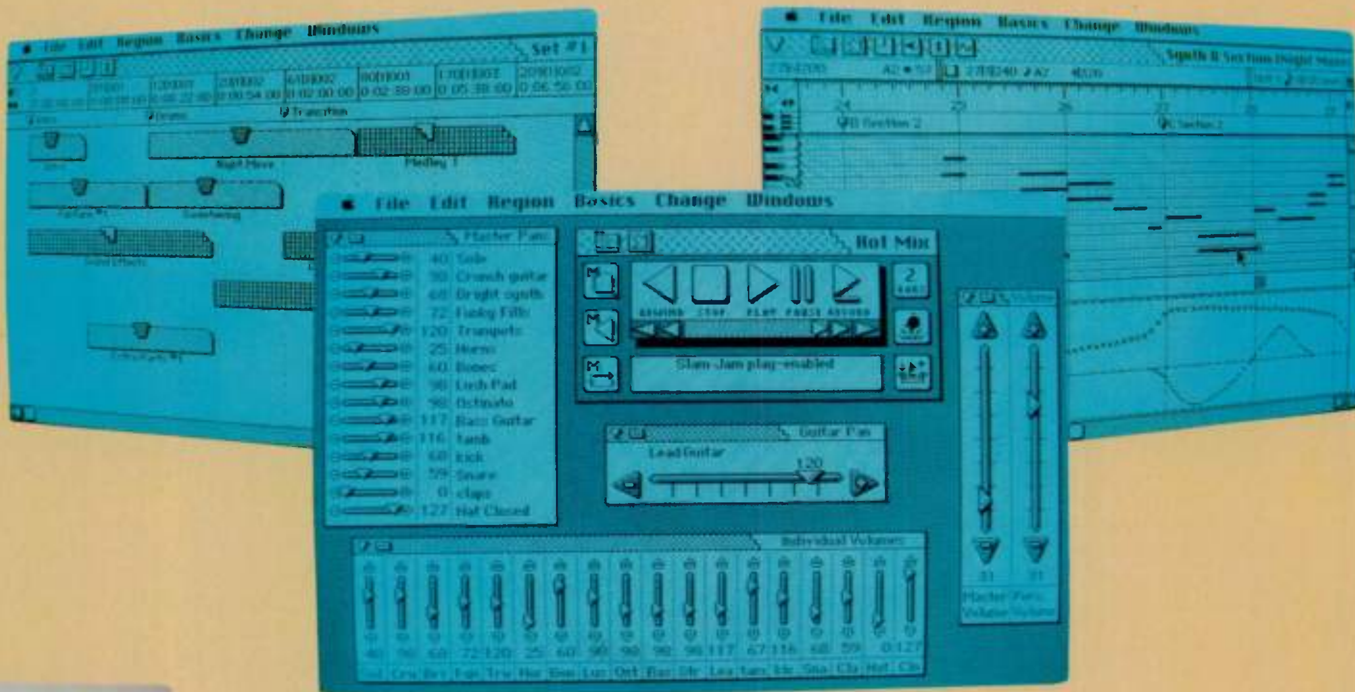

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In the context of an arrangement, a sound does not exist on its own, but as a complement to existing sounds. Sometimes a thin, reedy sound is just perfect if the background is dense and full.

☛ You're using a very simple-sounding patch to create the chord progression you need for a melody line you've composed earlier. When later orchestrating the changes with a more polished sound you discover that your beloved Amaj9 chord now sounds dissonant. Those same notes sounded fine before, so what's the problem?

The more polished sound has a harmonic spectrum that emphasizes frequencies which don't correspond to notes within the chord. By playing a simple major chord instead, you may be able to achieve the original effect.

☛ Your income tax refund has just arrived, and your setup really needs beefing up. You love your PDQ-100 digital synth, so you go out and buy the rackmount version. Funny, but it doesn't provide the sonic nirvana you hoped for. Why? Because it duplicated your sonic vocabulary instead of adding to it. You might have been better off getting an analog synth and a used CZ-101, both of which use a different method of synthesis and can provide contrast.

☛ It's the twentieth time the guitar player has laid down a lead, and things still aren't right. You've suggested playing slower, playing faster, doing block chords instead of single notes, but nothing seems to be working. Perhaps the problem has nothing to do with the notes, but with the sound.

☛ A guitar/bass/drums rhythm track sounds kind of thin, so you figure it's time to add another instrument—maybe a little background keyboard, or some string pads. But the solution may really be adding a little EQ to fatten up the existing sounds, rather than adding more notes.

☛ Your mix simply isn't working. You don't have extensive EQ, but you feel the need to adjust the overall sound of the piece, so you edit each of the individual patches you're using, lowering filter-cutoff points, adjusting operator output levels, etc., in order to remove a portion of the frequency spectrum that's clogging up the midrange. All of a sudden the piece breathes and works as you had envisioned.

In all these cases and many more like them, you're learning how the fundamental elements of sound—harmonics—are affected by various types of gear and/or procedures. (For more information see "Harmonics: The Basis of Sound" on pg. 76.)

On the hardware side, every piece of electronic music gear (that is, those with audio inputs or outputs) affects the waveforms of the music you're creating. Essentially, the gear either creates or modifies harmonics, the particular method for doing each is dependent on the type of gear. As an example, graphic equalizers allow you to increase or decrease the amplitude of all harmonics that occur within particular bands of frequencies. A lowpass filter within a synthesizer or sampler, on the other hand, does a broadband reduction of all the harmonics in a waveform that occur above the filter's cutoff frequency.

Once you start to understand this—as well as figure out what types of products perform what kinds of functions and how—you should be able to get a better total picture of your capabilities as an electronic composer. First, though, it's important to understand the relationships between various musical, physical, and technical concepts.

THE CONNECTIONS

Though it's often far from apparent, you can tie music theory concepts together with physical and technical ones. Pitch, for example, is primarily a musical concept, but it can be described physically and technically by giving a particular frequency for the pitch's fundamental. Middle C has a fundamental frequency of 261.63 Hz.

What this means in practical experience is if you run a sequence of notes (using a simple timbre) through an equalizer in which the amplitude of a specific frequency has been greatly increased, the note whose fundamental pitch corresponds to that frequency will sound louder than the other notes. If you needed to consistently emphasize a particular note throughout a song, you probably wouldn't find this method to be particularly intuitive. In many cases it would be easier to increase the velocity level (which, in most synth patches, is programmed to adjust amplitude) for all occurrences of that pitch within the sequence. In the first case, you would only emphasize the fundamental frequency of the note, while in the second, you would emphasize all the harmonics of a given timbre that happened to play that note. Regardless, it's interesting to consider that two completely different approaches could produce a similar end result.

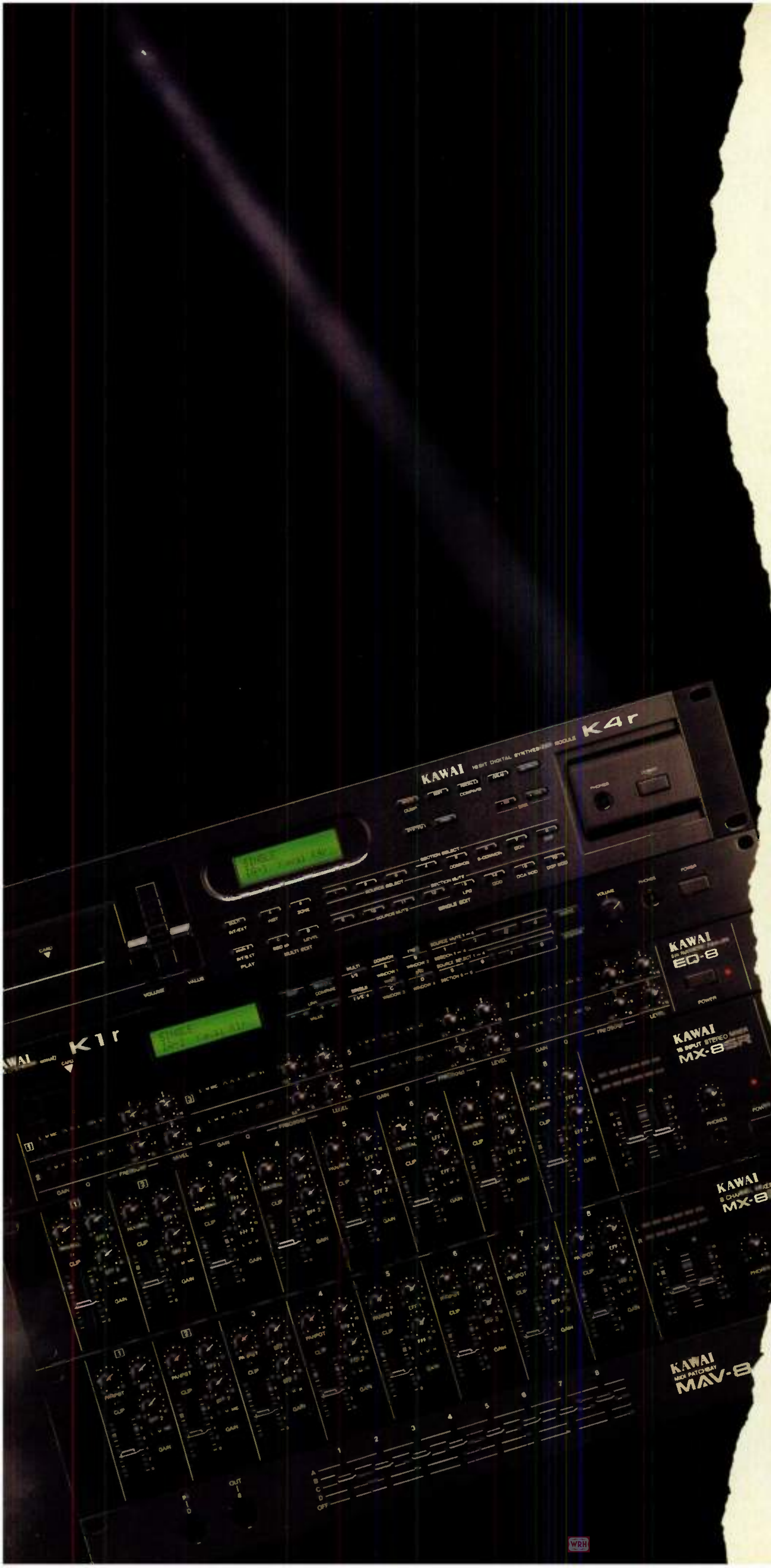
Likewise, chords can be understood on many levels. Musically, of course, a chord is a collection of notes that sound at the same time. Physically, a chord is the combination of all the frequencies for a given group of notes with a given timbre, or tone color. Technically, a chord could be understood and heard as a single note with a given timbre. The reason for this is that timbre is determined by the combination of harmonics, at particular frequencies, that go into making a given sound. For example, by playing a large, widely voiced chord using sine waves, you could create a waveform that began to sound like a sawtooth wave at a single pitch (the chord's fundamental). Thus, you can think of timbre as "chords of harmonics."

> *One way of thinking about timbre is as "chords of harmonics."*

The implications of this are many. If you accept the principle that pleasing sounds cover a wide frequency range without placing too much emphasis on any one particular area, you can get a more scientific explanation of why some sounds seem to work in certain situations and not in others. Various timbres or groups of notes can overload a particular frequency range and cause a particular section of music to just plain sound bad.

For example, if you have a piece that needs something on the high end to fill it out, you can take one of many different actions. Conceivably, you could use an analog synth and play some notes in the upper registers (in other words, notes that have high fundamental frequencies). Because analog synths create their sounds by removing different amounts of high-level harmonics from a full-frequency waveform, however, this may end up being too bright and cutting. High harmonics above a high fundamental frequency will probably be overwhelming.

Instead, you may want to consider playing the same notes a few octaves lower, using a very bright sound that has a strong high-frequency content but a dip in the level of mid-frequency harmonics. An FM synth such as a DX7 would be a good choice because it's capable of creating that type of frequency spectrum. Even better would be an additive synthesizer, which would allow you to select precisely the harmonics you want to include in a particular sound as



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well as control their level. (For more on additive and other new forms of synthesis, see "Synthesis Techniques for the 1990s and Beyond" on pg. 62.) A sampler or sample-player might also work, but the raw, sampled waveform would have to fit into the frequency range described. You generally don't have as much flexibility adjusting a sound's timbre with a sampler as you do with a synth.

Orchestration, then, or the process of determining which instruments should be used to play which musical lines, takes on new meanings when it comes to electronic music. Back in the days before electronic instruments, if you wanted more treble, you wrote in a flute part; if you wanted more bass, then the contrabass did its thing. When listening to Bach, many people are impressed by the masterful use of harmony and counterpoint. But could it be that the effectiveness of these is augmented, if not made possible, by the sounds chosen to play those harmonies? Bach managed to balance not just notes, but sounds—as did all the great composers.

As a composer who electronically orchestrates his or her musical ideas, you need to concern yourself with the notes you play as well as the timbres you use. These concepts always have been and will continue to be fundamentally related. It's just that as an electronic composer, you have more ultimate control over the final combination they create: the music, or to think about it another way, the sound.

MUSIC AS SOUND

To many people, the idea that music is simply sound is a bit difficult to accept. Nevertheless, definitions of music usually describe it as an organized group of sounds that occur over time and attempt to express feelings. We're not going to attempt to address the last point because that boils down to a question of taste; instead, we'll address the former.

The notion of music as sound is really brought home by digital samplers and other digital recording systems. Though samplers were initially thought of as devices with which you recorded single notes of particular instruments and played them back, they quickly grew into much more. To a sampler, a single piano note is no different than a passage from a Beethoven symphony. Philosophically, the difference between the two seems enormous, yet the unblindingly objective technology reduces them both to a single waveform. Aesthetic questions aside, it's important to recognize this fact. Also, consider a piece of music that contains nothing more than a single note, or even no notes at all, yet whose sonic elements change and grow over time. (It's an idea that hasn't been lost on synth programmers either. Many newer synths have enormous, one-finger presets that evolve over a long stretch of time; they practically exist as pieces by themselves.) Is such a composition any less musical than a traditional song?

Many twentieth-century composers have used this duality to their advantage and produced works that build on the converse of this statement. If music is sound, they reason, then sound is music. John Cage and many other avant-garde composers have produced and continue to produce "sonic music" that bears little relation to the melody and harmony-based music that historically preceded it. Most of the works have never had much mainstream success because many people want their music in more sonically palatable and recognizable forms, but it's easy to see the logical reasoning behind their compositions. In addition, they've had a tremendous impact on popular music. Many Top 40 singles start off with "pure sound" introductions, designed to catch the listener's attention, and then segue into a strong rhythmic pattern. Atonal effects and sound purely for sound's sake are showing up to a greater degree in today's music. In many cases, new age music, with its lush pads and lack of structure, places an emphasis on sound over compositional form.

Music and sound are entwined inextricably and shouldn't be separated into their own "boxes." Sheet music doesn't tell you much about timbre, and patches don't tell you much about which notes work best with that patch. You may be able to make good judgments, based on your musical sensibilities as well as your ears, but it's important to know why certain things work and others don't.

You should also be able to apply these ideas to real-world examples. What follows is a brief look at applications.

THE SHAPING OF SOUND

Signal Processing

When you call up a reverb algorithm on your digital synth, you're not patching in a reverb. Instead, the synth's computer is processing the numbers that generate sound to make reverb a part of that sound. Echo is no longer necessarily a function of an acoustic space, but a function of a few hundred lines of code that tell a computer to spit out the same bunch of numbers several times, at a periodic rate, and (in all likelihood) with less level for each pass. The effect becomes an integrated part of the waveform generated by the instrument.

So what's the difference between processing your synth's audio signal through a fancy effects box and modifying the patch settings on the synth to get the same sound? Practically speaking, there's no difference whatsoever. Nor is there much remaining difference between sound generation and sound processing, a point made obvious by computer-based, software-configurable music systems.

Or consider multi-effects boxes. One might think they contain a separate fuzz, equalizer, reverb, etc., but in truth, there's just one multifaceted program that emulates all those effects through digital processing. Want to add chorus? Don't think about patch cords; call up a subroutine instead.

Mixing

Mixing is an excellent example of sound being an integral part of the music. Equalizing, balancing instrument levels, and adding special effects combine to work on arrangements (i.e., notes, chords, etc.) *after the fact* to create a proper sonic balance. (For more on the philosophies and techniques of mixing, see the July 1989 EM.) You have to avoid loading up the highs or lows, and you can't permit several voices to "clutter" a particular part of the frequency spectrum. With a good mix, sound is manipulated to provide spaciousness and clarity, even when there's an orchestra's worth of instruments playing.

Multitasking

Some software companies make a big deal about the fact that you can let a sequence run while opening a patch editor to alter sounds. We know intuitively that this makes sense, but now we know intellectually as well: you can't tweak a sound without knowing its context. Multitasking also means being able to edit MIDI signal processors and other tools in real time, while a sequence is playing, to make sure that the timbres all complement each other.

• ON SOUND

Purchasing Equipment

One of the biggest problems facing today's electronic musician is how to get the most value for one's equipment dollar. In light of the above, the answer is simple: go for variety of sound. If you have a sampler, get a synthesizer. If you have an FM synthesizer, get an analog one. This way you can pick the proper timbre from a wide palette to complement existing sounds.

Equally important, consider your entire collection of equipment and software as a system. Think of your studio as one giant "additive synthesizer," where different harmonics (sounds) are brought in at different levels to create a composite effect. All the production steps are parts of a whole, from designing the studio, creating a proper gain structure (setting the input and output levels of each piece of audio gear to minimize unwanted distortion and maximize electronic efficiency throughout the signal path)and generating a raw waveform, to the finished sound heard in the listening room (remember, the room is part of the system, too). In a sense, you can orchestrate the studio itself.

THE FINAL NOTE

Throughout the article we've been expounding on a simple, single premise: Music is a collection of sounds. On the surface it seems absurdly simple, yet by digging a bit further into the notion, a wealth of possible opportunities, applications, and solutions can be mined from it. None of these solutions will make a bad melody a better one, but the kind of subtle differences that can be made may make the difference between a good recording and a great one.

Ultimately, what we're talking about is problem-solving, but not by applying any magic formulas. The approach offered here is to *think* about sound as music and music as sound and to approach the two in a complementary, integrated fashion.

No previous generation of composers has had such easy or complete sonic control as that offered by electronic instruments. By being aware of the connection between all the various elements of sound and music, and hence, better understanding the equipment we use, we can best take advantage of those opportunities and achieve the goal toward which we're all striving: creating satisfying, personal music. ■

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synthesis techniques

for the 1990s and Beyond

Many ideas presented here assume a familiarity with the building blocks of sound. For a discussion of the basics, refer to "Harmonics: The Basics of Sound," page 76.

TO SAY THAT SYNTHESIZERS have completely redefined the modern music-making process is hardly a revolutionary statement. Their evolution as musical instruments has been so dramatic that for many people it seems difficult to remember musical life without them. In fact, certain new types of music wouldn't have come into being if synths hadn't achieved acceptance.

One of the main reasons for their growth has been the development of increasingly sophisticated new synthesis methods. From the early days of analog instruments, through FM and sampling, synthesizers have progressed to the point where they are capable of creating an enormous variety of rich, complex sounds. But there's still room for further growth, and many people want more.

For certain individuals, the ultimate goal of any synthesis method is the ability to re-create the ever-elusive, life-like quality of acoustic instruments. Others don't necessarily want re-creations, but they want to be able to generate completely electronic sounds with the same degree of depth and sophistication.

Existing synthesis methods have had a certain degree of success with achieving these levels of sonic complexity, but thanks to recent advancements in computing speed and power, a new round of even more advanced, real-time, sound-generating techniques is on the horizon. Specifically, additive synthesis, resynthesis, physical modeling, and granular synthesis are starting to make their presence felt both in the R&D labs of major synth manufacturers and, to a certain degree, in the music stores.

It may be a few years before instruments that use these new synthesis methods start to appear in force, but when they do, it will be an exciting time for electronic musicians working toward either type of sonic goal.

By Peter Gross

SOME BACKGROUND

Before jumping into descriptions of how these new synthesis methods create sound, it's important to recall a few basic points and some history. First, remember that all synthesis methods ultimately do the same thing: they create different sets of harmonics that we hear as sounds. Each method approaches the end result in a different way, and each offers different types of shortcuts, but they all share a similar foundation. Also note that certain methods make use of other methods in the process of creating sounds. Resynthesis, for example, uses elements of sampling and additive synthesis, and many methods make use of subtractive synthesis techniques or build on other existing technologies—which is why you need a bit of history...

Early on, commercial analog synthesizers most commonly used *subtractive synthesis*. (Modular instruments offered additional capabilities; I'm referring to hard-wired instruments such as the Minimoog, the Prophet-V, etc.). In subtractive synthesis, a filter removes (or subtracts) certain portions of a waveform's harmonic spectrum to make adjustments in a sound's tone color, or timbre (see **Fig. 1**).

In spite of its simplicity, the hard-wired, analog, subtractive method proved flexible and pseudo-intuitive, though it had limitations. One important limitation was that most instruments' oscillators could only produce a handful of harmonically rich, though not very complex, waveforms. Manufacturers responded by adding greater numbers of oscillators, more elaborate filtering combinations, more envelope generators, and a greater variety of oscillator waveforms until the point of diminishing returns was reached. Many synthesizers became difficult to program and yet they still generated sounds that were similar to other subtractive synthesizers (despite recognizable "signatures"). Getting the newest and greatest sound on the block became increasingly difficult.

REALITY CHECK

Real-world sounds from acoustic musical instruments have harmonic spectra that can be extremely dense, and the ratios between harmonics are not constant over the pitch range of an instrument. On top of that, the harmonic spectrum of an acoustic musical instrument is rarely static; an acoustic instrument tends to have unique spectra during the attack, sustain, and release parts of its sound.

To confuse the situation even more, emphasis by a performer, such as blowing into a clarinet harder, tends not only to alter the volume of the instrument, but also the ratio of harmonic intervals (see **Fig. 2**), noise components, and other complex factors. A clarinet note, for example, has an attack spectrum (when the reed first encounters air from the musician's lips) that is completely different from the sustain spectrum and the release spectrum. As you play a scale from a low note to an octave or so up, the ascending notes each have their own unique harmonic ratios. So simply tuning all the harmonics of a low note up to get a higher note doesn't work. (Samplers have this problem. Have you ever noticed that a sampled musical instrument is only credible for one or two notes around the sampled pitch? The human brain is just too good at interpreting harmonic structures for samplers to be effective with one or two samples per instrument.)

For practical purposes, analog subtractive synthesis is not capable of dealing with the demands of control and harmonic density necessary to create sounds as pleasing to our ears as acoustic musical instruments. Unless you could get a machine to turn the controls of 200 Minimoogs with microsecond accuracy, you have to accept the limitations of subtractive synthesis and examine additional synthesis techniques.

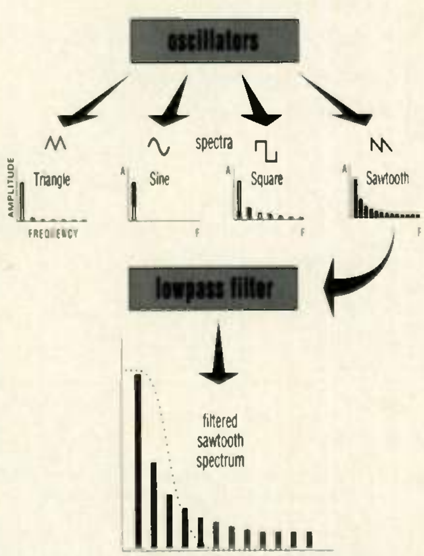


FIG. 1: In analog subtractive synthesis, a filter removes harmonics from an oscillator-generated waveform.

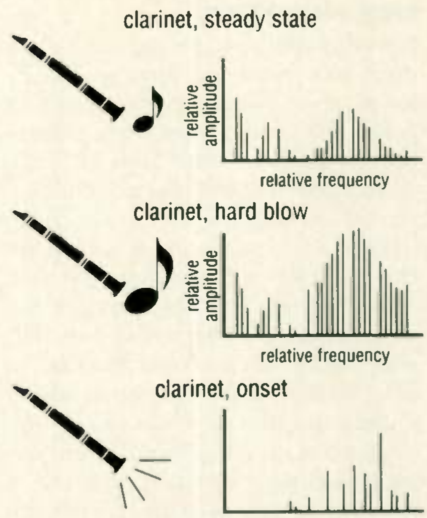


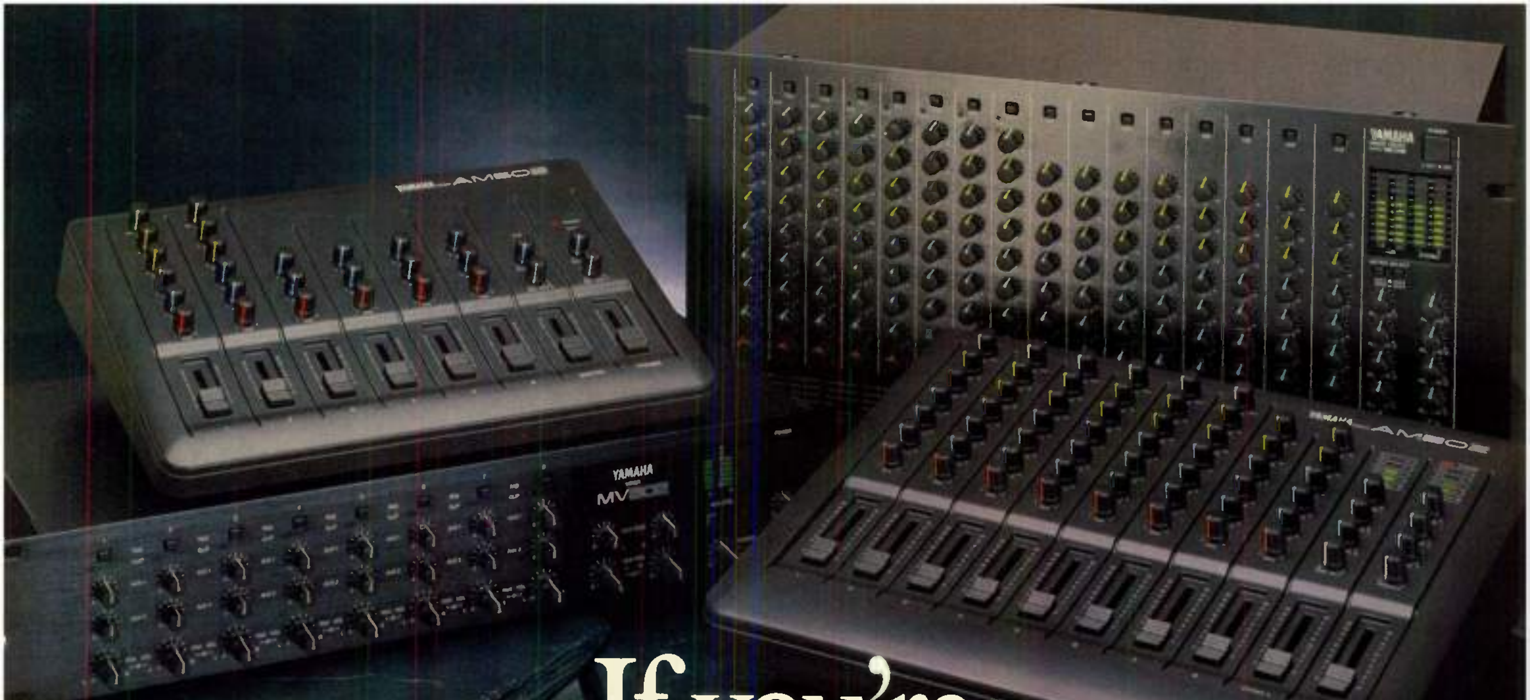
FIG. 2: The harmonic spectrum of an acoustic instrument is rarely static. Most acoustic instruments have unique onset, steady-state, and sustain spectra.

ADDITIVE SYNTHESIS AND RESYNTHESIS

The solution to our synthesis limitations comes from a 19th-century French mathematician named Fourier. The Fourier theorem states that *any sound* can be described as a series of *harmonics* in the form of sine waves of various frequencies, amplitudes, and phase relationships. Instruments that use this synthesis method, *additive synthesizers*, create musical waveforms by adding large numbers of sine waves with precise frequencies and amplitudes whose values are adjusted over time with envelope generators. Normally, no filters are used on an additive synthesizer, though some do offer them for subtractive sound-shaping capabilities. If you have a synthesizer capable of precisely controlling the amplitude and frequency of a hundred or so sine waves, you essentially have no limitations on the types of sound you can produce.

So if one synthesis method can re-create every sound ever heard and any sound we're capable of hearing in the future, then why bother with others? Well, there's a bit of fine print here. Fourier requires an infinite number of sine waves in order to completely duplicate sounds accurately—at least in theory. Thankfully, reality is not quite as demanding. Experience shows that getting an easily recognizable re-creation of a familiar sound requires about 30 harmonics (see **Fig. 3**). If you use around 100 harmonics, the synthesized wave gets

(turn to page 68)



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PLAYERS IN THE ADDITIVE SYNTHESIS GAME

Of all the synthesis methods discussed in the article, additive synthesizers and resynthesizers are the only types of instruments that have actually made it to market. Here's a list of currently available products that make use of either additive synthesis or resynthesis.

Lyre FDSS Studio

The Lyre (tel. [418] 654-0181) FDSS is a rack-mount unit that requires a computer for its user interface. Presently, the Lyre system speaks IBM, but a Macintosh version is planned for the first quarter of 1990. Resynthesis ability is also part of the standard system.

The basic $5,000 unit gives you access to 128 harmonics, expandable to 256. Additional expansion units can be added, up to 1,024 harmonics. Each harmonic has its own amplitude and frequency deviation envelope, and each envelope is made up of 128 segments (as opposed to the familiar four segments: attack, decay, sustain, and release). Accurate control is necessary with any additive synthesizer, and the FDSS resolves frequency to .0001 Hz and amplitude to increments of .0001 of total level. Envelopes are edited graphically on the computer, as are all functions of the FDSS. Digital-to-analog conversion is at 44.1 kHz, with 16-bit oversampling.

An interesting feature is a MIDI-controllable, real-time, lowpass filter that offers a subtractive synthesis postmortem for any sound your additive synthesis generates.

Resynthesis is done on the control computer, so if you want to do lots of 256-harmonic analyses, either buy lots of coffee or buy the fastest computer your bank account will allow. However, the Lyre folks are presently working on speeding up this process with an add-in card to move resynthesis to the world of real time. FDSS offers MIDI and supports Apple's AIFF (Audio Interchange File Format), which allows communication with software such as Digidesign's *Sound Designer* or Blank Software's *Alchemy*. No question about it, this is serious hardware.

Technos Acxel Resynthesizer

The Technos (tel. [418] 835-1416) Acxel consists of a large NeXT-like black box called "the Solitary" with a briefcase-like user interface called "the Grapher." The user interface is essentially a high-density LED touch screen, with buttons for common functions surrounding the LED array. Everything is done by glancing your finger across the touch screen. Waveforms can be drawn, harmonics turned on/off, all by touching them with your finger. This is by far the most sophisticated machine/human interface I have seen for any synthesizer.

The Acxel doesn't speak of oscillators and envelopes in the conventional sense. Instead, it uses "synthesis cells," each containing a digital oscillator, a digital amplifier, a pitch envelope generator, and a volume envelope generator. Up to 1,024 synthesis cells are possible, depending on the system you order. All envelopes can be edited on the touch screen, pitch with .01 Hz accuracy, and time to .1 second accuracy. Envelopes can be zoomed, unzoomed, combined, edited, etc.

The Techos Acxel

In addition, the Acxel can resynthesize very quickly. Incoming audio is analyzed and shown on the touch screen. The machine can then program itself to reproduce the incoming audio. Technos calls this resynthesis process "Acxelization."

Other functions include post-filtering (lowpass, highpass, bandpass) in an implementation of subtractive synthesis. Sonic density freaks will like the ability to reprogram the synthesis cells from generating sine waves to playing any hand-drawn waveshape or a choice of preprogrammed waveshapes. The system has multitimbral capability and a full MIDI implementation.

Signal conversion specs and price quotes were not available, but a basic system goes out the door for under $10,000. Yes, it's expensive, but for the price of a professional sampler, you get an extraordinarily powerful machine

Korg DSS-1, Casio FZ-1, E-mu Emax SE and II

Speaking of samplers, these instruments offer limited additive synthesis functions for waveform creation. The Korg DSS-1 (no longer in production) and Casio (tel. [201] 361-5400) FZ-1 ($2,499) samplers only allow you to adjust the initial amplitude of a limited number of harmonic partials, but it is a start. The E-mu (tel. [408] 438-1921) Emax SE HD (no longer in production) and Emax II ($3,495), on the other hand, offer true additive synthesis under the guise of Spectrum Interpolation Digital Synthesis. (For more on the Emax SE HD, see the October 1988 EM.) You can adjust the initial amplitude and pitch of up to 24 harmonics and give each an independent pitch and amplitude and envelope. It takes a while to calculate the waveforms you specify, but you can create some interesting sounds. Now if they were only able to convert the samples they generate into this format...

Kawai K5

Though it doesn't meet the requirements for a "true" additive synthesizer, Kawai's K5 (no longer in production) does offer up to 126 sine wave partials with which to work. You can't adjust the initial frequency of individual partials, nor are pitch envelopes available, but you can turn different groups of harmonics on and thereby create different harmonic spectra. Initial amplitude level adjustments for each harmonic can be made, but each harmonic must be assigned to one of eight 6-stage, user-definable envelopes. Kawai's K3 synthesizer (which, like the K5, is no longer made) offers waveform creation possibilities similar to those found on the Korg DSS-1 and Casio FZ-1.

Software-Based Synthesis Systems

If you already own a sampler and a computer, a software-based, additive synthesis package could get you airborne at minimum cost. Digidesign's (tel. [415] 327-8811) *Softsynth* for the Macintosh and Atari is such a program, and it supports just about every conceivable sampler on the market as well as the MIDI sample dump standard. In the IBM world, Lyre offers *FDSoft* ($250).

No longer available as a separate program, Softsynth is included with Digidesign's Sound Accelerator package ($1,295; for more on the Sound Accelerator, see the review of Digidesign's Sound Tools in the November 1989 EM).

Softsynth allows you to program 32 oscillators, each with a unique amplitude envelope and pitch envelope. (Filtering is also available.) Once you have selected your parameters, Softsynth calculates a wavetable and downloads it into your sampler.

Softsynth also communicates with its family cousin, *Turbosynth* ($349). This package allows complex manipulation of the sample files created by Softsynth to produce an extraordinary array of sonic possibilities. (For more about Turbosynth, see the November 1988 EM.) Many Softsynth features have been incorporated into Turbosynth V. 2.0. The oscillator module includes an FFT (Fast Fourier Transform)-based algorithm that separates a waveform into its spectral components, and the Harmonic editor allows you to adjust each of 64 harmonics individually or in groups.

Lyre's FDSoft supports additive synthesis with up to 128 harmonics and, in addition, is capable of doing resynthesis with samples you load into the program. The samplers supported by FDSoft are a bit limited (though the MIDI sample dump standard is included), but the company recently included a utility that can translate SampleVision format samples into a form FDSoft can read. As with Softsynth, you can create independent pitch and amplitude envelopes for each harmonic—either ones you generate yourself, or those generated by the resynthesis process. (For more on FDSoft, see the review in the December 1988 issue.)

The disadvantage of software-based additive synthesis programs like FDSoft and Softsynth is that they can be quite time-consuming. Calculation of a full, 32-harmonic wavetable can take as long as twenty minutes, depending on the computer you are using. By using the Sound Accelerator card for the Mac, however, Softsynth can perform these operations in real time.

Two sample editing programs, Blank Software's *Alchemy* ($495; tel. [415] 863-9224) for the Macintosh and Steinberg/Jones' *Avalon* ($349; tel. [818] 993-4091) for the Atari ST, also offer some basic resynthesis capabilities. Both programs allow you to perform an FFT on the sample you are currently editing and then adjust the pitch and amplitude of individual harmonics. Again, the calculations can take some time, but the possibilities are endless. (For more on Alchemy, see the February 1989 EM; for more on Avalon, see the December 1989 issue.) Finally, check out this month's First Take of *Wave* (pg. 111), a synthesis program for the Commodore Amiga.

–Peter Gross and Bob O'Donnell

very close to the original wave, and above 250 harmonics, things are just peachy. In fact, I seriously doubt that anyone could tell a 200-plus harmonic re-creation from the real thing. Also keep in mind that the lowest frequency harmonics are most important to our sensory apparatus. As the harmonics grow in number and hence, frequency, their absence is less noticeable.

The biggest problem with additive synthesizers is that dealing with hundreds of harmonics, each of which have envelopes with nearly infinite numbers of break points, adds a new dimension of meaning to the word tedious. The sheer amount of numerical data involved can be completely overwhelming, particularly for the user. What you need is a shortcut, something that would provide an interesting set of harmonics with which you could work. This is where *resynthesis* fits in.

Instruments using resynthesis take advantage of the fact that you can use the Fourier theorem to analyze a musical waveform and deduce from it the constituent harmonics and their frequency/amplitude characteristics. This makes it possible to analyze a musical instrument, say, a note on a piano, and with the assistance of a computer, deduce the harmonic structure of its timbre. Then, by having the same computer automatically program 100 or so sine wave oscillators, you can re-create the original piano timbre.

A resynthesizer works by first sampling a sound and then converting it into the additive synthesis format of sine waves with varying pitch and amplitude envelopes. In theory, resynthesis actually offers the best of both worlds—the "reality" associated with sampling and the flexibility offered by additive synthesis. The process of converting a sample into a similar-sounding collection of sine waves is extraordinarily difficult, however, and the real test for resynthesis machines is how accurately that conversion is done.

In case you're still a bit confused about the connection between samplers and resynthesizers, the fundamental point to remember is that a sampler is dumb. It's like a tape recorder but uses digital memory instead of tape. It doesn't know anything about the timbre of the sound inside its memory and doesn't care. As noted earlier, each note on an acoustic instrument has unique, dynamic harmonic ratios, but sampled notes don't, so they sound artificial when transposed. An additive synthesizer/resynthesizer is different. With resynthesis, you can analyze a sound, recreate it, and then alter it. You could take an additively synthesized piano note that starts off like a piano note and sweep the frequency envelopes all over the place, creating a unique "piano from hell" sound. The degree of change you can impart to a sound is almost unlimited.

Before you start salivating all over the pages of this magazine thinking about what you could do with a resynthesizer, it's important to note that the electronic and computational problems of generating, in real time, 100 or more harmonics, each with a unique amplitude envelope and frequency envelope, are not trivial. Stratospheric computational demands must be met with finesse to

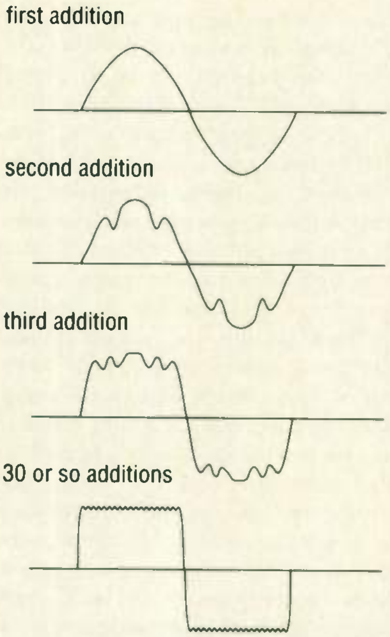

FIG. 3: Fourier reconstruction of a square wave via the addition of sine waves.

create a musically useful additive synthesizer. This is why the world is not exactly swimming in these types of instruments; they're very hard to make.

Recent advances in high-speed computers, parallel processing, and digital signal processors have made personal additive synthesizers financially realistic. However, don't be fooled by imitators! A real additive synthesizer must have a unique amplitude and frequency envelope for each harmonic, and the more harmonics, the better. Currently available instruments are on the pricey side, but keep your eyes peeled for new developments in this extraordinarily powerful method of synthesis.

PHYSICAL MODELING AND KARPLUS-STRONG

If additive synthesis and resynthesis are so powerful and flexible, how can things get better? Well, despite the advantages that resynthesis brings to additive synthesis, the two can still be confusing and a bit overwhelming. Physicists might love it, but perhaps a more intuitive method would be more usable by musicians. Enter *physical modeling.*

"Physical modeling" is a loose term used to describe synthesis by mathematically replicating the acoustic physics of real-world musical instruments via complex formulas. In physical modeling, it is necessary to come up with a mathematical description of each sound-generating and sound-altering component of an existing musical instrument.

In 1960 a mathematical model for vibrating strings based on physical resonating systems was proposed. This theory was extended and implemented by Alex Strong in 1978, resulting in realistic and rich timbres. Since then, Strong and Kevin Karplus have made improvements to their models so that extremely realistic synthesis of string instruments are possible. Because these models are based on real-world situations, they are far more musically intuitive than straight Fourier synthesis/resynthesis. The mathematics and theory that occurs behind the scenes, however, is not exactly intuitive or simple.

Basically, the Karplus-Strong algorithm consists of a digital filter representing the string and a short noise burst representing the string "pluck." In a vibrating string, high-frequency harmonics quickly diminish in intensity, leaving lower-frequency harmonics. This process is analogous to lowpass filtering, so the string is modeled by a filter (see **Fig. 4**).

When a string is plucked, it vibrates in aperiodic motion (which is why the pluck is represented as a noise burst) until the physical properties of the tensed string cause the string to fall into periodic motion. In order to implement the Karplus-Strong algorithm, a buffer capable of holding enough samples to constitute one period of a waveform must be set up. Initially, the buffer is loaded with numbers from a random-number generator to simulate a pluck. The onset of the note starts the buffer loading samples, one at a time, to the digital filter, with the digital filter returning the processed samples back to the beginning of the buffer.

This repetitious process results in a rapid transition from a random-vibration system to an ordered-vibration system oscillating at a frequency determined by the length of the buffer. The lowpass filter causes high-order harmonics to be filtered out as the samples make successive passes through the system. Altering the lowpass filter cutoff and slope is analogous to damping the strings of an instrument. Increasing the length of the buffer results in a decrease in pitch and decreasing the buffer length increases the pitch, just like a real string. Continuously changing the buffer length causes glissandi, slurs, and vibrato.

In a real string instrument, open strings and resonances of the instrument's body contribute to its overall character. The Karplus-Strong method is easily modified to handle this situation by setting up several copies of the algorithm, each one tuned to a different string length. By passing a small percentage of the output from a plucked string to an open string via a transfer coefficient, this interaction can be modeled (see **Fig. 5**). In this case, a separate Karplus-Strong algorithm models the vibration of the acoustic body, with corresponding filter parameters to simulate the characteristics of wood, plastic, or other materials.

Simulating moving-pick positions is easily accommodated by filtering the noise burst. By removing the even harmonics from the noise burst with a digital filter, the effect of plucking a string at its midpoint is simulated. Similarly, a pluck at the end point is simulated by attenuation of odd harmonics. Plucking

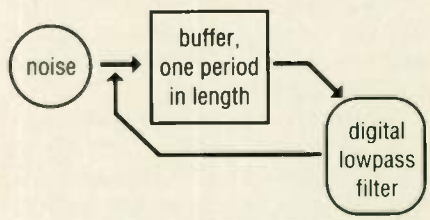

FIG. 4: The basic Karplus-Strong algorithm.

at all points on the string can be accommodated by differing degrees of harmonic suppression. Lowpass filtering of the entire noise burst characterizes plucking with a finger rather than a pick. Lower cutoff frequencies characterize meatier fingers.

Harder plucks on a stringed instrument result in greater excursions of the string from its rest position. This causes the onset pitch of hard-plucked strings to be higher than the sustain pitch. This higher-than-normal pitch is rapidly shifted down to the sustain pitch as the energy in the string is radiated into the air as sound. Hence, the harder you pluck, the greater the initial pitch-shift. This effect is also easily modeled by the Karplus-Strong algorithm by momentarily decreasing the buffer size for hard-plucked notes and then quickly returning the buffer to normal length. The speed of the buffer change is analogous to the density of surrounding air, with denser air absorbing energy from a vi-

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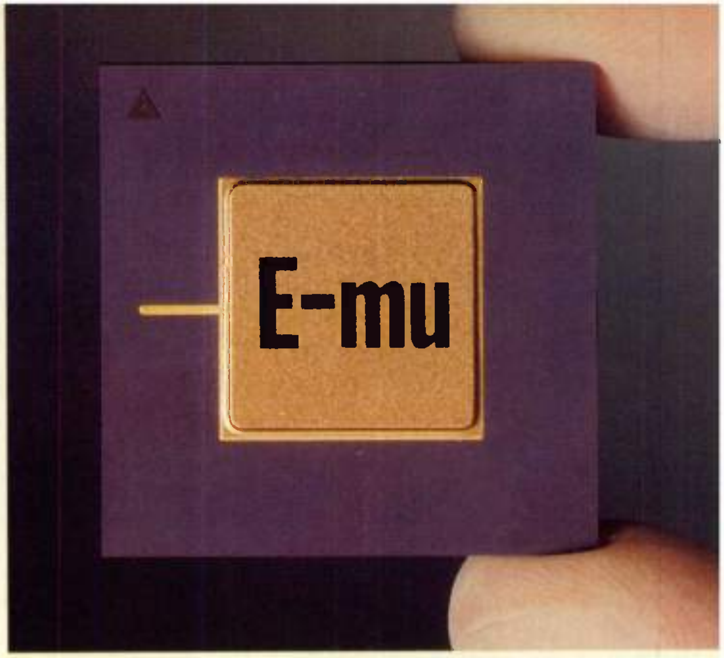

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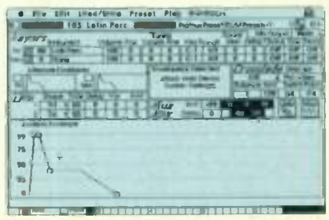

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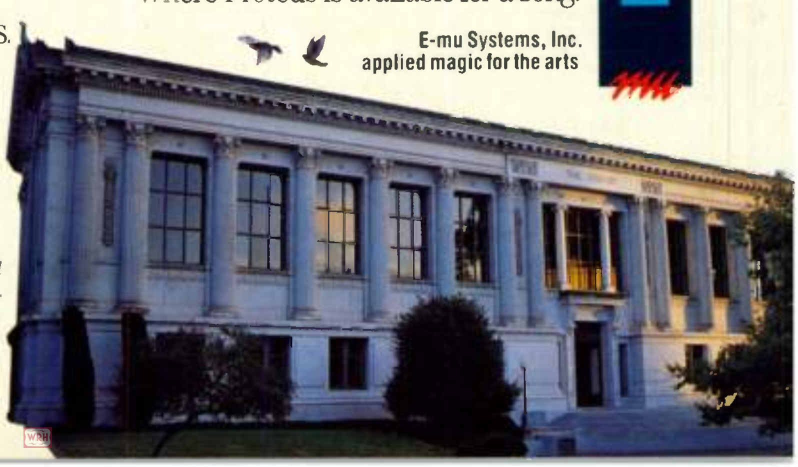

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brating string more rapidly and resulting in shorter pitch-change times.

The power and beauty of this implementation of physical modeling should be apparent. Not only is this method useful for simulating existing musical instruments, it's terrific for creating "virtual" instruments. For example, how about a guitar-like instrument with 400 strings? Make the strings range from twelve inches in length to twelve miles. Throw in pluck forces from 2 ounces to 40,000 pounds, too, what the heck? Because all these parameters are implemented by making changes to buffer lengths, filter parameters, and number of algorithms, it's easy to create "new" instruments that could never exist in the "real" world. Just think, you could be the first person on your block to play a "non-Euclidian" guitar.

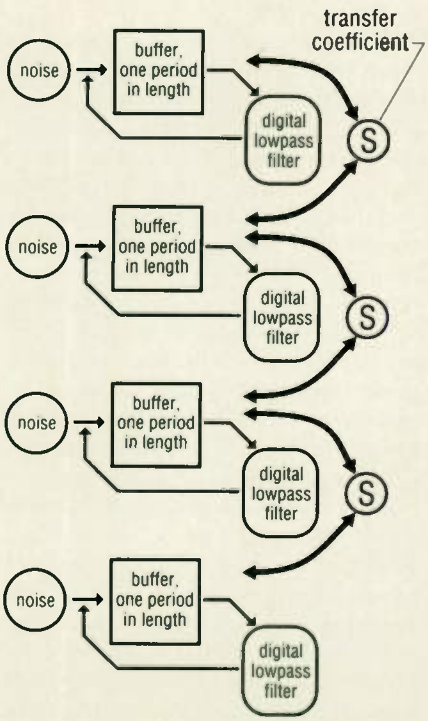


FIG. 5: Four-string Karplus-Strong algorithm showing transfer of energy to simulate open strings.

The intuitive part of sound modeling stems from the fact that you can simply plug in new variables to the equations to create new instruments. Consequently, depending on how a user interface was set up, programming could consist of something like calling up a clarinet algorithm, choosing the body material from a list of available choices (such as wood, steel, plastic, etc.), and determining the width of the bore and the length of the instrument. No doubt you'd have to deal with other types of numbers, but ideally you'd be able to create very complex, unique sounds very easily. Like all the new synthesis methods discussed here, sound modeling should actually encourage everyone to start creating (i.e., programming) their own sounds.

MORE MODELING

Physical modeling is not limited to the Karplus-Strong method. Another approach is to describe mathematically each physical interaction of a vibrating acoustic system. For example, the sound an organ pipe produces can be described as a result of the following physical systems:

- The aeolian (wind) tone produced by the air jet striking the lip cut-up of the organ pipe;
- The acoustic resonance characteristics of the waveguide created by the organ pipe;
- The acoustic resonances in the organ pipe wall itself;
- The impedance and damping characteristics of the air that fills the organ pipe;
- The interaction of all the above factors.

To illustrate the complexities of this method, consider the first physical system only, the aeolian tone produced when air strikes the lip cut of the pipe (see **Fig. 6**). The tone depends on the thickness of the air jet reaching the pipe cut, the width and length of the cut, the pressure of air reaching the cut (and hence the velocity of air), the physical shape of the lip cut-up, and the Reynold's number of the jet. (The Reynold's number describes when the air will make a transition from a smooth, or "laminar," airflow to a turbulent, or "nonlaminar," airflow. The nonlaminar airflow in the jet causes a periodic oscillation resulting in a tone.) All these properties must be described mathematically, in painstaking detail, to predict what properties the resulting tone will possess.

Physical modeling synthesis requires extremely accurate mathematical models in order to function, and the physics involved in creating such models can be staggeringly difficult. We are dealing with acoustic physics here, and a quantitative theoretical description of acoustic phenomena can be extremely difficult to come up with because acoustic musical instruments function partly in nonlaminar modes. Nonlaminar-fluid dynamics (remember that air is a fluid) is an extremely difficult branch of physics. Fluid dynamics theory is unable to handle precisely the nonlaminar airflow encountered in portions of musical instruments and unfortunately, these nonlaminar airflow situations generate important audible characteristics. The theory has to come first, and by then, hopefully, the computing power to implement it will exist.

Another problem is utilizing the incredible sonic control physical modeling offers. In the case of a flute model, you can alter the air pressure blowing into the mouthpiece, the size of the air jet, the angle the air strikes the mouthpiece, etc. Are we to manipulate all these parameters with a conventional MIDI controller? And if not, what kind of physical control mechanisms will exist to take advantage of tomorrow's synthesis techniques?

For these reasons, physical modeling has a formidable R&D barrier before it can become practical and justify the years and years of research involved. Nevertheless, I believe strongly that physical modeling will be the synthesis method of choice in the future—just not the near future.

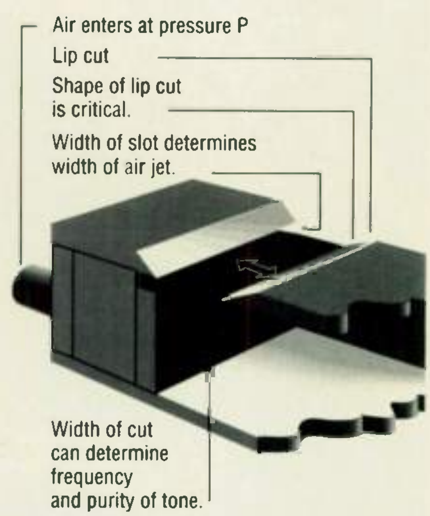


FIG. 6: The sound an organ pipe produces is controlled by factors such as incoming air pressure, width of the slot, shape of the cut, and the material from which the pipe is made.

● NEW SYNTHESIS

GRANULAR SYNTHESIS

Granular synthesis is the quantum approach to sound generation, using thousands of small sonic "grains" to form larger sounds. A form of additive synthesis, granular synthesis promises the same open-ended future that Fourier synthesis now provides.

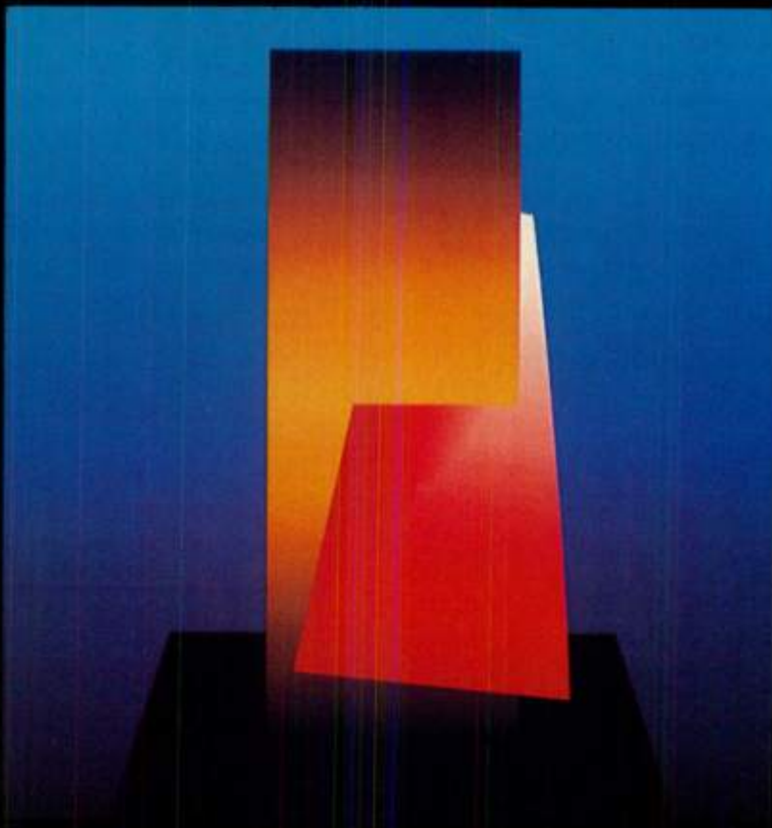
The "quantum" of granular synthesis is the sound grain. Each grain has a specific duration, waveform, and peak amplitude, with an amplitude envelope in the shape of a quasi-Gaussian bell curve, typically around 1 to 50 milliseconds in duration. Grains are combined, usually hundreds or thousands per second, to form "events." An event contains a start time, duration, initial waveform, waveform slope, initial center frequency, frequency slope, bandwidth, bandwidth slope, initial grain density (number of grains per second), grain density, slope, initial amplitude, and amplitude slope.

The typical method of granular synthesis is to slice a sound event into time slices. A slice, or "screen," contains the amplitude and frequency dimensions of hundreds of events. These screens are assembled into larger sound structures called "books." A book defines a complete sound object. Using this procedure, highly complex and fascinating sounds can be created.

Granular synthesis is still in its infancy. Perhaps the future will hold a place for this conceptually abstract, but powerful, synthesis method. Electronic musicians seeking extremely dense sounds will be drawn to it and I expect we'll see commercial granular synths in the next year or two.

In the end, many challenges remain before certain types of synthesis will be found in affordable instruments—as the complexity of the sounds being created increases, so do the efforts required to make these powerful new techniques of sound creation usable—but the future looks to be as sonically revolutionary as the past few years have been. It should be exciting.

Peter Gross *is a computer programmer/systems analyst at the Dept. of Biomedical Engineering, Queen's University (Canada), and is the president of CAMTEC, a digital audio consulting firm. His latest projects include designing a digital mixing console for broadcast applications, and creating a new product, the MIDI Baton.*



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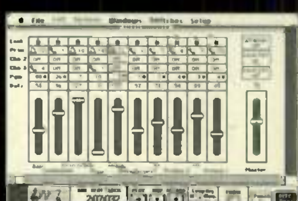
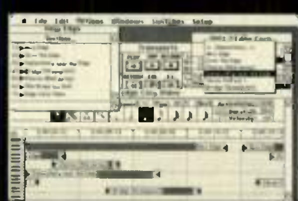
FOR THE APPLE MACINTOSH

We could have used a million cute phrases to describe our powerful new Macintosh sequencer, BEYOND. Instead, we decided to let the specs speak for themselves: complete graphic user interface; scrolling realtime piano-roll note editing; graphic controller and tempo editing; 4 programmable looping cue points with gapless recording; smart instrument setups; layer multiple MIDI channels to each instrument; support for 32 MIDI channels; 384 PPQ resolution; MIDI Time Code support; transpose, reverse, harmonize, and invert based on scales; "human" quantize; supports standard MIDI Files; controller shaping; markers; SMPTE display on edit screens; color; multi-track overview; graphic song display in block-like "Sections" with up to 99 tracks per Section; 32 programmable onscreen graphic sliders; not copy protected; and a manual by EM editor Craig Anderton. And if that's not enough, the list price of BEYOND is not beyond you — it's only $319!





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HARMONICS

THE BASICS OF SOUND

IT USED TO BE THAT NATURAL SOUND SOURCES (plucked string, struck wood, etc.) were the limits of our musical vocabulary. Today we can generate virtually any sound and, once generated, manipulate that sound with signal processors. The methods of generation are extremely diverse: sampling, subtractive synthesis, additive synthesis, FM synthesis, resynthesis, and so on. All of which seems to imply that we understand these facets as individual pieces; for example, we might think sampling exists in a different "box" from equalization, and what we know about analog synthesis doesn't relate to sampling.

Wrong. At least, not totally right.

Underlying all sounds are a few basic principles that create a unifying common denominator between all methods of sound generation and manipulation. Once you understand these principles, it's possible to see the whole picture: how equalization is just a variation on sound generation, how timbre relates to waveforms, etc.

Resynthesis, additive, sampling, subtractive, FM, equalization—they may seem different, but trace them back far enough and you'll find them rooted in harmonics.

The most fundamental point is that all sounds—from the simplest sine wave to the orchestral crescendo in Beethoven's *Ninth*—can be broken down into simpler sounds. Adding these simple sounds together creates complex sounds, just as stringing together a series of points creates a line.

These simple components are called "sine waves." A sine wave is the world's least complex waveform; it contains no harmonics, concentrating all its energy at a fundamental frequency of

LESLIE COBER

By Craig Anderton, Jim Conger, and Bob O'Donnell

oscillation. All complex sounds can be broken down into discrete sine waves, each of which oscillates at a particular frequency and has its own amplitude that varies over time. Breaking down complex sounds into simple sine waves is called "harmonic analysis."

STRINGING ALONG WITH THE HARMONIC SERIES

Before getting into sound analysis, we need to investigate the concept of a "natural harmonic series." All complex, natural sounds have numerous components, or harmonics (sometimes also called "partials," though there's no relation to the term as it's used in Roland's L/A synthesizers), and the harmonic series demonstrates the relationships between these parts. The easiest way to explain this concept is to think of a plucked string, like the kind you find on a guitar. **Fig. 1** shows a string tied down at both ends (i.e., fixed at the bridge and the nut). If you pull up on the center of the string and let go, you can imagine the string going up and down as a unit.

The sound this vibration creates is the "fundamental," or first partial, as shown in **Fig. 1a**. The fundamental is the basic frequency to which all harmonics are referenced. Let's assume the string goes up and down 220 times a second. This is a frequency of 220 Hz (Hertz means cycles per second), which corresponds to a musical pitch of A below middle C.

If you place your finger lightly on the center of the string and then pluck one end, the string will emit a tone an octave higher. This is because if you halve the length of a vibrating string, the speed at which it vibrates doubles. So instead of vibrating 220 times a second, the string now vibrates at 440 Hz, heard as the A above middle C. With this change, the string is no longer vibrating at its fundamental frequency, but at its first harmonic (**Fig. 1b**). By the way, be aware that the terminology of sound is almost as confusing as synthesizer terminology; the first harmonic is equivalent to the second partial.

The next harmonic (**Fig. 1c**) comes from placing your finger lightly at one-third the length of the string and then plucking. This produces a 660 Hz tone (the second E above middle C). This second harmonic, or third partial, is an interval of a perfect fifth above the first harmonic.

These multiples of the fundamental

create the harmonic series, in which each new harmonic adds one more wave to the sound of the vibrating string and yields a higher frequency tone. These waves are not heard as independent entities; it is their combination that produces the familiar sound of a vibrating guitar string.

Because the ends of the string are fixed, you can only get frequencies that are even multiples of the fundamental; you cannot obtain, for example, a 300 Hz wave from a string with a fundamental of 220 Hz without moving the end of the string. These simple multiples of the fundamental are called "harmonic partials"; nonharmonic partials, like the proposed 300 Hz wave, have a more complex relationship to the fundamental.

Here's the harmonic series for a string starting from A = 220 Hz:

	Fundamental	1st	2nd	3rd	4th	5th
Freq:	220 Hz	440 Hz	660 Hz	880 Hz	1,100 Hz	1,320 Hz
Note:	A1	A2	E3	A3	C4	E4

This same series of harmonics can be generated with any wind instrument by holding the fingering of a low note and then overblowing to successively higher notes. (The harmonic series also forms the basis behind our Western musical scale and common chord constructions, which demands an article to itself.)

A real-world vibrating string will have many harmonics present at the same time. The mix of harmonics, coupled with the string's interaction with the instrument's body, gives an instrument a distinctive timbre, or "color."

Then there's the matter of how the timbre changes over time. When you first pluck a string and jerk it into action, there will be a few milliseconds of chaos caused by the abrupt mechanical shock before the string settles down into a nice vibrational groove. This shock creates lots of harmonics, which is why a guitar sounds very bright when first plucked.

As the string decays, its mechanical inertia will tend to favor those harmonics with more energy, while the weaker harmonics decay sooner. This is easy to confirm if you look at a plucked guitar string on an oscilloscope or waveform analysis program: The initial waveform will be a complex harmonic structure, not a sine wave (**Fig. 2a**), but toward the end of the decay the waveform will become a sine wave because only the fundamental has enough energy to keep the string vibrating (**Fig. 2b**).

SOUND ANALYSIS

The basic idea behind sound analysis is to break a sound waveform down to its individual harmonics. The concept that any periodic wave can be expressed as a mix of individual harmonics was first demonstrated by the French mathematician Baron Jean Baptiste Joseph Fourier (1768-1830). In his honor, we now call this process "Fourier analysis."

A waveform that repeats a pattern is called "periodic." The period is the time required for the waveform to complete one full cycle. In music, the term "periodic wave" can be replaced with "pitched sound." The pitch of the sound is the frequency with which the wave repeats itself, such as 440 times a second for the A above middle C. Unpitched sounds, such as percussive sounds, contain waveshapes that change more radically over time. These waveforms are more complex than pitched sounds and, for the sake of reader and author sanity, will not be covered in this article. Do note, however, that most natural and complex electronic sounds have nonharmonic, nonrepeating elements in them.

Let's do a Fourier analysis of a square wave to determine which sine waves, at which amplitudes, need to be summed together to create a square wave. (Square waves are commonly found on analog synths. They're roughly equivalent to the sound from a stopped organ pipe.) **Fig. 3a** shows a square wave com-

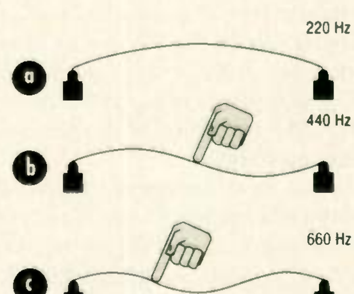


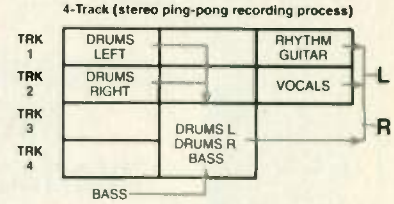
FIG. 1: (a) The fundamental frequency, or first partial. (b) The first harmonic, or second partial. (c) The second harmonic, or third partial.

Before you play ping-pong with a multi-track, read this...

The main weakness of four track recorders is their inability to adequately ping-pong tracks in stereo. If you want to record more than five instruments in stereo, you're going to need more than four tracks.

The Sansui WS-X1 has six tracks that can play back and record simultaneously. This means you can maintain a true stereo mix of up to 10 instruments in just two tape generations.

4-Track (stereo ping-pong recording process)



6 vs. 4 in the Real World

Here's a real-world example of six versus four tracks. The four track story is pretty short. Use tracks 1 and 2 to record a stereo drum mix, then ping-pong those over to 3 and 4 adding bass as you go. Now you can record rhythm guitar on 1 and a vocal on 2. Without taking a second bounce, the only thing left to do is the final mix.

6 Will Get You 10

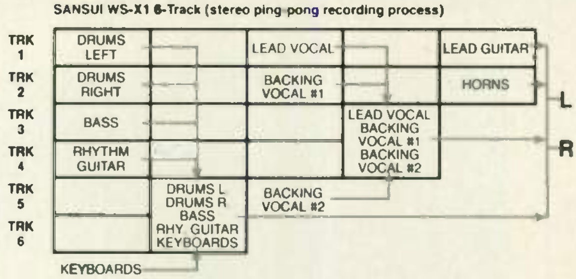
Now let's take a look at how the Sansui WS-X1 handles the situation.

The WS-X1's extra tracks let you record stereo drums, bass and rhythm guitar on tracks 1 thru 4 before you have to think about the first ping-pong. Selecting "Tape" on channels 1 thru 4 of the 8 channel mixer and choosing "Buss" on multi-track channels 5 and 6 are all you need to do to set up the ping-pong. You can add stereo keyboards to tracks 5 and 6 as you make the transfer. Just bring the left and right outputs of your keyboard into mixer channels 7 and 8 and record the keyboards as you make the mix.

Tracks 1 thru 4 are again available for recording. You can put a lead vocal on track one and a backing vocal on track 2. Using the above technique, you can transfer these tracks to 3 and 4 adding another backing vocal on the way.

Now you've got a stereo rhythm track mix on 5 and 6, and a stereo vocal mix on 3 and 4. You can now go ahead and record lead guitar and horns on tracks 1 and 2. You now have 10 distinct takes in easy to manage sub mixes.

SANSUI WS-X1 6-Track (stereo ping-pong recording process)



That's Not All, Folks

With its extra tracks, full featured mixer and on-board mixdown deck, the WS-X1 has everything you need to make professional sounding recordings in one easy to use package. See your authorized Sansui dealer today for a complete demo.

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For a full color brochure send $2.00 for postage and handling to:
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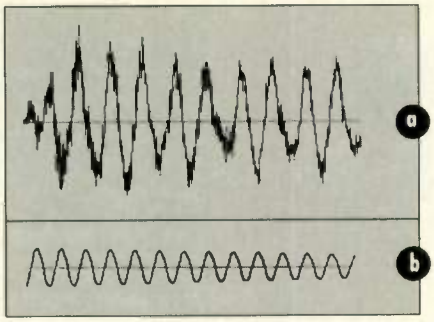

FIG. 2: (a) The harmonic structure is complex at the beginning of a string's attack. (b) The harmonic structure decays into a sine wave over time.

pared to a sine wave of the same period. It certainly seems this sine wave would be a part of the square wave, since its amplitude correlates closely to the square wave. The only difference is that the square wave has much sharper corners, so let's see which harmonics need to be added to sharpen those corners.

Let's try adding in a second harmonic (**Fig. 3b**). Hmmm...this seems to take us further away from a square wave, although it sure does look like we're starting to approximate a sawtooth. So we'll file this information (that a sawtooth seems to contain a second harmonic) for some other time and try adding a third harmonic instead.

Fig. 3c shows what happens when we add in some third harmonic. Bingo! Our waveform now looks much more like a square wave. If you keep adding odd harmonics (5, 7, 9, etc.) of the proper amplitudes (higher harmonics usually have lower amplitudes than lower harmonics), the shape will become more and more like a square wave. If, on the other hand, we add both even and odd harmonics, we'll end up with a sawtooth.

It so happens that the human brain processes the sound waves picked up by the ears in a manner that closely parallels Fourier analysis. This explains how we perceive pitch and is part of our almost incredible ability to pick a flute solo out of the mass of sound produced by a full symphony.

Because of this parallel, the Fourier sound spectrum is an excellent way to see what a sound looks like—much better than looking at the raw waveform. If you are trying to make a synthesizer sound like an acoustic instrument, work on duplicating the Fourier spectrum with your synth. To check your results, sample the synthesized sound into a sampler and view the results with a sample-editing program on a computer or the instrument itself. The raw waveforms may look different on a sampler's display (due to the harmonics being out of phase), but if the Fourier spectra match, the two sources will sound the same to our brains.

The lesson we can draw from all this is clear: Harmonics create a particular sonic signature. All our timbre-shaping and tone-generating tools rely on harmonic manipulation. As we look toward the future, we can expect that someday our instruments will integrate signal generation and signal processing seamlessly , making the transition from synthesizers/signal generators to harmonic manipulation devices.

Jim Conger *is the author of the books* C Programming for MIDI *and* MIDI Sequencing in C. *Someday he might learn how to actually play all the MIDI gear he has collected.* **Craig Anderton**, *is a musician/author who knows how to play all the MIDI gear he has collected but couldn't program in C if his life depended on it.* **Bob O'Donnell** *is currently collecting MIDI gear and programs that other people have written in C. Is that all clear now?*

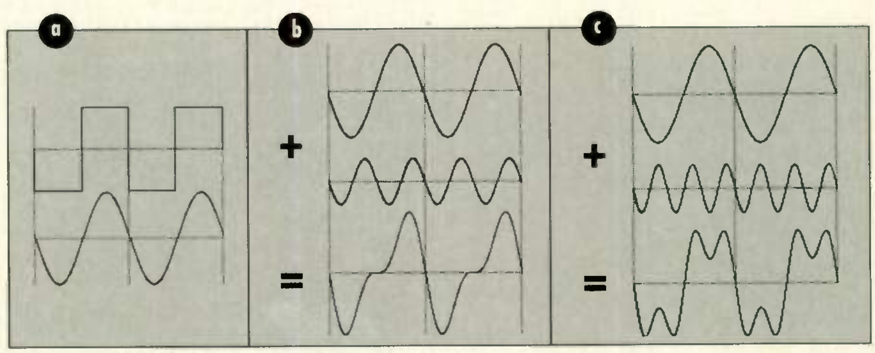

FIG. 3: (a) A comparison of a sine wave and square wave of the same period. (b) Adding in the second harmonic. (c) After adding in some of the third harmonic, the waveform starts to approximate a square wave.

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* MIDI LAN – MIDI Local Area Network

● MIXERS *from page 48*

ends, which may put budget boards to shame, but don't be too envious just yet. First, this type of EQ is expensive and raises the cost of the board substantially. With today's samplers and synths, where you can control a lot of the sound at the instrument, you may not need the flexibility a parametric offers but just a simple, sound-shaping EQ. Second, some parametrics sound better than others. I'd rather have simple equalizers that sound good than complex ones that sound bad. Finally, a parametric is not always the ideal equalizer for a particular instrument; for example, graphic equalization has much to recommend it in some applications.

The bottom line: If you're going to rely on the EQ in the console, spring for the best-sounding and most comprehensive types you can afford. If you're going to use a lot of electronic sounds that don't need as much EQ and are willing to patch in outboard EQ as needed, you can get a console with much simpler (and less expensive) EQ.

One nice touch is a bypass switch for the EQ so you can compare equalized and nonequalized settings. Having to reset the boost/cut controls to zero to compare settings is no fun at all. This also provides a way to evaluate EQ sound quality; set the EQ for flat response, then switch between the flat and bypassed settings. There should be no sonic difference.

Another nice touch is having the clipping indicator located post-EQ so that if you start adding large amounts of boost, you have some indication you're approaching clipping.

AUXILIARY SENDS

Auxiliary sends let you "pick off" a signal from the input module and route it elsewhere, such as to an effects unit or separate monitor mix. Both mono and stereo sends, or some combination of the two, are common; sometimes stereo sends are created by simply assigning two mono sends to left and right buses, while in other consoles there may be level controls with associated panpots.

Here is where space problems become a consideration—what if you want, say, sixteen auxiliary sends? One solu-

FOR THE BEGINNER

About Phantom Power

The condenser microphone (which is based on the construction design of a capacitor) consists of two metal plates, one fixed and one flexible, separated by an insulator. Capacitance changes will show up as voltage changes if the capacitor is *biased* with an electrical voltage; sound waves hitting the flexible plate cause capacitance changes.

Some condenser mics have a permanently charged plate, but others require an external power supply to provide the charge. Phantom power runs this voltage along the mic cable, thus dispensing with the need for an additional power supply at the mic itself. Also note that some active direct boxes can be powered from phantom power instead of their internal batteries.

Perfect Pitch method verified at Ohio State University!

They laughed at me and doubted me.... until I showed them the *secret* to Perfect Pitch!

A true story by David L. Burge

We were in ninth grade when I first heard that Linda had "Perfect Pitch."

Supposedly, she could name any pitch *by ear!* I was told she could even play any song after hearing it on the radio!

I doubted it. How could she know F# or Eb just by *hearing* it? An ear like that would open up unlimited possibilities for any musician.

It bothered me. Did she *really* have Perfect Pitch?

"Yes," she told me casually.

Perfect Pitch was too good to be true. I rudely asked, "Can I test you sometime?"

"OK," she said cheerfully.

Now I was going to make her eat her words...

I carefully picked a time when Linda had not been listening to music. Then I challenged her to name tones for me—by ear.

I made her stand so she could not see the piano keyboard. I made sure other classmates could not help her. Everything was set just right so I could expose this ridiculous joke.

Nervously, I plotted my testing strategy. Linda appeared serene. With silent apprehension I played a tone: F#. (She'll *never* guess F#!)

I barely touched the tone. *Instantly* she said, "F#"!

I was astonished.

I quickly played another tone. She didn't stop to think. *Immediately* she announced the correct pitch. I played more and more tones here and there on the keyboard, and each time she knew the answer—without effort. She was SO amazing—she could identify pitches as easily as colors!

"Sing an Eb," I demanded, determined to mess her up. Quickly she sang the proper pitch. I asked for more tones (trying hard to make them increasingly difficult), but she sang every one perfectly on pitch.

I was totally boggled. *"How in the world do you do it?"* I blurted.

"I don't know," she replied. And that was as much as I could get out of her!

The reality of Perfect Pitch hit me hard. My head was dizzy with disbelief, yet I now knew that Perfect Pitch was real.

I couldn't figure it out...

"How does she do it?" I kept asking myself. On the other hand, why can't *everyone* identify tones by ear?

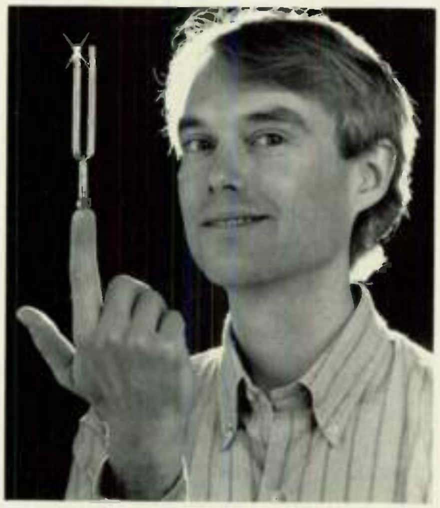
It dawned on me that most musicians go their entire lives without knowing C from C#, or G major from F major. That's like an artist who paints picture after picture without knowing green from orange. It seemed odd and contradictory.

I found myself even more mystified than before I had tested her.

Humiliated and puzzled, I went home to work on this problem. At age 14, this was a hard nut to crack.

You can be sure I tried it myself. I would sweet-talk my brothers and sisters into playing tones for me, then try to determine each pitch. Almost every attempt failed miserably.

I tried day after day to learn the tones. I tried to visualize the location of each pitch. I tried playing them over and over in order to memorize them. But



nothing worked. I simply could not recognize the tones by ear. It was hopeless.

After weeks in vain, I finally gave up. Linda's gift was extraordinary. But for me, it was out of reach.

Then came the realization...

It was like a miracle. And it happened all because I had stopped *trying* so hard. I had stopped *straining* my ear and started to listen NATURALLY. Then the incredible secret to Perfect Pitch jumped right into my lap.

I began to notice faint "colors" within the tones. Not *visual* colors—but colors of *pitch*. They had always been there. But this was the first time I had ever really "let go" enough to hear these *pitch colors* which reside in every tone.

Now I could name pitches by ear! It was simple. An F# sounded one way—a Bb had a distinctly different sound. It was as easy as naming red or blue.

The realization struck me: THIS IS PERFECT PITCH! This is how Bach, Beethoven and Mozart could mentally hear music on a page—and identify tones, chords, and keys at will—by *listening to these pitch colors*. It's that simple!

I became convinced that *any* musician could have Perfect Pitch by just knowing this secret of "color hearing."

When I first told my close friend Ann, she laughed. "Oh, I could never have Perfect Pitch," she asserted. "You can develop a good *Relative* Pitch [the ability to compare one tone with another], but you have to be *born* with Perfect Pitch."

"That's because you don't understand what Perfect Pitch *is*," I said. "It's easy!"

I showed her the secret and she heard it *immediately*. Soon she too could name any tone and sing any pitch requested. We became instant celebrities. Everyone was amazed.

As a keyboardist, Perfect Pitch allowed me to progress faster than I ever thought possible. I completely skipped over required college courses. Perfect Pitch made *everything* easier—performing, composing, arranging, transposing, improvising—and it skyrocketed my enjoyment as well. Music is definitely a *hearing* art.

Of course, music professors were highly skeptical when I started teaching Perfect Pitch years later. Most would laugh at the mere *suggestion* that anyone could have Perfect Pitch. But when I showed them how to hear the pitch colors *themselves*, they changed their tune!

Now there's *more* proof...

Research at Ohio State University has now independently verified my Perfect Pitch method (March '89). Their findings? *It works*, according to OSU researcher Dr. Mark Rush in an interview with *The Hartford Courant* (call our studio below for more info). I was pleased. They're just now finding out what thousands of musicians I've taught already know: that you really CAN have Perfect Pitch if you know *how to listen!*

YOU can have Perfect Pitch too, but you have to discover it. All you need are a few basic instructions. I've put everything I know into my **Perfect Pitch® SuperCourse,™** available on audio cassettes with handbook. The ***Color Hearing Technique*** I'll teach you is *totally guaranteed* to work for you, regardless of your style, instrument, or current ability level. It's easy—*you don't even have to read music!*

Like most musicians, you will *immediately* hear the beginning Perfect Pitch colors—*or you can return the Course for a full refund*. You've got my word on it.

Or you can check out your progress for 40 full days (use the handbook and first two tapes). If you don't experience a distinctly sharper, more musical ear by that time, just return the Course and I'll make sure you *still get your full refund—no questions asked!* I'm eager to prove that *you* can have Perfect Pitch, too!

If you'll try the Course *right now*, we'll also include my 90-minute companion cassette on **Relative Pitch**, which you can keep FREE *even if you return your Perfect Pitch course!*

Imagine your friends' disbelief when YOU can name tones and chords with laser-like precision! Don't laugh! At least not until you've heard the secret for *yourself!*

Hear the simple secret to Perfect Pitch! Order your tapes NOW!

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✔ I'll try it! Send me David L. Burge's **PERFECT PITCH® SUPERCOURSE™** with the **40-Day Money-Back Guarantee** that he describes above. Enclosed is my check or money order for $85 plus $4 shipping. (*NEW*: Course now includes handbook plus six audio cassettes, including *NEW* Perfect Pitch Solo Tape and 90-minute bonus tape on Relative Pitch!)

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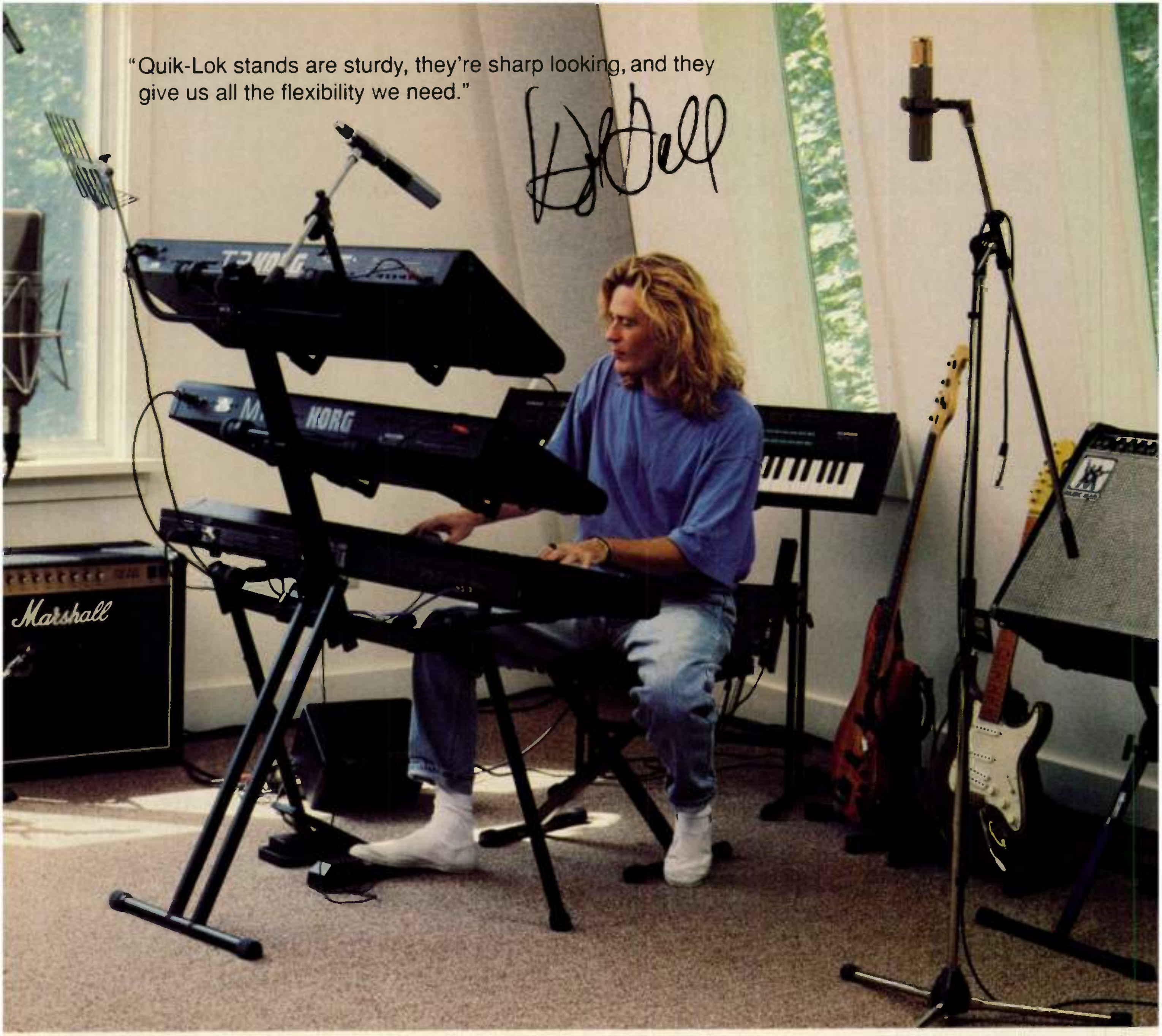

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tion is to have a lesser number of controls but be able to switch these to any of sixteen aux buses. Or, there may be an associated button so that, for example, four controls that normally feed buses 1 through 4 feed buses 9 through 12 when switched.

An important consideration is that the aux sends should be switchable between pre- and post-fader positions. Post-fader means that the outputs "track" any changes in fader level; as one example, the reverb send would get lower in volume along with the channel fader. However, there may be cases where you don't want the aux bus mix to vary as you move the fader. For these applications, select the pre-fader signal.

With some consoles, aux sends can turn into aux ins, great if a lot of signal processor outputs are returning to the console. An even more important point is that each aux bus should have its own master output control; otherwise you'll end up tweaking the send controls far too much for comfort. A mute control for each aux bus is also helpful.

MONITOR MIX CONTROLS

Although aux buses can be used for monitoring, you'll usually want a dedicated set of monitor controls. One popular monitoring method, *in-line* monitoring, uses a single control that can switch between monitoring a bus output or tape output. For example, the monitor control on input channel 1 would be able to listen to either bus out 1 or tape out 1. Most monitor sections will also include a panpot, and in some cases, you may find a switch to dump the monitor mix to the main stereo bus.

MUTE AND SOLO SWITCHES

A mute switch simply mutes a given channel to take it out of the mix. This is great for keeping a noisy signal (e.g., a hissy guitar amp) out of the mix until right before the downbeat where the instrument starts playing. A solo switch leaves the selected channel active but mutes all others. This is handy for adjusting EQ and other fine settings where hearing the other channels would get in the way. Almost all mixers let you solo multiple channels if desired.

OTHER SPECS 'N' STUFF

The input module, while important, is not the only part of a mixer worth examining. Here are a number of other features you should consider.

• MIXERS

Switching and routing: One basic routing feature involves the outputs. It's often useful to switch between two pairs of speakers, or between mono and stereo, to check for possible phase problems. Switching can get pretty esoteric, however, especially in smaller mixers where one bus may be required to do multiple tasks.

Meters: There are three main types of meters: mechanical movement (VU), mechanical movement with peak LED, and multistep LED VU meters. Mechanical meters show average signal levels; large, short-duration transients may not even budge the needle, which is why having an additional peak LED is useful. LED VU meters are not limited with respect to response time and can thus indicate peaks well, but sometimes (especially on stereo outputs) you'll also want to see average readings. Some pricier consoles provide LEDs on the inputs but VU meters on the outputs (which is great), but if you have to choose one or the other, I'd say go for the LED meters.

Note that most meters will be switchable to monitor a variety of sources, such as aux bus outputs, post-fader outputs (useful when mixing), and so on.

Channel-activity LEDs are also great. These are located at the input modules and simply show that audio is being received at a particular input. This is a good reality check for when you don't hear something you think you should be hearing, as you at least know the signal is making it to the board.

Long-throw faders: The longer the fader travel (throw), the easier it is to create subtle mixing "moves." In pro boards, 100mm faders are standard.

Built-in test oscillator: Sure, why not. Not essential—you can use a commercial equivalent or build your own—but handy.

Modular vs. fixed construction: With modular mixers, you can remove input modules and replace them with other types of modules. For example, you might want to replace an input module with an input/output module that sacrifices some input module features to gain more functions overall, or replace an input module with an auxiliary send module if you happen to have a lot of aux buses.

Frame size: With modular mixers, when you order a large console, there will usually be a choice of frame sizes. The larger the frame size, the more modules can be accommodated. Since modules are of varying width, the frame size will usually be expressed as holding a certain number of input modules rather than as a specific linear measurement. One advantage of modular mixers is that you can order a large frame initially, and fill it up with more modules as your studio expands.

Dedicated stereo effects returns: So there you are with a bunch of neat-o stereo signal processors. The only trouble is, every time you plug one of them into the board, you use up two more inputs. Dedicated stereo effects returns are inputs but lack some of the features of the main inputs. As far as I'm concerned, the more of these, the better. Also advisable: a trim control (preamp) for those -20 dB operating level guitar effects you've been known to use from time to time when you thought no one was looking.

> *Once you use automation, you'll never go back to doing mixes manually.*

Control-room section: This includes controls for setting listening levels and routing so that you can, for example, switch between the multitrack and your master tape without repatching hassles, or switch between a couple of different decks (cassette dub, DAT dub, etc.). Some consoles also have a separate level control to feed studio speakers.

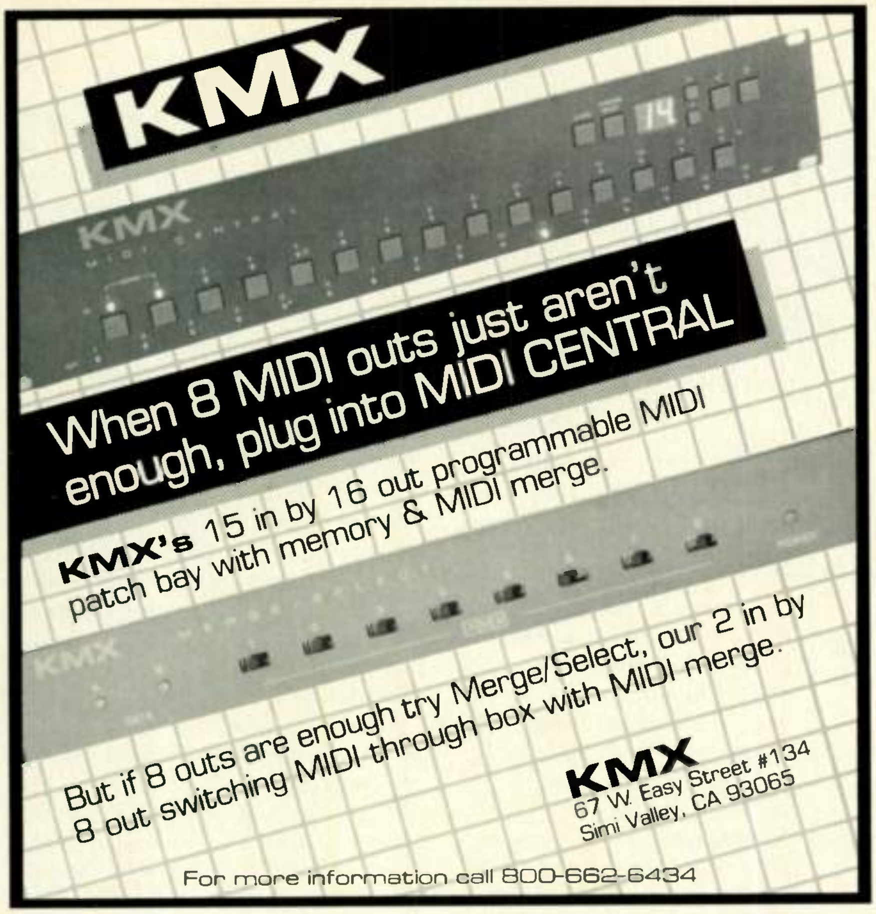
Internal patch bay: Yes, it means a huge price increase, but at least you won't have to sit there and wire a patch bay yourself.

Talkback module: This consists of a microphone and switching system for letting those in the control room talk to those in the main studio. The switching lets you put the talkback signal into, say, the headphone mix for communicating with vocalists, or on the stereo bus for recording take numbers at the beginning of each mix.

AUTOMATION

Simply stated, once you use automation, you'll never go back to doing mixes manually. While automation generally means "fader automation," that is not the only way to go. For example, several years ago, Allen & Heath brought out a mixer in which mutes and routing could be synchronized to SMPTE or MIDI, thus providing many benefits of automation but without the expense of moving faders. They were probably ahead of their time, but these days, fader automation is the name of the game.

More expensive consoles have complete automation facilities built in, but you can also retrofit existing consoles with products such as J.L. Cooper's Magi, the Twister automation package, or the console retrofits by Jellinghaus. The *ne plus ultra*, though, is moving fad-



ers. These not only remember your moves, but play them back during mixdown. Editing is simple; just grab a fader and move it to the new position. With non-moving faders, it's usually necessary to move the fader to a "null" point (i.e., the recorded position), then punch in the automation move.

Yamaha started another automation trend with its DMP7, which can place almost all mixer parameters under MIDI control. This ties up a computer to provide sequencing functions but allows a great degree of control at a reasonable price. Akai's MPX 820 took a different approach, storing mixes as "snapshots" that could be recalled under MIDI program change commands. The one wrinkle that added to its value was that you could program a variable crossfade time between snapshots, thus eliminating the unmusical aspects of an instant change.

No matter what automation system you use, it's better than no automation at all and is well worth the added price.

ON THE HORIZON

Right now mixers are in flux. Some talk of "virtual mixers," where mixers will be programmed like a synthesizer—call up a channel and use a slider to adjust EQ, level, and so on. This sounds tedious but workable, especially if you have one fully functional input module that can switch to different channels. Some day, the huge consoles loaded with zillions of knobs may become as obsolete as modular synthesizers with zillions of knobs, but don't count on it just yet—ergonomics are crucial in the studio, and studios are willing to pay the price for increased productivity. It's more likely that virtual mixers will gravitate toward budget studios, and those who need to add an additional submixer to their main mixer.

I've also heard rumors of a mixer from Allen & Heath that replaces its previous automated model and provides some aspects of MIDI control at the mixer itself. There's no reason why, for example, you couldn't run your sequencer from a console, or use program changes that control the rest of the system, rather than just have the mixer respond to the outside world. If you throw in automation and lots of stereo inputs, this would be a very strong contender for MIDI-oriented studios.

Speaking of which, I think that some companies will also come up with rela-

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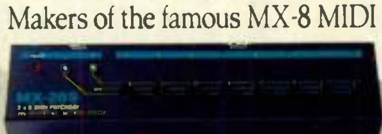

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5312-J Derry Avenue • Agoura Hills, CA 91301 • TEL. (818) 991-3881 • FAX (818) 991-4185 tively low-cost mixers designed for studios that do a lot of MIDI work. These would lower costs by cutting the number of mic preamps, scaling back the sophistication of equalization, and instead concentrate on providing lots of inputs and aux sends. In some respects, line mixers such as the ones by Roland come close to meeting this goal; it's just a matter of time before consoles in this format become more common.

We're also seeing a transitional period, much like the days of the Sequential Circuits Prophet-5 and Oberheim OB-8 (digitally controlled analog synthesizers), in which analog consoles are controlled with digital electronics. Aside from providing opportunities for automation, these types of consoles can offer other features such as onscreen displays of EQ curves and sends. They also form a logical bridge between the all-analog consoles of the past and the all-digital consoles that will most likely be part of the future.

The big change, though, will come when mixers with digital inputs that bypass the analog domain altogether start to take over (Yamaha has such products out now). As we see more and more digital-output sound sources, it will seem silly to convert them to analog for the sake of a mixer, then digitize them again to go on a master tape.

ARE YOU ALL MIXED UP YET?

Confusing stuff, eh? Today's mixers have so many options and special features that it is not at all easy to choose one. What you really need to do is study the block diagrams thoughtfully provided by manufacturers with their brochures, and spend as much time *listening* as possible. As stated in the beginning, specs don't really tell you much since there is no standardized test for measuring consoles; the more you listen, the more you'll be able to trust your ears and make an informed opinion. And of course, your budget will immediately put some limits on your desires anyway.

Talk to other recordists, analyze your own set of needs, ask manufacturers what their consoles offer that others don't, and above all, take your time. A mixer represents a sizable investment—give yourself a chance to really study the matter before you jump in. Good luck!

(Thanks to George Petersen, *Mix* magazine's official mixer master and multi-linguist, for comments and assistance.) ■

Build a Tube Preamp

If you've been longing for that much sought-after "tube sound," and a solid-state simulation just isn't good enough, here's the ticket.

By Tom Dahlin

Photograph of author's prototype.

Given all the hullabaloo regarding the "unique, warm" sound of tubes, I decided recently to build a tube preamp for my home studio. Please note, however, that *this is not a circuit for the novice electronics builder*. The circuit itself is simple, but unlike most semiconductor projects, this one uses voltages near 300V. The danger of electrocution is present and neither the author nor this magazine can assume responsibility for protecting you. If you are not familiar with good construction and safety practices for high-voltage electronics, do not attempt to build this circuit.

SO HOW DOES IT SOUND?

This preamp won't make the sound of an overdriven Marshall stack, nor is it a heavy metal distortion unit, but it does produce tonal colors ranging from a slight "warming" to soft distortion. Turning down the drive control produces a somewhat brighter sound with no distortion; turning the drive all the way up produces distortion that really bites on the bass strings and softens out on the higher notes. Driving the unit hard also produces a bit of a compression effect. With my two-finger power-chord playing style, I find the overall effect quite pleasing.

CIRCUIT DESCRIPTION

The preamp circuit (**Fig.1**), based on a design in the 1975 *RCA Receiving Tube Manual,* uses a 12AX7 tube as a two-stage, voltage gain amplifier. This "generic" circuit is similar to what you will find in most guitar amp preamps.

The 12AX7 (the 7025 and ECC83 are equivalent tubes) is probably the most common tube used in the music business. Almost all current tube effects use this tube, as do most vintage tube amps. The 12AX7 was designed for use in audio applications requiring exceptionally low hum and noise levels. It is a high gain, twin triode device, having two identical sections sharing a common heater filament.

The guitar or other input feeds J1 and goes to the grid of the first tube stage, which produces a voltage gain of about 30. This boosts the guitar's 40 mV (nominal) input up to about 1.2V. Coupling capacitor C2 picks off the amplified signal while blocking the high voltage plate supply. The signal then goes through the tone control circuit (R5-R9 and C3-C6) before feeding the second stage's input. Potentiometer R10 is a voltage divider that sets the drive.

The second tube stage provides a gain of ten. The output signal couples through blocking capacitor C8 to the load resistance (R13 and R14). Control R14 varies the output signal level at J2, up to a maximum of about 1.5V. This should feed a relatively high impedance stage; 600-ohm mixer inputs are not recommended. Resistors R4 and R12 are *bias resistors* that allow the tube cathodes to develop a positive potential with respect to ground. Capacitors C1 and C7 filter any ripple appearing across these resistors.

Fig. 2 shows the power supply circuit. Transformer T1 is a dual secondary type rated at 250 VAC (Volts Alternating Current) center-tapped for the tube's plate supply and 6.3 VAC for the tube filament. Diodes D1 and D2 form a full-wave rectifier, which has a DC output filtered by C10, R15, and C9. The output from a full-wave rectifier is 0.7 times the input AC voltage, so about 185V appears across C10 and under load, about 140V appears across C9. These capacitors carry high voltages and retain their charge for some time after the unit is turned off, which could be a problem

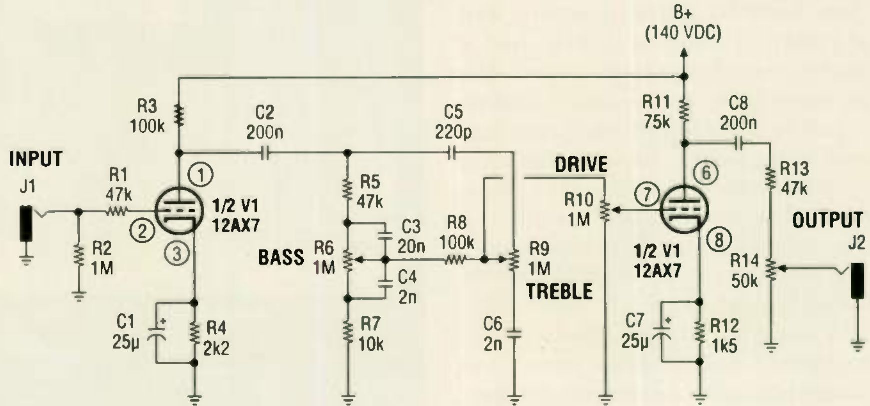
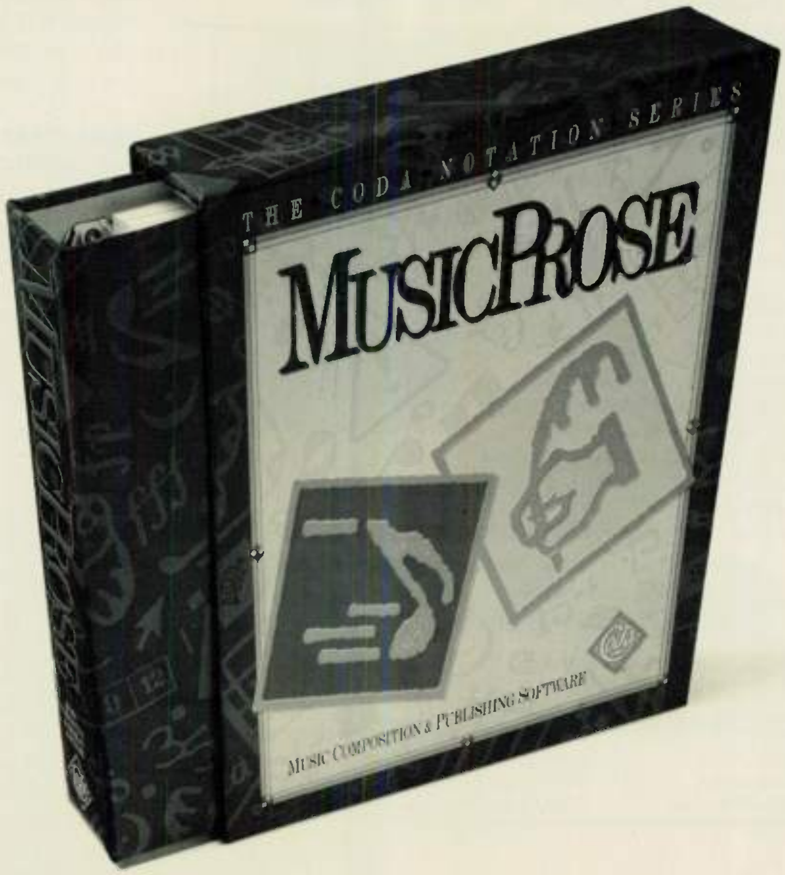


FIG. 1: Tube preamp schematic, based on a 12AX7 dual triode.



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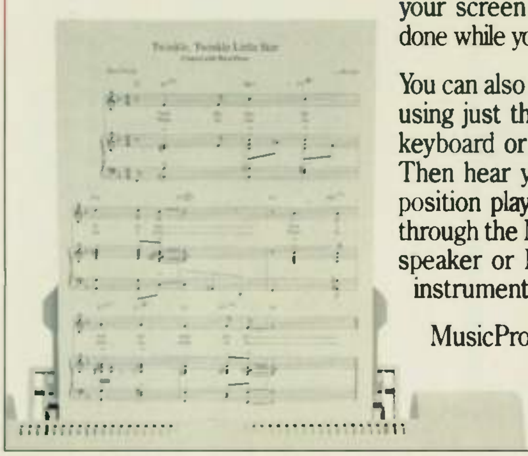

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when servicing the unit. So, "bleeder" resistor R16 drains the filter capacitors over a period of about a minute when power is removed.

THINGS THAT ARE FUSSY (AND THINGS THAT AREN'T)

Vacuum tube circuits, including this one, are amazingly forgiving of component tolerances and substitutions. In my experiments, I've used components of half to double the specified values with success. While you shouldn't exactly randomize all the suggested values, by all means use a 200 nF capacitor in place of a 100 nF if it's all you have.

FOR THE BEGINNER

The Tube Bibliography

For more on tube circuits, check out:

RCA Receiving Tube Manual (RC-30), 1975 edition, RCA Special Products Division, Camden, NJ 08101. This has been out of print for several years, but was once the definitive handbook for tube design. You can still find these at ham radio operator conventions for about a buck each. An absolute necessity.

Tube Amp Book #2, by R. Aspen Pittman, 1988, Groove Tube Publishing, Sylmar, CA 91342 ($14.95). This is an update of *The Tube Amp Handbook*, and includes schematics for just about every guitar amp known to Western civilization. Also has tips on souping up your amp. Highly recommended.

Audio Anthology, Old Colony Books, PO Box 243, Peterborough, NH 03458; tel. (603) 924-6371. A collection of 38 articles from *Audio Engineering Magazine* during the heyday of tube circuits (1947 to 1950). No guitar amp stuff in here, but lots of history and good reading.

Glass Audio Newsletter, Circulation Department, PO Box 176, Peterborough, NH 03458; tel. (603) 924-9467. $10 for one year (two issues). A neat little publication that just appeared on the scene. Its audience is primarily anachronistic audiophiles who for whatever reason prefer the sound of tubes to solid-state devices. They are definitely not into distortion, overdrive, etc., but do have some interesting ideas.

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However, the capacitor voltage ratings and resistor wattage ratings are extremely critical. You can always use a greater value for either, but *never-repeat, never-use.*

There is nothing magic about T1's output voltage value. Any transformer that produces 150 to 275 VAC should work fine. If you can't find a center-tapped type, splurge and use a couple of extra rectifiers to make a full wave bridge. You want to end up with a plate voltage value in the area of 100 to 350V. I haven't found a great deal of difference in the sound as a result of varying the plate voltage; in fact, some of the popular tube pedal effects use less than 30V across the plate and cathode.

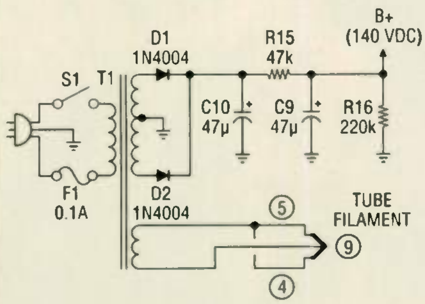

FIG. 2: Power supply schematic.

You can also use a transformer with a 12 VAC filament winding by grounding ground pin 9 of the 12AX7 and running the 12 VAC into pins 4 and 5.

CONSTRUCTION DETAILS AND HINTS

I've built one large and one small version of this circuit. The larger unit was much easier to construct and had less hum due to the fact I had more room to route and dress the internal wiring. The smaller unit shown in the opening photo looks cooler, but was a real pain to build. I also didn't have enough room to mount the drive control on the front panel; I instead mounted the pot internally, although this wasn't a big problem since I usually have it cranked wide open anyway. Moral of the story: don't try to cram too much into a small box.

Use a metal chassis, as it will reduce hum and is more structurally sound than plastic. (I used a Bud CU-2108 for the larger unit and a Bud CU-2106 for the smaller one.) If you use a different transformer from the one I did, make sure it will fit in the chassis. The trans-

PARTS LIST

RESISTORS (all resistors in ohms; ½ watt, 5% carbon types except as noted)

R1, R5, R13	47k
R2	1M
R3, R8	100k
R4	2k2 (2.2k)
R7	10k
R11	75k
R12	1k5 (1.5k)
R15	47k, 1W
R16	220k
R6, R9, R10	1M audio taper potentiometer
R14	50k audio taper potentiometer

CAPACITORS (rated at 350 VDC or greater except as noted)

C1, C7	25µ, 50V electrolytic
C2, C8	200n (0.2µ) ceramic disk or metallized polyester
C3	20n (0.02µ) ceramic disk or metallized polyester
C4, C6	2n (0.002µ) ceramic disk or metallized polyester
C5	220p ceramic disk or metallized polyester
C9, C10	47µ electrolytic

OTHER PARTS

D1, D2	1N4004 rectifier (1000 PIV, 1A)
F1	0.1A fuse
J1, J2	¼" open-circuit phone jack
S1	SPST switch
T1	transformer, 250 VAC CT at 25 mA, 6.3 VAC at 1A (Allied #705-0001 or equivalent)
V1	12AX7, 7025, or ECC83 vacuum tube
Misc.	9-pin tube socket, chassis, fuse holder, knobs, line cord, grommets, terminal strips, wire, etc.

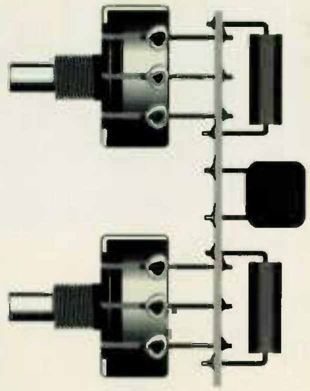

FIG. 3: Tone control construction detail. Having tone control components directly above the pots keeps lead length short, minimizing hum.

former I used barely fits on top of the small box.

The order in which you build this circuit helps to keep it quiet and improves access to test points. Start by wiring the power supply circuit—everything in Fig. 2 except the filament wiring. I mounted S1 on the rear of the unit to keep the AC power from running through the chassis to the front panel. I also kept all the power supply components to the rear of the chassis, leaving the front half for the tube circuit.

Next, begin building around the tube socket. Wire in the tube filament first. Use a twisted pair for this and keep it close to the chassis, away from where you will be routing other components. Then wire in the plate power resistors, R3 and R11. Use a scrap of insulation or "spaghetti" over the leads to prevent shorts. Now add the cathode resistors and capacitors. By following this order, you will be able to get in to tweak and probe without digging components out from under a "pile-o-wires."

A small piece of perf board, mounted over the tone control pots, holds the tone control circuitry (resistors R5, R7, and R8, as well as capacitors C3, C4, C5, and C6; see **Fig. 3**). I used stiff (#16) tinned wire to interconnect the pots to this daughter board, effectively supporting the board. If you don't like the idea of floating the board by the stiff wires, use a couple of stand-offs. Three leads connect to this assembly: one to C2, one to the drive pot R10, and one to ground.

Keep the leads going to the tube grids short and away from other leads, especially the filament leads carrying AC. I used shielded wire for the connection from R10 to the second stage grid as this connection seems to be very susceptible to noise. Finally, make sure you use a grounded line cord both for safety and for hum reduction.

CHECKING IT OUT

Take a little time now to perform a few simple checks before you actually plug in the unit; you'll save time overall.

The following tests are conducted from the chassis top, at the tube socket (the tube is removed). This makes it very easy to access test points and avoid shorting something out. I generally clip the test leads to a couple of short wires (1-watt resistor leads work great) and poke into the tube socket with these.

Ohmmeter tests: Perform these with the power *off*. To be on the safe side, put the wall end of the AC cord in front of you before you start and short out filter capacitor C9 with an alligator clip jumper (in that order).

1. Measure the resistance from pin 3

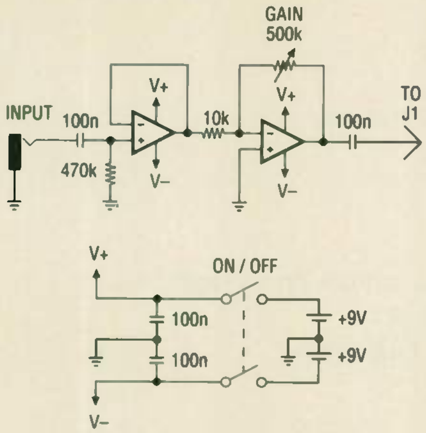

FIG. 4: Simple op amp preamp suitable for overdriving the tube preamp.

to ground. This should approximately equal cathode resistor R4 (2.2 kΩ).

2. Measure the resistance from pin 8 to ground. This should be approximately the resistance of cathode resistor, R12 (1.5 kΩ).

3. Clip one ohmmeter lead to C9's positive lead and measure the resistance to pin 1. This should be about the same as the first-stage plate load resistor, R3 (100 kΩ).

4. Leaving one side of the ohmmeter on C9's positive lead, measure the resistance to pin 6. This should be about the same as R11, the second-stage plate resistor (75 kΩ).

Signal injection tests: In these tests you inject a signal into a point in the circuit and look (or listen) for it in another. *Power should still be off at this point.* Ideally, use a signal generator to inject the signal and an oscilloscope to monitor it. Lacking these, you can use any audio source for an injection signal and a cheapo amp and/or headphone for a monitor. The idea is to verify that the signal is making it through to where you expect it to be. The signal level you use is not critical; one volt or so will work fine.

1. Inject a signal into input jack J1 and look for it on pin 2 of the tube socket (the first grid).

2. Turn the drive control pot R10 up halfway. Inject a signal into pin 1 and look for it on pin 7. Play around with the tone and drive controls and note the effects on the signal at pin 7. You should be able to adjust level and tone to a certain extent.

3. Inject a signal into pin 6 and look for it at output jack J2. You should be able to vary the output signal level by adjusting R14.

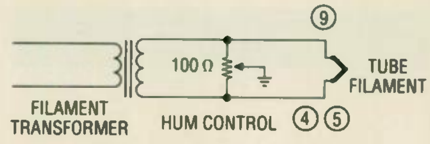

FIG. 5: Hum adjustment circuit.

Voltage tests: Now let's test the power wiring. Remember we will have almost 300V running around, so *watch out!* If you have an alligator clip lead across C9, now is the time to remove it. The tube should still not be in its socket. Connect your voltmeter's negative lead to the chassis ground.

1. Turn on the power and check for plate voltage at pin 1. It should be around 140 to 150V.

2. Check for second stage plate voltage at pin 6. It should also be about 140 to 150V.

3. Check for 6.3 VAC between pins 4 and 9 and between 5 and 9. (Remember pins 4 and 5 connect together.)

That's it for the tests; if the circuit has passed them all, the odds of success are in your favor. Power down, insert the tube into the socket, plug in the AC cord, and turn on the power switch. If all is well, you are now the proud owner of a tube preamp.

SO MUCH FOR LEAVING WELL ENOUGH ALONE...

Here are a couple of mods you might want to try. If you're not adverse to mixing silicon and tubes, **Fig.4** shows a simple op amp preamp that lets you hit the tube stages with a bit more "oomph," providing a more overdriven, distorted sound.

If you encounter hum, try adding the hum control potentiometer in **Fig. 5**. Note that the 100-ohm pot must be a 5-watt, wirewound type.

I hope you enjoy building and playing with this project; here's your big chance to add some tubes to your life and get that great tubular sound before they become extinct.

Acknowledgment: Thanks for assistance from the good folks at Knut-Koupee Music.

Tom Dahlin *is a computer scientist employed by a very large company in Minnesota with a very short name. Tom designs both hardware and software for image processing systems. The work in this article has led him to his current interest, vacuum tube computers; Tom theorizes that tube computers will exhibit softer overflow errors.*

FINDING PARTS

Parts for this project are available by mail order or possibly from a local TV repair shop. Allied Electronics carries everything you need to build this project; while not as cheap as outfits like Mouser Electronics, Allied has everything necessary. Like Mouser, it takes credit cards and is reasonably fast. Mouser is generally my preferred source for these parts, but its catalog doesn't list a suitable transformer. If you can find another source for the transformer, I'd recommend Mouser for everything else.

The 12AX7 tube can be purchased from several mail order sources. Groove Tube and Mesa Boogie offer selected devices that are hand-picked for guitar applications. Chances are your local music store will have some also.

What I don't recommend is salvaging parts from that old radio or TV that's been sitting in your attic for the last ten years. Most likely the electrolytic (and other) capacitors are shot, the pots are noisy, and any component you manage to unsolder or cut out will have leads a half inch shorter than what you need. Even the transformer could have cracked, old insulation on the leads. If you can pull a 12AX7 out of the unit, it may work as a spare, but use a new tube when you first check out the preamp.

Addresses:

Allied Electronics Incorporated, 401 E. 8th St., Fort Worth, TX 76102; tel. (800) 433-5700. $5 handling charge for orders less than $25.

Mouser Electronics, 2401 Highway 287 North, Mansfield, TX 76063; tel. (800) 34-MOUSER. $5 handling charge on orders less than $20.

Groove Tubes Inc., 12866 Foothill Blvd., Sylmar, CA 91342; tel. (818) 361-4500. Great source for all your tube needs. Get the *Tube Amp Book #2* while you're at it.

Line Mixer Duo

If you're all mixed up about not having enough inputs in your studio, don't replace your board—get these simple, inexpensive submixers into the act.

By Alan Gary Campbell with Cliff Sadler

Completed passive line mixer (stereo version).

No matter how many mixers, submixers, preamps, and Y-cables you kluge together, it seems there are always more instrument and effects outputs than mixer inputs (and MIDI virtual tracks just make things worse). These simple line mixer circuits can help. They work great, cost little, and are easy to build—you can finish 'em before your day job catches up with you.

The duo consists of a passive, 8-input mixer that features lowest possible noise and minimum parts count, along with an active 10-input mixer that can be expanded to provide up to 32 additional inputs.

PASSIVE LINE MIXER

The passive circuit (Fig. 1) mixes multiple inputs quietly and inexpensively. Since there are no active devices, there's no power supply, noise, or distortion.

Of course, you don't get something for nothing. The input levels interact somewhat, and for best performance, the total number of inputs should be eight or less. Also, the circuit doesn't include input coupling capacitors or level controls; the output circuits of the electronic instruments and sound processors feeding the mixer must provide these functions.

The circuit is so simple that point-to-point wiring is best. Keep the leads short and use a metal enclosure, preferably steel. For lowest possible noise, use metal-film resistors for R1 to R8.

For stereo mixing, duplicate the circuitry. For semipro audio and patch bay applications use RCA-jack inputs, premounted in strips, to further simplify wiring.

ACTIVE LINE MIXER

While a tiny bit noisier and more expensive than the passive version, the active circuit (Fig. 2) is expandable and can accommodate input coupling caps, level controls, and stereo operation with panpots. It uses just one IC, a dual operational amplifier ("op amp" for short; see "For the Beginner" sidebar). IC1A forms an inverting, summing block—a very common op amp application. The op amp inverting input looks just like a ground connection to the inputs, preventing any interaction among the inputs. The input impedance is that of each input resistor (100k). Dividing the feedback resistor value by the input resistor value gives the voltage gain per input (in this case, 10k/100k = 0.1).

IC1B forms a phase inverter so the outputs are not 180 degrees out of phase with the inputs. It also allows for variable gain, via potentiometer R14, of 0.01 to 20. In conjunction with IC1A, this provides an overall gain of 0.001 to 2.

Perf-board construction is fine for this circuit. As with the passive version, a metal enclosure, preferably steel, will minimize electromagnetic interference and hum pickup, and will allow the use of unshielded wire to connect the input and output jacks to the board. Still, keep the leads as short as possible since they act as "antennae" for interference. Connect the jack tip terminals only; connect the chassis to the board ground to serve as a common ground return for all the jack ground points.

If you use a plastic enclosure, run a length of 2-conductor, *shielded* cable from each jack to the board. The two inner conductors carry the audio signal; one connects to the jack tip terminal, the other to the jack sleeve terminal. The shield connects to ground *at the board end only*. All shield connections should terminate at a common point (also called a bus). If you still pick up spurious signals, add a 100 picofarad cap (C3) between IC1A's inverting and noninverting inputs, as indicated.

For IC1, use a dual, low-noise, internally compensated op amp such as the XR4739, TL062, TL072, LF353, NE5532, etc. Pinouts are not shown on the schematic since this may differ for different op amps, so check the data sheet supplied with the op amp to determine the power supply, input, and output pins. Use metal-film resistors and a cermet pot for the lowest possible noise; use carbon-film resistors and a carbon or conductive-plastic pot in less critical applications. (Incidentally, because the frequency response of the circuit in Fig. 3 goes down to DC, it makes an excellent control-voltage mixer for use with analog synthesizers. In this application, a low-noise, dual op amp isn't required, so

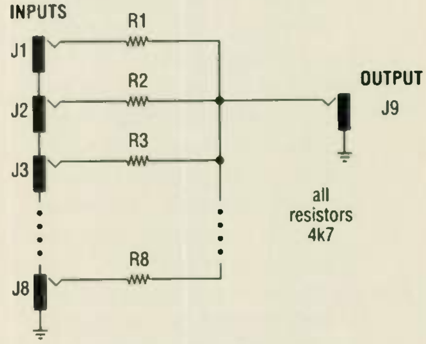

FIG. 1: Passive line mixer schematic.

a 1458 will do; carbon-film resistors and a carbon pot are fine.)

Two 9-volt batteries, connected as a bipolar supply, can power the mixer, but if you have a regulated bipolar supply, use it. Note the two 1N4148 silicon diodes; they provide protection in case the batteries are accidentally reversed. If you make permanent connections to a regulated power supply, omit the diodes, since they degrade the headroom by about 1.4 volts.

You can add as many inputs as you want, although each input adds a very

FOR THE BEGINNER

Understanding Op Amps

The operational amplifier, or op amp, is the most common type of analog integrated circuit. Op amps were once composed of many discrete, "unintegrated" components. These early types were used in large, analog computers as "math blocks," i.e., circuits easily configured via external components to perform the analog functions of addition, subtraction, multiplication, division, integration, differentiation, etc.—sort of like the electronic equivalent of a slide rule (remember those?).

As op amps increased in popularity and decreased in size and cost, they became ubiquitous, indispensable analog-design components, and rapidly found their way into electronic music circuits: mixers, lowpass filters, and highpass filters are the audio equivalents of analog-computer circuits for addition, integration, and differentiation. Despite the modern predominance of digital circuitry, most electronic music devices still incorporate one or more op amps.

For further information, read Radio Shack's *Getting Started in Electronics* and *Engineer's Mini-Notebook: Op Amp IC Circuits*, both by Forrest M. Mims. For technical information, see the *Op Amp Cookbook*, by Walter Jung, from Howard W. Sams & Co., or data books such as Radio Shack's *Archer Semiconductor Reference Guide* and National Semiconductor's *Linear Databook*.

—*AGC*

slight bit of noise. Thirty-two inputs is about the practical maximum. If you prefer more or less inputs, change the values of resistors R11 and R12 according to the following formulas:

R11 = 100k ÷ N
R12 = 100k ÷ 2N

Where N is the number of inputs. Determine the values for R13 and R14 as follows:

R13 = R12 ÷ 100
R14 = R12 ÷ 20

Use the closest standard resistor and pot values. You can modify the formulas to accommodate input-resistor values larger than 100k, if you like, to take advantage of parts you may have on hand or to handle high-output impedance devices such as electric guitars. Just substitute the new value in the first two equations. But don't go above 150k; higher impedances aggravate electromagnetic interference and add noise. (Also, make sure that all the input resistors are the same value!)

To add input-level controls and coupling capacitors (the latter are necessary if your instruments generate any DC offset, although this is unlikely), modify each input as indicated in **Fig. 3** (input 1 is shown as an example). Audio-taper pots are preferable; the coupling caps, preferably poly types, can be any value from 220 to 470 nF.

To build a stereo line mixer, duplicate the circuit of Fig. 3 (use a quad op amp, e.g., the XR4136 or TL074), and add the panpot circuit of **Fig. 4** to each input. Note that the dual pot's two sections are wired in reverse of one another so that as the output of one section increases, the output of the other decreases. A dual pot with one audio-taper section and one reverse audio-taper section is ideal for the best "feel," but these are very difficult to find. A dual, linear-taper pot will do, but *don't* use a dual audio-taper pot, as it won't work in this application.

To incorporate input-level controls, coupling caps, *and* panpots, place the level controls first, panpots next, and coupling caps last, before the input resistors.

There! That wasn't so difficult. By now, you've probably whipped out the old soldering iron and made your obligatory visit to the local parts retailer. You should consider it a musical sin to not build at least one of these mixers; whether used as stand-alone devices or submixers for a larger board, these simple projects will help out during those many situations where your existing mixer just doesn't have quite enough inputs.

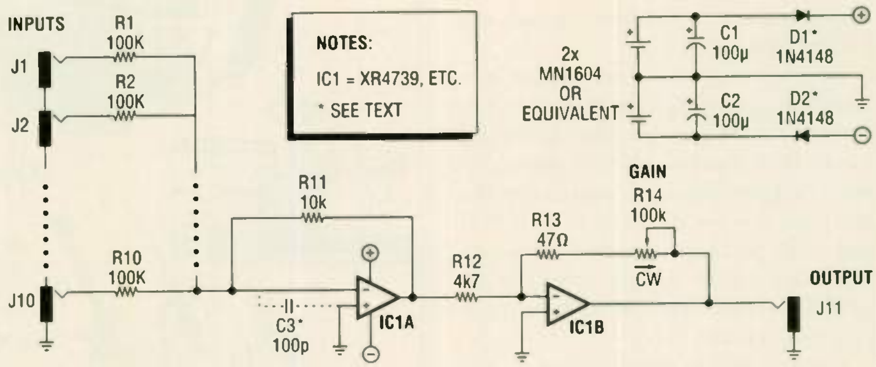

FIG. 2: Active line mixer schematic.

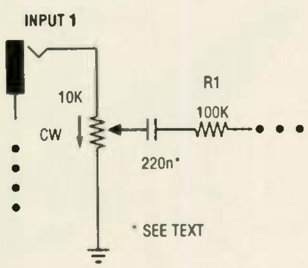

FIG. 3: Adding input attenuators and coupling caps.

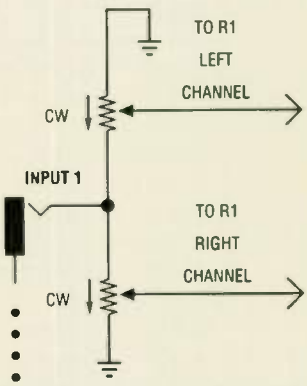

FIG. 4: Adding input panpots for stereo operation.

PARTS LISTS

All resistors ¼-watt, 5%
All capacitors 16 working volts DC, or greater

PASSIVE VERSION

R1-R8	4k7
J1-J9	¼-inch, 2-conductor, open-circuit phone jack (see text)
Misc.	Enclosure, hookup wire

ACTIVE VERSION

R1-R10	100k
R11	10k
R12	4k7
R13	47 ohm
R14	100k pot
C1, C2	100μ electrolytic
C3	100p ceramic (see text)
D1, D2	1N4148, 1N4001, or equivalent
IC1	XR4739 dual, low-noise op amp (see text)
J1-J11	¼-inch, 2-conductor, open-circuit phone jack
Misc.	Enclosure, perf board, hookup wire, battery clips and holders

Alan Gary Campbell *is author of "Service Clinic" and contributing editor to* **EM**. *He has constructed electronic music devices since age 12, and constructed electronic music devices that actually work since age 16.* **Cliff Sadler**, *unfortunately, has a real job and cannot devote full attention to his MIDI/music passion. He does, however, possess a highly respected talent for talking his family out of buying such extravagances as food and clothing, so he can further clog his MIDI data-stream for the good of all.*

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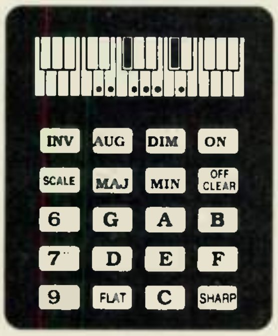

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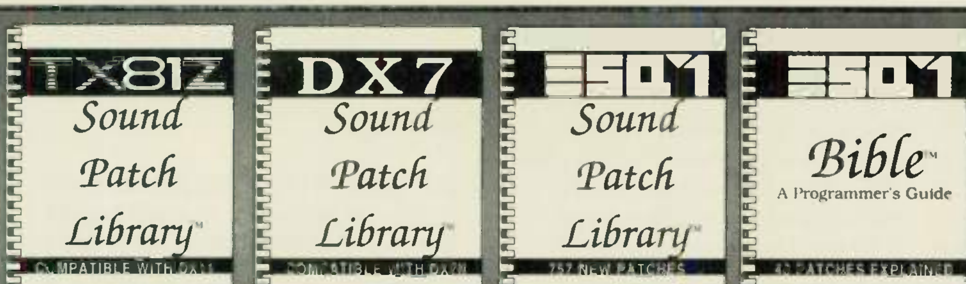

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SERVICE CLINIC

Questions and Answers

Piezo pickups, fuse fundamentals, crosstalk on cables, ear-protection enlightenment, and an ailing ARP Omni II string synth's symptoms are covered this month.

By Alan Gary Campbell

Q. Should MIDI and audio signals be carried by the same cable or snake? Is there any danger of crosstalk or weird interference effects?

A. In general, the use of an individually shielded, twisted pair for MIDI transmission is adequate to prevent radiative interference with audio signals carried by nearby pairs, in multiconductor cables or snakes. The current-loop nature of MIDI precludes most interference effects. Nonetheless, for best performance the MIDI pair or cable should be a low-capacitance-ype with a continuous-coverage, foil shield.

However, in MIDI guitar-controllers and other devices modified to send MIDI and audio through the MIDI cable, considerable capacitive crosstalk can occur at the traces near the output jack. While the MIDI bit rate is ultrasonic and generally doesn't affect audio devices downline, the MIDI *word* rate is clearly audible and in extreme cases can pollute the output with metallic, insect-like buzzes. Often, extensive PC board reworking is required to correct this, though sometimes you can increase the controller audio output level sufficiently to "bury" the crosstalk in the system noise floor (*not* recommended in ultra-quiet digital recording setups).

Q. Can I use elements from piezoelectric buzzers to replace defective piezo pickups in electronic drum pads *à la* Simmons SDS and others?

A. Yes. My favorite pickup is the big, two-lead element from Radio Shack's 273-073 Piezo Transducer. Just pry the element out of the plastic case (*gently;* don't nick it), stick it in place with some silicon sealer and solder the leads to the output jack. Be careful when you strip the insulation from the leads; they're small and fragile.

These elements make great audio pickups too, as described in James Chandler's "Two Buck Pickup" article (March 1987 issue). I taped one to a solid-body kalimba and ran it through my Alesis Midiverb II reverb unit. Awesome.

Q. I want to replace two blown fuses in my Ensoniq Mirage sampler. Both appear to be in series with transformer secondary outputs, but carry 250V (volt) ratings. Is this correct? For that matter, should 250V fuses be used with 125V lines?

A. It's acceptable to use a fuse with a voltage rating *higher* than the level encountered in the circuit, but *not* vice versa. Thus, a 250V fuse can be used in a 250V, 125V, or even 24V circuit. However, a 125V fuse should *not* be used in a 250V circuit; this is an important point, since some devices incorporate switchable 250/125V power supplies. Use only 250V fuses with such equipment.

Always replace fuses with the proper type and rating. If you're not sure, contact the equipment manufacturer or a reputable service center to find out. Never replace a fast-blow fuse with a slow-blow-type; don't replace a fuse with one of a higher rating, or with a fuse case wrapped in foil. Any of these misapplications pose a serious fire hazard. Fuses mounted in spring-clip holders should be removed with a nonconductive fuse puller (GC 5525, Radio Shack 270-1199, or equivalent) designed for the fuse size and type in question. Once removed, fuses can be tested for continuity with a VOM (volt/ohmmeter) if visual inspection proves to be inconclusive.

Q. According to a safety-product supplier in my area, 3M has stopped making the model 6300 Foam Ear Plugs that you recommend for hearing protection. Is this correct? Do you know if these are still available? Is it possible to clean and reuse the model 6300s?

Q. Why are the 3M hearing protectors, and others, available only in bright colors? Orange projections sticking out of one's ears look rather silly on stage! Does any company make similar hearing protectors in flesh tones?

A. 3M has stopped making foam hearing protectors subsequent to recent patent-infringement litigation. The deletion of the model 6300 Foam Ear Plugs is unfortunate, as they provide superior performance and comfort. Fortunately, Mix Bookshelf still has a considerable stock of 6300s available ($6 for 10 pair, $10 for 20 pair; see the FYI page for info). Sound reinforcement companies and

LINDA BLECK

others who purchase hearing protectors in quantity might be able to find a local distributor with some remaining stock. Check the Yellow Pages under "Safety Equipment & Clothing."

The model 6300s are *disposable* hearing protectors, intended to be discarded after one period of use. Nonetheless, it might be possible to rinse off and reuse the ear plugs (soaps, cleaners, and solvents should, in all likelihood, not be used on them), though the user is solely responsible for any degradation of the ear plug material and attending reduction of performance that might result—including slime mold growing out of your ears!

Hearing protectors are manufactured in bright colors to aid safety supervisors in verification of compliance with OSHA (Occupational Safety & Health Administration) regulations that require the wearing of hearing protectors in noisy industrial environments. There are many manufacturers of reusable, non-foam (read: uncomfortable) protectors in "flesh" tone (i.e., tan), and apparently most manufacturers of these devices assume that all races have the same color ears! I know of only one company that makes foam protectors in tan; contact EAR Division of Cabot Corporation, 5457 West 79th Street, Indianapolis, IN 46268-0940; tel. (317) 872-6666 for a distributor in your area. Other visually unobtrusive foam hearing protectors are the lime green ones made by Moldex (no, I'm *not* making this up). Extraterrestrials should be able to wear these undetected.

I'll continue to evaluate new hearing protection products and report any significant results.

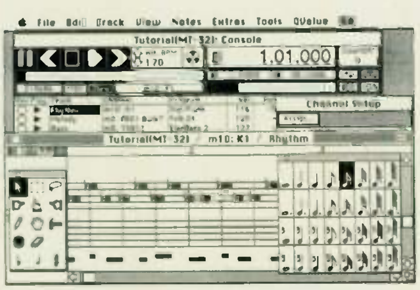
Q. I have an ARP Omni II string synth that I can't bear to part with, but it's showing signs of age. High F plays when any key is pressed, then fades away at the full release value. The D-flat below it behaves similarly when the Release slider is turned up past the 30% point. What causes this?

A. The Omni II, based on top-octave-divider technology, uses two (one per rank), simple, bipolar transistors per key as gain-control devices. If, for a given key, the diode or the timing capacitor in the keying circuit (refer to an Omni II service manual) becomes leaky, the transistors will remain on and the note will bleed through or drone. Replacing the cap or diode will fix it. For best results, use 22µF, 25V (or 35V), low-leakage tantalum caps and 1N4148 diodes as replacements. Be sure to observe proper polarity for the cap or you'll see tantalum fireworks! Low-leakage, Panasonic EF-series caps are available from Digi-Key Corporation, PO Box 677, Thief River Falls, MN 56701; tel. (800) 344-4539. Write or call for catalog and ordering info.

Less frequently, a leaky transistor array will cause bleed-through, but only on one rank (i.e., 8-ft. or 4-ft.) not both. The TDA0470 transistor arrays, other parts, and Omni service manuals are available from Music Dealer Service, 4700 West Fullerton, Chicago, IL 60639; tel. (312) 282-8171. Write or call for ordering info. ■

First Takes & Quick Picks

Inexpensive hardware and software sequencers, Amiga synthesis software, and more are on tap in this month's collection of condensed, action-packed reviews.



A main screen and graphic editing window from Deluxe Music Recorder.

Electronic Arts Deluxe Recorder 1.0 for the Macintosh ($149.95)

By Wheat Williams

D*eluxe Recorder* is a noteworthy by-product of the recent, heated Mac sequencer wars. It's an inexpensive, "junior" version of Resonate's powerful *Portrait,* a program intended to compete with Opcode's popular *Vision* program. Alas, Resonate is out of business, and Portrait has yet to see commercial release.

Electronic Arts is publishing Deluxe Recorder as a home-hobbyist program. However, it merits professional attention. While it lacks some features, it incorporates some innovations in programming and user-configurability. Its powerful and unique graphic editing features blow away some products costing three times its price.

the EM rating system

All EM reviews include 11-step "LED meters" showing a product's performance in specific categories chosen by the reviewer (such as ease of use, construction, etc.) and a "VU meter" indicating an overall rating. The latter is *not* a mathematical average, since some categories are more important than others. For example, if a guitar synth has great documentation and is easy to use, but tracks poorly, it could have several high LED meters and a low overall rating.

The rating system is based on the following values, where "0" means a feature is nonfunctional or doesn't exist, while a value of "11" surpasses the point of mere excellence (a rating of 10) and is indicative of a feature or product that is truly groundbreaking and has never before been executed so well.

Please remember that these are opinions, and, as always, EM welcomes opposing viewpoints. We urge you to contact manufacturers for more information, and, of course, tell them you saw it in EM.

Deluxe Recorder is actually a series of interdependent and user-customizable software "modules" that run off of a central set of source code called the "Kernel." All modules can be up and running at the same time. The sequencer records at an impressive 480 pulses per quarter note and supports 32 MIDI channels.

The program records incoming MIDI data into a buffer that is independent of its sixteen tracks. The data can then be used to replace any given track, or can be merged into that track's existing data. The buffer can record on multiple MIDI channels simultaneously (as with a 6-channel guitar controller) and can automatically split each channel's data into a separate track, if requested. The buffer also enables you to "loop" a specified number of measures in record mode, holding in memory whatever you played during the last full pass. Deluxe Recorder is also capable of recording rubato, without the metronome. The performer can turn on the sequencer, record music that changes tempo at will, and then go back and specify where the measure lines are to be created. Deluxe Recorder then creates a sequenced track with the correct tempo changes calculated automatically.

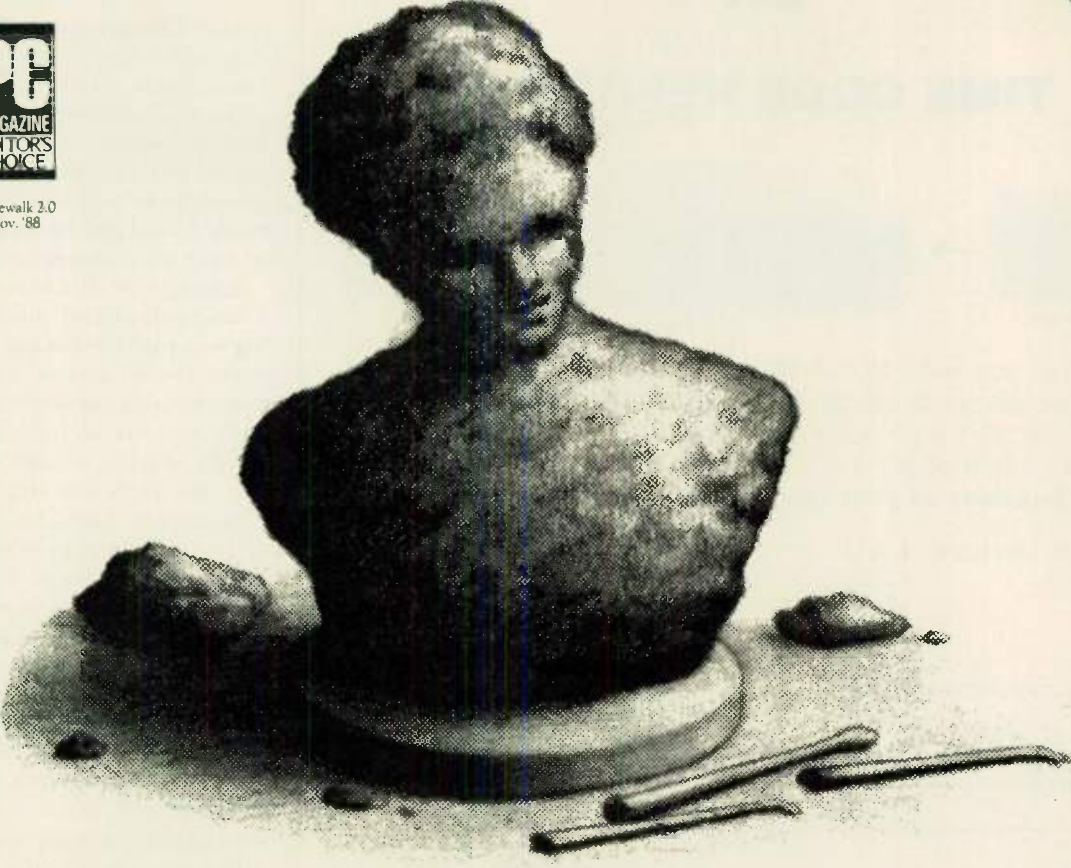
The Console window, which lists track assignments, also acts as an audio mixer. It sports pop-up sliders, for each track, that can change the MIDI volume, left/right panning, and transposition of any MIDI instrument responding to those commands, in real time.

Deluxe Recorder's editing capabilities are most impressive. A track's notes, controller data, and tempo and time sig-

NEW

PC MAGAZINE EDITOR'S CHOICE

Cakewalk 3.0
Nov. '88



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SYSTEMS

nature changes are displayed graphically on any combination of eleven user-customizable "staves," which can be seen by scrolling through a single window. You can configure how much data is displayed at a time. Note data can be displayed on a "piano roll" or on a sheet music "grand staff," where the durations of notes are shown as horizontal bars.

Editing note data in real or step time is extremely precise, thanks to two floating tool palettes that can be positioned anywhere on screen. You can select notes by using any combination of three tools: the Arrow, for selecting whole ranges of notes in time; the Marquee, for drawing boxes around groups of notes; and the Lasso, for picking out discontiguous notes by drawing irregular shapes around them. Thus, arbitrary groups of notes can be modified while leaving neighboring notes untouched. One drawback is that notes are deselected as soon as one operation is performed on them: To quantize a group of notes and then transpose them, you must select the notes twice.

In addition, there are specific tools for changing a single note's duration, attack time, pitch, velocity, and MIDI channel, and a second palette for selecting step-time notes of any articulation and rhythmic value up to 128th-note tuplets. And, thankfully, Deluxe Recorder has an extensive "Undo" command for most operations.

All of this is enhanced by "power keys," single keyboard commands that duplicate most functions selectable by the mouse. However, Deluxe Recorder becomes redundant and cluttered here; information is displayed in numerous pop-up menus and sliders that are duplicated or even triplicated by dialog boxes or other nearby menus.

Another innovative idea comes into play with the Channel Setup, Name Editor, and Name Tables modules. Deluxe Recorder not only refers to synth patches and channel assignments by channel and patch number, it also includes a large online, pop-up database that displays the actual names of each factory preset on a host of popular synth modules (the M1, DX7, TX81Z, and MT-32, to name a few). You can create your own tables with custom names for your own synths and patches.

Deluxe Recorder reads and writes standard MIDI files and is designed to exchange files with *Deluxe Music Construction Set* (DMCS) V. 2.5, Electronic

Art's entry-level, PostScript, music notation package ($129). In theory, this provides a way to get a sequencer file transcribed into a good-quality band chart, lead sheet, or the like. In reality, it's very disappointing. Deluxe Recorder can hang up when trying to convert complex MIDI files. Converting Deluxe Recorder files to DMCS format can be very slow and requires a great deal of patience as the user must make crucial trial-and-error decisions about how the program will quantize data. When converting files of any complexity at all, Deluxe Recorder simply crashes.

From the professional standpoint, Deluxe Recorder has its shortcomings. It can't chain sequences for live performance, and there is no provision for recording or sending system exclusive data. It has no direct support for SMPTE (it displays time in terms of measures, beats, and clock ticks rather than minutes, seconds, and frames). However, external MIDI synchronization is supported.

Deluxe Recorder comes with a clearly written manual that, through numerous tutorials, explains sequencing from the standpoint of the novice. The professional will find it lacking in its coverage of MIDI, however.

For now, one must overlook the program's failure to provide a practical bridge to DMCS. It packs surprising versatility and sophistication into an inexpensive and accessible package. As a home-market product, Deluxe Recorder is a winner.

Wheat Williams *is studying vocal performance in college in Nashville, Tennessee. He plays classical music in churches and sneaks off to play with computers when no one is looking.*

Features 8
Ease of Use 8
Documentation 9
Consumer Value 10

Electronic Arts
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San Mateo, CA 94404
tel. (800) 245-4525
or (415) 572-2787

Brother MDI-30 Sequencer ($299)

By Daniel Alan Phillips

When equipment is being designed for the low-end market, it's generally not possible to include every feature that can be dreamed of. The designers of such gear must instead ask themselves what is truly basic to the functioning of the device, and what is a frill that can be done without. Disk drives are normally one of the first things to go, as producers opt for significantly less-expensive tape interfaces. Brother has bucked this trend with their disk-based hardware sequencer, the MDI-30, released at the breakthrough price of $299.

Brother is aiming the MDI-30 at the beginner market; the manual even includes Hal Leonard E-Z Play scores of the songs on the demo disk. The company obviously hopes that beginners will appreciate the simplicity of what is essentially a tape machine emulation, with the additional MIDI advantage of being able to record a part slowly and then

Brother MDI-30 Sequencer

speed it up on playback.

The MDI-30 offers two linear tracks with a very respectable 96 pulses per quarter note, although there's no pattern/song programming for flexible song arrangements. The tracks can be merged repeatedly, allowing the creation of thick, multichannel compositions, but there is no way to unmerge data. Once you've merged a part, it's there to stay. Pitch bend and aftertouch may be filtered during (but not after) recording, a much-needed memory-saving feature. System exclusive data may be stored and recorded as separate files, allowing disk storage of synthesizer voice data.

When it comes to editing, the MDI-30 is just about as bare-bones as a sequencer can get. There's just a 2-digit LED display and absolutely no editing features beyond a simple, real-time spot delete. You won't find any cut-and-paste, transposition, or quantization capabilities. If you just want to record something and play it back without worrying about anything else, the MDI-30 is fine, but if you want editing features, you may want to check out something more sophisticated, like the Alesis MMT-8.

The 3½-inch disk drive on the Brother MDI-30 has a capacity of 240K or 30 files, whichever is least, and loads the full memory in about thirteen seconds. That memory, 32K of RAM, is reportedly capable of storing 7,000 notes, although I couldn't ever get it far above 4,000. The MDI-30's big brother (ahem), the MDI-40 ($379), features a 6-character display and 64K of RAM, but I only had the '30 for this review.

As one might expect from a box with a 2-digit display and a mere six buttons, the user interface is somewhat less than utopian. Even so, it would have been possible to design the command structure with more consistency. The Enter key, for instance, is sometimes used as a data increment key, as opposed to its normal "return" function, and the Cancel key often works in a way that is tantamount to *confirming* data settings. To make things worse, if you happen to have a question about the device, you'll have to call Directory Assistance; neither an address nor a phone number is listed in the manual.

In my mind, the people most likely to want a disk drive are gigging musicians, who might want to dump their computer-edited sequences to a small, portable system. For this purpose, the memory of the MDI-30 is rather small, sufficient for one song's bass line and pads, but the higher-priced MDI-40 may hold enough data for a fairly full production. I believe its real usefulness is as a sysex filing system, and, as such, the MDI-30 is priced below competing Yamaha and Alesis products.

Brother's current sequencers are not really designed for sophisticated MIDI-philes, but if you're a sequencer neophyte who doesn't want any type of editing functions, the MDI-30 may get you started.

Daniel Alan Phillips *graduated recently from the University of California-Berkeley with a B.A. in music. He sings and plays keyboards in the Bay Area-based band Rapid Transit.*

OVERALL 6

Features 5

Manual 6

Cost-Effectiveness 7

Ease of Use 7

Brother International Corp.
8 Corporate Pl.
Piscataway, NJ 08854
tel. (201) 981-0300

MIDImouse Wave for the Amiga ($249.95)

By Jeff Burger

A new wave (pun intended!) of MIDI software allows a computer to act as a digital synthesis design module for sounds that can then be transferred to traditional MIDI samplers. *Wave*, an Amiga program from MIDImouse Music, presents the musician with a toolbox of ten modules to create, combine, and modify sounds for devices supporting the MIDI Sample Dump Standard.

A wave is the program's smallest building block and can be created in several ways. The Wave Control window displays a large graph of a single wave cycle along with controls for specifying the frequency, amplitude, and geometric waveshape for each of 32 harmonics. After all the desired parameters are adjusted, the program calculates how the wave will sound and look. A wave can also be extracted from an existing sample downloaded from a MIDI sampler. Individual waves can be stored on disk for later use.

Since any real-world sound consists of a series of hundreds or thousands of waves that vary greatly, specifying them all would be tedious at best. The Generate Sound module comes to the rescue, using interpolation wavetable synthesis to create a smooth transition between up to 34 waves. Sliders are provided so each transition stage can be moved horizontally along the time line and given its own amplitude.

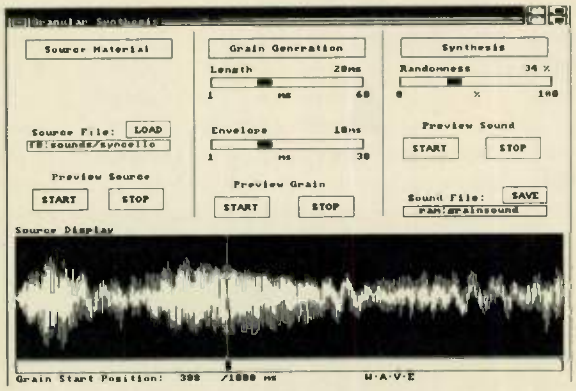
You can modify as well as create sounds. The Edit window resembles a traditional loop editor, including a full-screen waveform display and numeric readouts for sample and loop endpoints. Two sliders control sample window scroll and magnification; a third slider determines the position of an endpoint before selecting Set Start or Set End from the menu. Most edit functions operate on a range bounded by these endpoints. These functions include normal loops, alternating loops, copy, cut, paste, and insert.

Wave stores two sounds in memory simultaneously. Copy and Cut actually send sounds to the background memory slot and the Swap Sounds command

exchanges foreground and background sounds. The greatest application of this feature is in the Chorus/Modulation page, which features sliders for frequency modulation (both linear and exponential) as well as chorus and amplitude modulation.

Several other windows round out Wave's options. The Envelope Window supports independent 16-stage envelopes for filter (with independent cutoff frequency and resonance), amplitude, and pitch. The Mixer window can mix up to six files together, each with its own volume and transposition (this can take five or six minutes, given a complex task). The Animate Sound window produces a sort of oscilloscope effect by depicting the waveform in motion.

A separate program, the Phase module, creates sounds through phase-modulation synthesis (*à la* Casio products), where the modulating waveform alters the phase of the carrier wave over time. Two existing wave files are loaded in and displayed as modulator and carrier. An envelope of up to sixteen stages can be drawn to specify the amplitude modulation. The sound can be previewed internally in the Amiga and saved for use in the Wave program.



MIDImouse Wave

Wave also provides an introduction to granular synthesis through the Grain module, which creates a sound by excerpting a section of a sample up to 60 ms in length, called a "grain." A uniform attack and decay is added to that grain, and the grain is repeated over and over, with a random factor applied to the grain length each time to make things more interesting.

After loading a sample from which the grain is to be extracted, this window displays the sample waveform along with a slider that determines the point from which the grain will be taken. Other sliders adjust the grain length from 1 ms to 60 ms and the envelope time from 1 ms to half of the grain length. Finally, the percentage of randomness can be adjusted from 1% to 100%. The source file, grain, and final sound can each be previewed through the Amiga's internal voices. While interesting sounds are obtainable, it takes some work to have the result sound like something more useful than a swarm of angry bees.

The potential of Wave is very enticing, but the package as a whole seems poorly designed. File requestors are non-

standard, and all time specifications used in creating a sound are relative. A graphic interface is employed, but not optimally; only one window can be open at a time, forcing you to constantly open and close windows to get to various functions. While internal Amiga auditions are supported, you have to close the current window, save the file, go to (and possibly load) the Preview module, then reload the file. On the positive side, the auxiliary Phase and Grain modules can play sounds directly.

The Phase and Grain modules are completely separate programs, and their user interfaces are totally unlike Wave. Another problem is that neither has file requestors, so you have to remember the desired file names and type them into a box accurately—argh! To make things worse, if a file name is typed inaccurately, the error message comes up in the AmigaDOS CLI which is completely hidden by the program. These factors all contribute to the feeling that the package was kluged together.

All in all, most of the elements you might want in experimenting with digital synthesis are present in Wave. The operative word, however, is experiment. By nature, the types of calculation performed in this type of program are very time-consuming. The manual is scant and doesn't do a good job of introducing the program to the user and lacks detail about actual operations. It covers no applications or background on the wide variety of digital synthesis techniques the program supports, many of which will be new to the average user. Wave indeed opens up a new spectrum of digital synthesis alternatives for sampler owners, but its viability as a serious tool is hamstrung by limitations in the user interface and documentation. Let's hope a revised version that better taps Wave's potential is in the works.

5
OVERALL

Interface 3
Functions 8
Documentation 3

MIDImouse
Box 877
Welches, OR 97067
tel. (503) 622-4034

Celestion SR Compact Speakers ($199 each)

By George Petersen

I liked these speakers the first time I saw and heard them. At a mere seven pounds apiece, the combination of a single 5-inch driver in a ported, injection-molded cabinet delivers surprising performance from a rugged, ultraportable box. The Celestion SR Compact uses the same driver design as the larger speakers in the SR Series, with a large, egg-shaped aluminum dome that extends high-frequency response. The speaker has a sensitivity rating of 91 dB (1 watt at 1 meter) and a power-handling rating of 100 watts program material. Consequently, the Compact is able to provide a maximum sound pressure level of 111 dB, a respectable amount of sound from a small enclosure.

In operation, the SR Compacts offered clear, punchy reproduction of high and midrange material. Due to the size limitations of the driver and enclosure, bass reproduction has a lower limit

of 80 Hz, after which it falls off dramatically. However, this is certainly adequate for many applications, and if you require more bottom end, a subwoofer can be employed and hidden somewhere in the room. On one occasion, I added a bass cabinet with a JBL 15-inch woofer (fed through a separate mono amp) to the SR Compacts and was impressed with the results.

A wide variety of mounting hardware attaches to the two threaded metal inserts under its cabinet, and the speaker is equally at home on stage (as a mic stand monitor, frontfill, etc.), as a keyboard monitor in a MIDI studio, or as the basis for a mini P.A. system. Versatile and great-sounding, the Celestion SR Compacts are well worth checking out.

Overall Rating: 8. *Celestion Industries, Inc., 89 Doug Brown Way, Holliston, MA 01746; tel. (508) 429-6706.*

Sound Reinforcement Handbook ($34.95)

By Steve Oppenheimer

The second edition of Yamaha's *Sound Reinforcement Handbook* begins with what a sound system is and follows with a thorough overview of sound reinforcement, going beyond acoustics, terminology, components, and system structure to discuss specifications (and why the ears don't always correlate with specs), test equipment, and MIDI. Authors Gary Davis and Ralph Jones also explore pragmatic topics such as environmental factors in an outdoor venue. They even add a chapter on synchronization for multimedia shows and include an appendix on logarithms.

Because they cover so many topics, the authors have limited space to explain each; this is a handbook, not an encyclopedia. Nonetheless, the discussions are reasonably complete, aided greatly by excellent diagrams and charts. Discussions such as using transformers vs. differential amps in console design are not limited to theory and electrical and audio matters; Davis and Jones also touch on practical financial considerations.

The book is aimed at the beginner, and the math is kept relatively simple (nothing tougher than logarithms), but it moves quickly; the ideal reader is probably the musician or relatively inexperienced sound person who has technical leanings but lacks a solid knowledge of sound reinforcement. (Professional engineers could use the book for a refresher but might be better off with something like *The Handbook for Audio Engineers: The New Audio Cyclopedia*, edited by Glen Ballou [Howard W. Sams & Co.], for theoretical information.)

The discussion of the decibel is particularly impressive because the authors not only cogently define the decibel but explain why and under what circumstances it is meaningful to correlate its variations (i.e., dBm, dBu, dBW, etc.). Far from an anomaly, the dB discussion is typical of the approach taken throughout the book. The reader is not talked down to but is quickly raised to a higher level of understanding. This is one of the best beginning- to intermediate-level books on sound reinforcement. (Both books are also available from Mix Bookshelf; see FYI page for more information.)

Overall Rating: 9. *Hal Leonard Books, 7777 W. Bluemound Rd., PO Box 13819, Milwaukee, WI 53213; tel. (414) 774-3630.* ■

Rane MPE 14 MIDI Graphic Equalizer

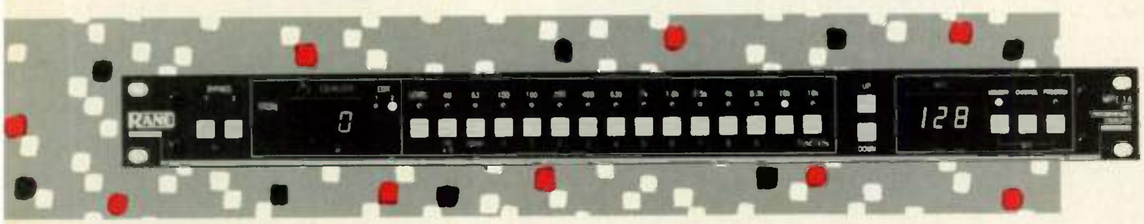
By Craig Anderton

Graphic equalization gets a lot more interesting when you can change frequency response dynamically under MIDI control. But is it worth the extra cost?

Graphic equalizers are very useful set-and-forget devices, but I've often wondered what it would be like to control the response in real time via a sequencer—bring up some midrange here and there to accent a drum part, boost a guitar's treble during solos then slide back to normal response, or even add rhythmic, equalization-dependent stereo effects. I was so curious about this that a few NAMM shows ago, when I heard that Rane was coming out with a MIDI graphic EQ, I tried to convince them (actually, maybe "pester" is the right word) this was an idea worth pursuing.

Much to my surprise, several months later I got a call from the project engineer. Turns out he not only got MIDI in there, but went way beyond my original wish list, and would I like an MPE 14 sent to me for review...

So I got one of the very first ones and I must say, this is a whole new type of tool. I'm rethinking EQ; it's not just a problem-solver anymore, it's a dynamically variable effect in its own right, and some of the things you can do are pretty wild. But first, let's look at what makes the MPE 14 tick.



BASIC SPECS

The MPE 14 is a 2-channel, 2/3-octave, fourteen bands-per-channel graphic equalizer, which lists for under $800 and takes up one rack space. Other models include the MPE 47 (4-channel, seven bands-per-channel, with filters optimized for tone control applications, $799), and MPE 28 (1-channel, 1/3-octave, 28 bands, $749). All are similar in features and operation.

Each unit stores 128 programmable EQ curves (programs), each of which can have its own programmed master level (useful for balancing out level differences caused by different curves). Like most modern signal processors, a mapping function can redirect incoming MIDI program change messages to call up the program of your choice (which can be displayed in octal or decimal—convenient for those whose synths number programs in octal).

One very nice touch is that you can set a global rate at which one program changes to another. Steps can be as small as 1 dB, which produces the slowest "crossfade" and eliminates any pops caused by sudden audio changes, all the way up to 27 dB (provides an almost instant changeover between programs).

In addition to the 128 main programs, there are also a bunch of factory ROM programs contributed by a variety of MPE users. Unfortunately, these were not finished by the time the review unit was shipped so I can't comment on their usefulness or exact number. Selecting either type of program loads it into a buffer, where it can be edited and optionally stored to one of the 128 main programs. The ROM programs are not accessible directly via MIDI program changes; they must be brought into the buffer and resaved as one of the 128 main programs.

THE INS AND OUTS

Each channel can be edited independently of the other and has its own bypass switch and overload indicator. The input/output structure is very flexible, providing XLR and 1/4-inch phone connectors capable of handling balanced or

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unbalanced inputs and outputs. The input impedance is 20kΩ balanced, 15kΩ unbalanced; this will interface with most pro gear, but not with low-level devices such as microphones or guitars. However, there is a schematic provided with the MPE 14, and it looks fairly easy to modify it for higher gain and input impedance.

LET'S PLAY EDITOR

The user interface consists of two very readable LED digit displays, 24 buttons, and twenty LEDs. It's not totally intuitive, especially for some of the more esoteric functions, but I was a bit surprised at how rapidly I became comfortable with the various functions. The manual is thorough and laced with just enough casual humor to keep one's interest.

Programming a curve involves selecting the channel (1, 2, or both) to be programmed, hitting the switch for a given frequency band, and using the up/down buttons to set the degree of boost or cut. You can compare what's in the buffer with the original program, as well as set all bands to zero if you want to start from scratch. There are also two levels of security access. You can lock out all editing (but still select programs), or lock out the front panel altogether. I can see a lot of studios making good use of the first option.

The MPE can combine a stored curve with the one in the buffer; for example, suppose you come up with a great curve for a particular vocalist, but then decide it needs a bit more treble. Rather than mess with your carefully worked-out curve, you can simply set up another curve that boosts the upper range a bit and add it to the existing curve. If adding the two curves produces an illegal setting (i.e., resulting in a boost greater than +12 dB or cut greater than -15 dB), the MPE will scale the results automatically for the best fit. Nifty.

MAXIMUM MIDI

Of course, you can choose omni or poly mode and a device ID so that multiple MPEs can coexist peacefully on the same bus. But the real excitement lies with what Rane calls expression parameters,

Product Summary

PRODUCT:
MPE 14

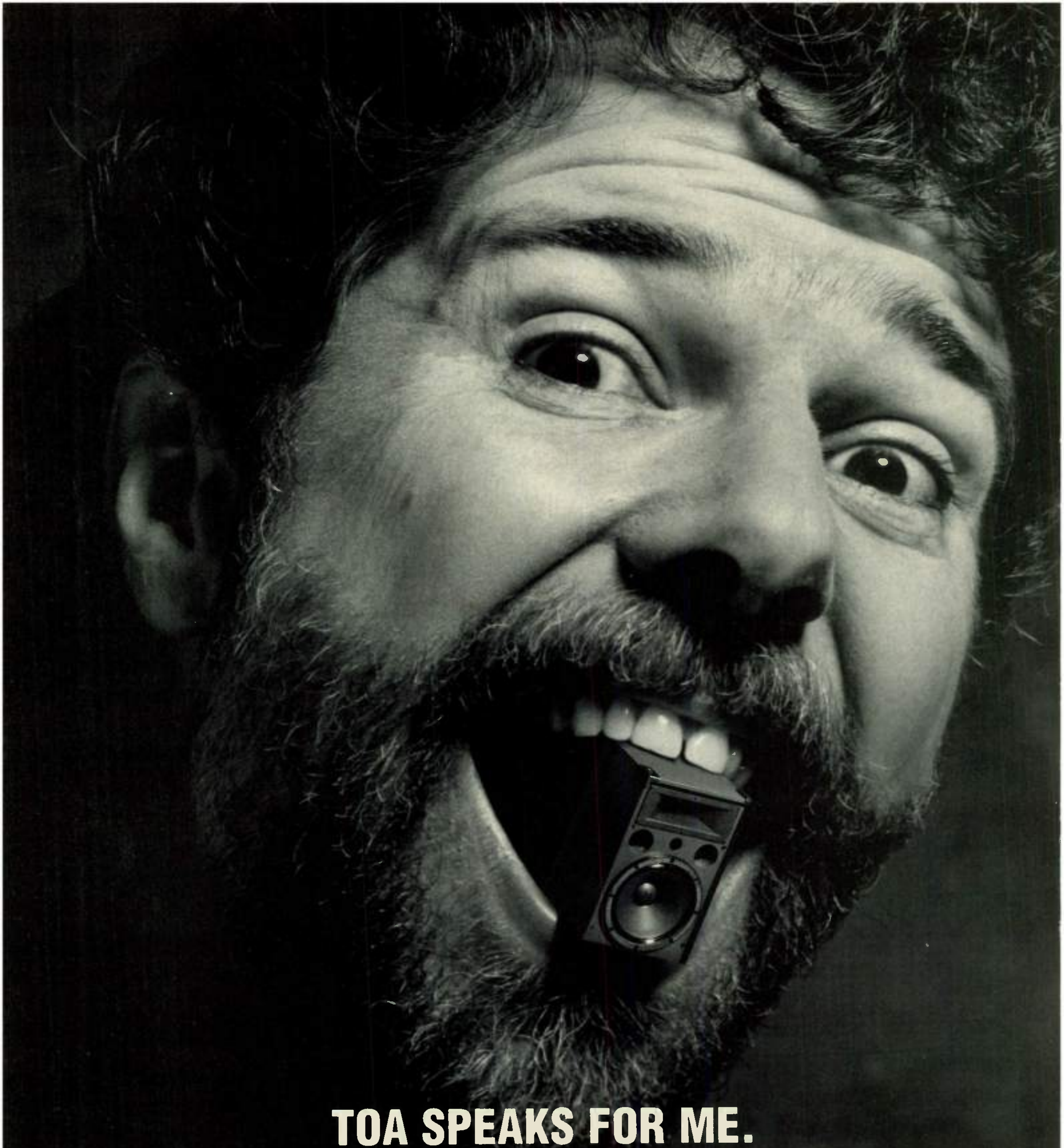
TYPE:
MIDI programmable graphic EQ

PRICE:
$799

MAIN FEATURES:
Dual 14-band graphic equalizer with 128 user-programmable and 128 ROM programs, dynamic boost/cut control via MIDI, XLR, and ¼-inch balanced/unbalanced inputs and outputs.

MANUFACTURER:
Rane Corporation
10802 47th Ave. W.
Everett, WA 98204
tel. (206) 355-6000

OVERALL 8
MIDI Implementation 8
User Interface 8
Cost-Effectiveness 7
Design 9


TOA SPEAKS FOR ME.

which lets you change EQ levels in response to MIDI messages.

You can set the unit to respond to any one of MIDI continuous controllers 1 to 120, all controllers, or channel (mono) aftertouch. An auto-detect option sets up the MPE to respond to whatever controller comes in next—great if you're not sure whether that footpedal connected to your synth is sending over controller 4 or was previously assigned to something else.

Each band's "vector" (direction of travel) can be set independently to boost or cut as the controller value increases (controller data can also be ignored). So if you wanted aftertouch to increase the bass, you could not only boost the lower bands according to how much aftertouch you apply, but cut the upper bands as well while leaving the midrange alone. It's not possible to set the scale of the boost or cut (e.g., have one band boost a little bit and another band boost a lot for a given controller setting), but that's getting pretty picky—and to scale each band would make programming time-consuming, anyway.

If all bands are centered and set to boost in response to a MIDI continuous controller, each band will change from 0 to +12 dB as the controller goes from minimum to maximum; there's no adjustment per se for controller sensitivity. However, there is a workaround: Set all bands to a common amount of boost or cut to provide an offset. For example, if all bands are set to +10 boost instead of centered, varying the MIDI controller from minimum to maximum will produce only a +2 dB change because the MPE cannot provide boosts exceeding 12 dB. Adjust the overall level with the level adjustment. (If you're concerned that boosting or cutting by a constant will mess up the sound due to passband ripple, don't worry; see the sidebar "Bench Test Specs.")

The system exclusive (sysex) implementation is also very complete, offering four main modes: program change (changing programs on the master changes programs on any slaves), memory dump (dump all, or selected portions, of memory), key scan (any slave MPE responds to key presses made on the master unit as if its own keys were being pressed), and parameter echo

BENCH TEST SPECS

S/N ratio unweighted, maximum gain, all bands centered: 101 dB. This meets the specs quoted in the manual.

Bands: 40, 63, 100, 160, 250, 400, 630, 1k, 1.6k, 2.5k, 4k, 6.3k, 10k, 16k, each frequency-accurate to better than ±0.5% (limited by resolution of test oscillator)

dB boost and cut at 1 kHz: 11.5/14.4 (slightly less than the 12/15 claimed)

Maximum output level: +20 dBu

Headroom: 4.5 dB after overload indicator lights

Passband ripple, all bands at maximum boost: less than 0.5 dB. This impressively flat response allows for "biasing" bands up or down should you need reverse response with MIDI continuous controllers (e.g., boosting all bands first gives a wider cut range). Also note that the filter sections interpolate; if you boost two adjacent bands, you'll get a broad boost located between the two bands.

(any parameter changes on the MPE are sent as sysex). The latter, when recorded into a suitable sequencer, can control the MPE dynamically during playback. This allows for even greater precision than simply tying parameters to a controller, but does require a bit more thought to apply.

APPLICATIONS

A device like this offers a field day for sonic explorers. Mixdown automation, which is what I wanted originally , fits the MPE 14 like a glove. Being able to vary equalization dynamically is wonderful—for example, suppose you have a guitar playing behind a singer. You can dip the midrange a bit during the vocals to make room for the voice, then bring the sound back to normal during fills—or change the EQ altogether for a lead. Or how about creating a "loudness" control for your synth? Offset the high and low bands positively, and set them for negative response to pedal motion; pushing the volume pedal down brings the highs and lows back into balance, but as you pull back on the pedal, the highs and lows increase a bit.

Keyboard players can increase the treble on horn parts with more aftertouch, or beef up the bass on pipe organ sounds. Wild. It's also possible to provide synchro-sonic effects by creating repetitive controller patterns in a sequencer and driving the MPE 14 from the sequencer.

Another application stems from the fact that I've never been too happy with the EQ sections in most guitar-oriented multi-effects processors. I plan to use the MPE in dual mono mode, with one channel before the processor and one after, and have it change programs along with the multi-effects so each sound can have its own EQ.

Here's one more tip: If you want to cut and then boost a sound under MIDI control (e.g., fade out the midrange on a rhythm guitar part but fade in some midrange boost during a solo), no problem. Set up one curve for the midrange cut and another for the boost, then issue a program change command when it's time to go from cut to boost.

Taking the larger view, most of the time we add dynamics to a mix with level changes. Changing EQ instead of (or in addition to) level provides another dimension of control. When you pull something back for a fade, reduce some of the highs as well. Want to accent a snare hit? Don't boost the volume, but kick in some extra upper midrange for a snare that really "cracks." My only complaint about the MIDI implementation is that there should be a way to reverse the curve of the incoming continuous controller; this would make it easier to boost or cut response with decreasing or increasing controller values, respectively, without having to resort to the offset workaround described earlier. Some way to control sensitivity would also be helpful.

Do I like the MPE 14? You bet, and I'm really looking forward to uncovering what it can do. It may seem expensive compared to other EQs that don't allow MIDI control, but just remember, if you think of this as only a graphic EQ you're missing the point—the fun starts when you feed it controller data and start experimenting. ■

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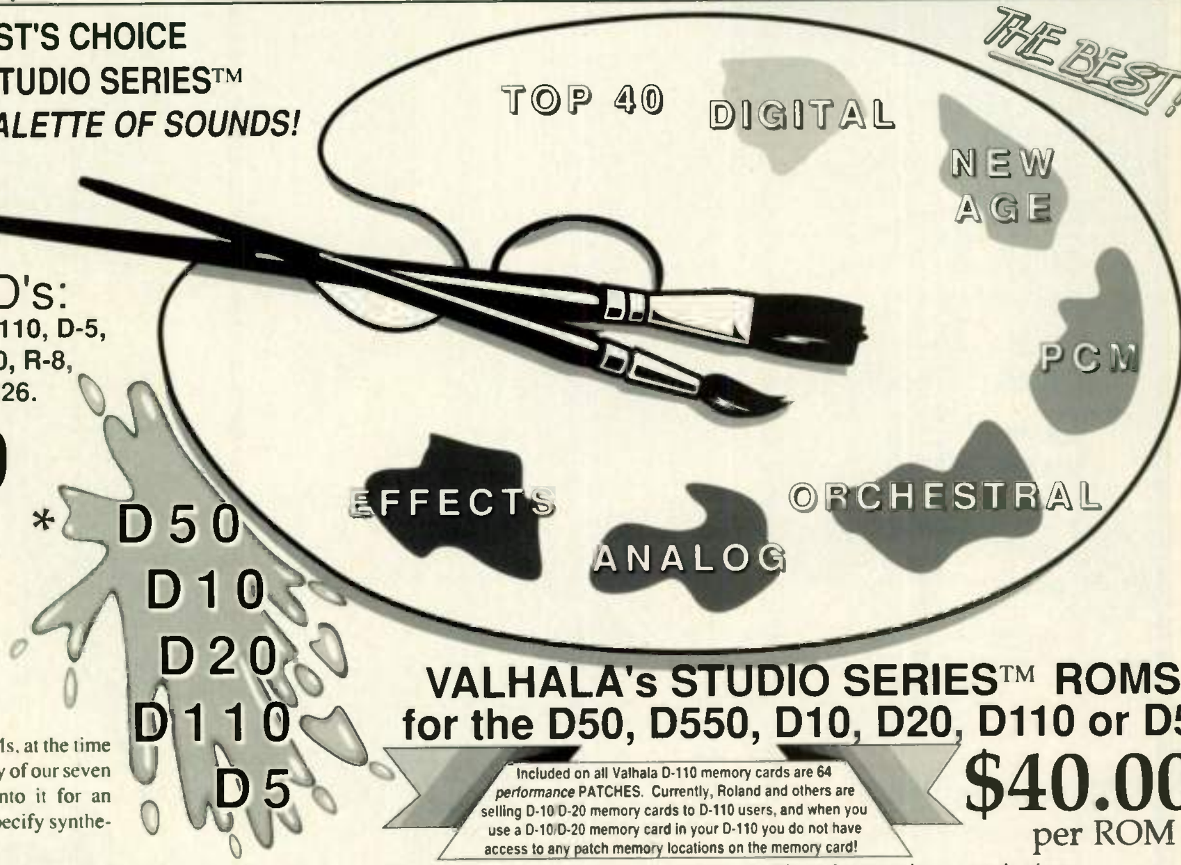

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Coda MusicProse

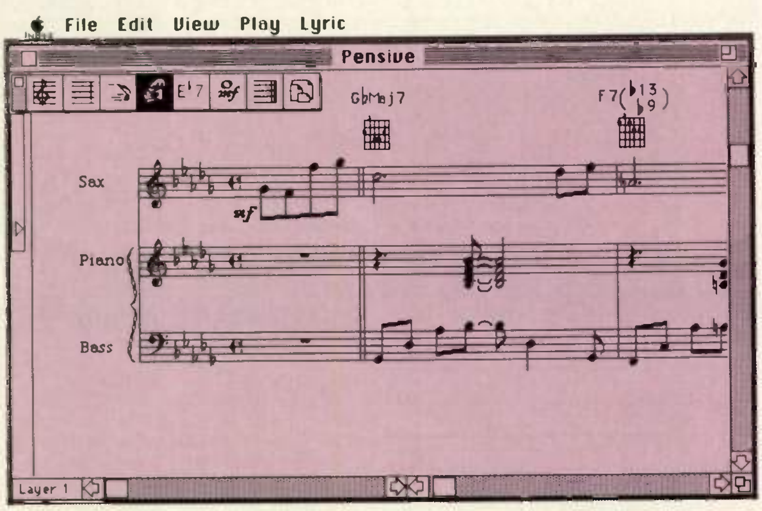
By Chris Many

More than just a low-end version of Finale, *the company's newest notation package offers a completely redefined user interace that's easy to use.*

When Coda released *Finale* for the Macintosh, the hoopla that preceded it showed how much the music software industry had matured. A full-featured, powerful, slickly packaged product with a price tag to match, Finale was in a class all its own when it came to music notation software. But there was one other price to pay besides the initial $1,000—since lowered to $600 and now raised again to $750 for Version 2.0—that didn't come across in the four-color, two-page ads: time. If you really wanted to put the program through its paces, you might have to spend the next decade learning to use all of its many features.

MusicProse is Coda's scaled-down version of Finale, with an excellent manual that cuts the learning curve down to size and a price tag that puts a full-featured notation program within most Mac-based musicians' budget. Don't get the wrong idea when you read the words "scaled down"; if you "scaled down" the Empire State building, you'd still have a Trump Tower. Built on ENIGMA (Environment for Notation, utilizing Intelligent Graphic Music Algorithms), the same engine that powers Finale, MusicProse incorporates many of Finale's features and adds some of its own. This program can go head-to-head with any mid-priced notation program on the market.

MusicProse is much easier to use than Finale; the interface keeps you on the right track without denying you any of the functionality and power available, and it's all presented in an uncluttered, crisp manner. So much has been written about making software "user friendly" that it's a cliché. But for a notation program with a lot of features, so much depends on the ease-of-use and simplicity of interface that it can make or break a program. It's evident Coda is listening.



The main window from MusicProse, showing the tool bar across the top, with the Lyric tool selected.

MusicProse comes nicely packaged with three disks, a manual, a quick reference guide, and plenty of templates and examples. The first thing you'll notice is the documentation. In an industry littered with incomplete, convoluted, and just plain hard-to-learn program information, MusicProse's manual is a wonderfully gradient approach to learning everything you need to know about the software. Combined with a generally intuitive interface, you are able to learn the program easily and (dare I say it?) quickly, with little or no frustration. The tutorials do an excellent job of taking you through notating a lead sheet, expanding that to a four-part chorale, and expanding it further to a full score. Bit by bit you learn the various features, speed up your input by using quicker methods, and get comfortable with the software.

MusicProse contains all the essentials required of notation software in today's marketplace. You can pretty much take

FINALLY!

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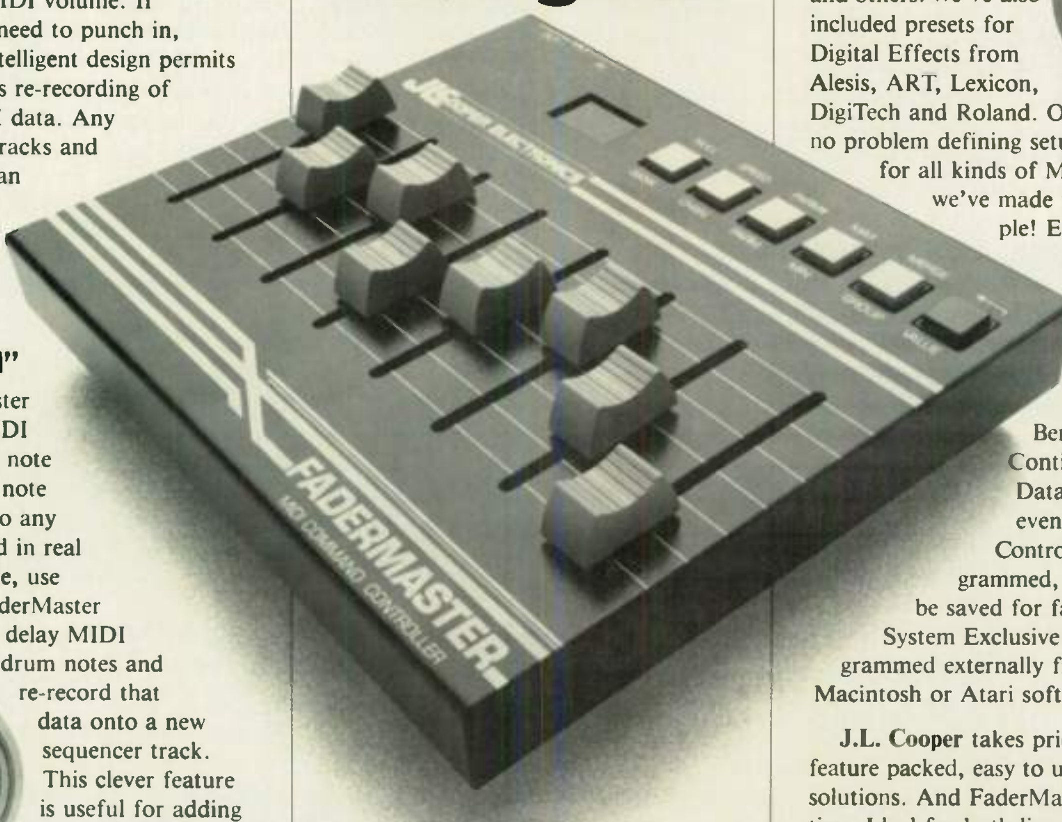

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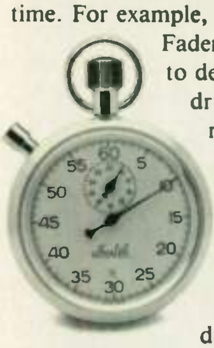

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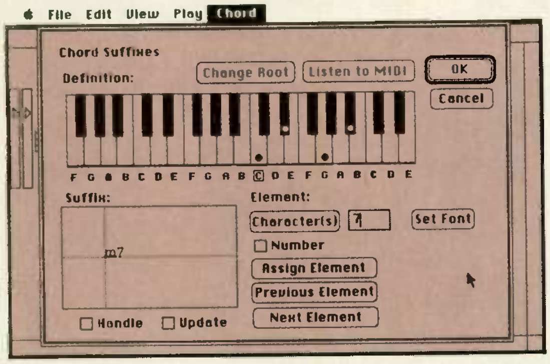

With MusicProse's Chord Suffix dialog box, you can create your own chord defintions and add them to the existing library.

for granted that the main functions you expect to see, you'll find well-implemented in MusicProse, but the features that set it apart from the crowd do so in a big way.

GIVE US THE TOOLS

MusicProse provides you with an eight-icon tool palette from which you select the main editing functions of the program. As you select the tool you wish to use, an additional menu line appears, from which you can select a variety of features. Choosing another tool replaces the menu line with one specific to that tool. You are directed to the functions you can access from the tool you are using. It sounds simple, and it is. If you've ever dealt with a program that lets you do everything at once, all crammed into a small amount of screen real estate, you'll appreciate this method of organization.

The main tool palette can be visible or hidden, but it requires so little space you'll probably just leave it there for convenient access. Any one of the eight tools can be selected either using the mouse, or via the keyboard, depending on your preference. In addition to this tool box, there are four other self-explanatory menus that are always available: File, Edit, View, and Play.

The tools are a straightforward implementation of the functions you normally use. The Staff tool provides the means of creating or deleting a single staff or groups of staves (up to the program's limit of eight), bracketing, transposing staffs, and assigning a variety of attributes to them (measure numbers, rehearsal marks, etc.) across the whole staff. The Measure tool covers selecting, adding, inserting, and deleting measures, as well as cutting-and-pasting, transposing, beat positioning, and a host of other functions. The Page tool generally administrates the look of your music. A Repeat tool covers adding graphic repeats or textual ones (such as D.C., D.S. al Fine, etc.); these work in conjunction with playback controls, so when you hear your score back, it will accurately follow all the standard music notation repeat symbols.

The Lyric tool has been very elegantly implemented and features the simplest method of lyric insertion against notes I've seen. You can type lyrics out before you slot them into your lead sheet or write the score and position the lyrics after the fact.

Product Summary

PRODUCT:
MusicProse
TYPE:
Music notation software
FEATURES:
HyperScribe real-time transcription; good documentation; PostScript output; standard MIDI file support
SYSTEM REQUIREMENTS:
Macintosh Plus/SE/II with System 4.2 or later and 1 MB RAM; printer
PRICE:
$249
MANUFACTURER:
Coda Music Software
1401 East 79th St.
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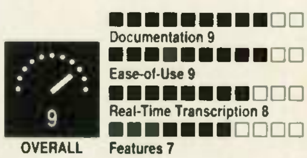

Based on an analysis algorithm of chord suffixes, MusicProse can determine what the chord you're playing is and add the symbol for that chord. However, the analysis might be different than what you had in mind (e.g., the notes C-E-G-A might be recognized as an Am7/C chord when you meant it to be a C6); as usual, computerized analysis requires human correction. Using the Chord tool, you can add the appropriate chord symbols. Guitar tablature (fingerboard charts) is also available.

The Expression tool will provide you with a number of different palettes from which to choose symbols: articulation, dynamics, shapes (ties, slurs, hairpins, etc.), and text (Andante, Allegro, Presto, etc.). Mouse technique works best in inserting these to the score, and symbols and shapes can be placed and dragged into the right location, expanding over bars as needed.

AND WE WILL DO THE JOB

The Entry tool establishes which method(s) you're going to use in entering music into MusicProse; you can choose one, or a combination of several. The most basic method, Simple Entry, involves clicking and dragging with the mouse to step-enter and place data, or using the keypad to select note values; nothing new here.

An alternative is the use of Speedy Entry keys, a combination of the keypad and the entire computer keyboard, which becomes, in essence, a music typewriter. Once you've learned the location of each note, symbol, etc., you can get up to speed using just the computer's keyboard. Using this entry method, you can choose to have the computer become "musically intelligent," in which case it only allows you to enter in the correct number of beats to any given measure and then automatically jumps to the next measure. There's also Speedy Entry using a MIDI keyboard and the keypad/computer keyboard, where playing single notes on the MIDI instrument logs the notes, and you set the note values using the keypad.

If you've already spent the hours playing all your parts perfectly into your sequencer, why should you have to do it all over again? Providing we're talking standard MIDI file format, MusicProse does a good job of turning your sequence into notated music. MIDI files come in three formats: all tracks on one staff, multiple tracks on multiple staves, different sections into different staves (verse 1 on the first staff, the chorus on the second, etc.). If there are more than eight tracks within the MIDI file, track numbers 9 and up are ignored, so you may need to return to your sequence and rearrange it to fit within the first eight tracks. You can also split a MIDI file track into two staves, helpful if you're transferring a keyboard part. And good news for drum programmers: After you've transferred over your drum parts, you can choose a rhythmic notation method for the percussion sections.

HYPER ACTIVITY

The feature most people loved with Finale, the HyperScribe function, is also included with MusicProse. HyperScribe is a real-time transcription function where you play your MIDI instrument and MusicProse notates it on the screen as you play.

MusicProse does a pretty good job of figuring out what you mean as you do this, all things considered. Just like working with a sequencer, you'll have to establish a quantization value, but do so *before* you play. You'll still have to go through each measure of each track

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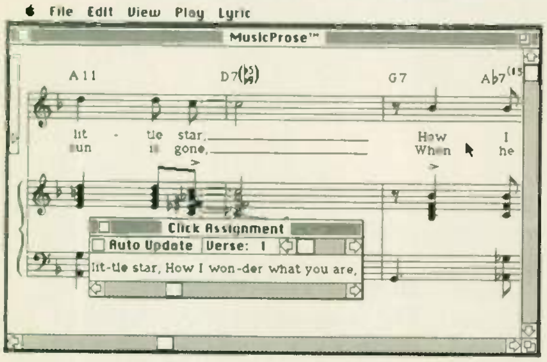
once they get transferred to verify the accuracy of the computer algorithms in translating your performance into notated music (don't expect much success with a polyrhythmic lead line, for example), and there will be some editing required to repair wrong notes you played. But I'd rather play most of my music in via a keyboard, correcting the few inaccuracies, than spend a couple of hours typing it on or pointing and clicking for every note. You can also split your performance out into two staves. By selecting the split point, MusicProse will divide your performance up into treble and bass clefs, very handy if you're creating a keyboard part.

The MIDI implementation is good, allowing you to assign MIDI channels to staves and hear your piece played across multiple channels. With the use of multitimbral instruments, you can get a good idea of your music from the playback mode. Keep in mind, MusicProse is not a sequencer and makes no airs about being one. It won't play back in a swing feel, for instance, and prefers that you compile your work into a playback file. Also note that, unlike Finale, MusicProse does not turn expression marks, such as dynamic markings, into playable MIDI data.



Using the Lyric tool and the Click Assignment option, you can easily and quickly place song lyrics on a page.

ALL THE NOTES THAT FIT, WE PRINT

The printing options are very simple and allow for printing on ImageWriter printers, ink jet printers such as the HP DeskWriter, or PostScript Laser printers. Petrucci and Seville screen fonts in three sizes are shipped with the program. Petrucci is the primary music font used with MusicProse, and Seville is a font for guitar fingerboard notation. Laser printer fonts for Petrucci and Seville are available separately from Coda for $89. In addition, three other music fonts, priced at $69 each for the screen and printer versions, are also available: Midicom for MIDI data and event notation; Newport, a jazz and percussion notation font; and Rameau, for theory and analysis notation. The printing quality is excellent whether using laser, ink jet, or dot matrix.

There are some things to be aware of when using MusicProse (aren't there always?). The screen-refresh rate can be abysmally slow. You can turn this off in HyperScribe, but if you have to sit through a lot of notes being redrawn on a slow Mac, it's distracting. You are limited to eight staves, so MusicProse isn't as useful for full-length orchestral scores, unless you're willing to work with only eight staves at a time. And no matter how easy it is to learn MusicProse, there is still a considerable amount of time required to become accomplished at using the software, though probably no more so than a mid-level sequencer. It also takes a lot of user correction and time to get a decent-looking, errorless score. (And while I'm on my soapbox, I would love to see some software or hardware with a graphics pad where you could draw notes in using a stylus/pen that interfaced with any of these notation packages.)

I'm sure we'll see conversions, but for now, you'll need a Mac to run MusicProse. Atari ST owners can use almost all of the features, excluding MIDI portions, by using Spectre GCR, a Macintosh emulation system.

Overall, MusicProse is powerful enough for most applications and is engineered for the ease-of-use most musicians want from a software package. If you're in the market for a notation program for the Macintosh, don't need large orchestral scores and every conceivable option, and aren't ready to lay out the bucks and invest the time into Finale, MusicProse will fill your notation needs.

Chris Many, *a composer/synthesist and Synclavier programmer who writes music for film, television, commercials, and industrials, is a member of Celestial Navigations, whose second album was recently released on the Nouveau label.*

(continued from page 115)

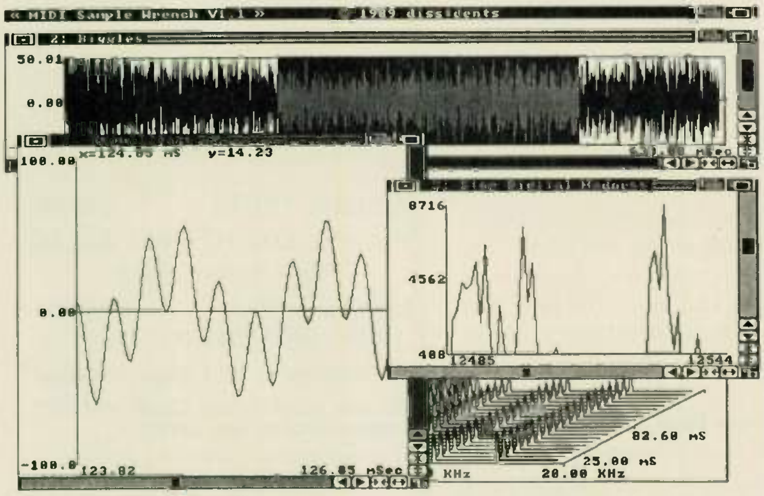
dissidents MIDI Sample Wrench

By Jeff Burger

Amiga fans, the wait is over: If you're trying to wrench high performance out of a sampler, here's a program that gets your sampler and computer talking to each other.

Amiga music software has finally evolved beyond the sequencer level. One recent entry in the Amiga software sweepstakes, *MIDI Sample Wrench,* is a digital sample editor that works with any MIDI sample dump standard-compatible sampler and with the Akai S700, S900, and S950. Two separate libraries are currently included—one for 12-bit samplers and a second for 16-bit machines—but the program is expandable, and additional libraries can be added to accommodate future products.

The program adheres to the standard Amiga environment (e.g., pull-down menus and point-and-click item selection). Machines with 512K will work, but at least 1 MB is recommended if you plan to multitask or edit long samples. Unfortunately, Sample Wrench uses an annoyingly popular (at least in the Amiga community) copy-protection scheme that forces the user to look up and enter a word from the owner's manual upon start-up. This approach inconveniences legitimate users by assuming (inaccurately) that software pirates won't copy the documentation as well as a disk if they want the program.



MIDI Sample Wrench showing several views of a sampled waveform.

SO WHAT'S ON THE MENU?

The program's primary menu bar includes three main categories: Project, Setup, and MIDI. The Project menu lets you open each of three identical sample-editing windows (multiple windows allow for easy cutting and pasting between different sample files). Upon opening one or more edit windows, the menu bar changes to a series of in-depth sample-editing commands. Each window and matching menu bar selections use different colors, making it easy to differentiate the current window from others .

The Setup menu provides palette customization options for the various windows. The display can also be set to 200-line (noninterlaced) or 400-line (interlaced) mode. This menu chooses between several backup modes for the editor's undo feature, since large clipboards of sample data (created by copy or cut operations) can require a lot of memory. The backup options are None (no undo feature), Internal (ability to undo depends on available memory), or External (clipboard data is stored to floppy disk).

The MIDI menu selects between the various sampler libraries. It also includes sample send and receive commands, where you specify the sample, bank, MIDI channel, and edit window. The receive dialog box also lets you name the incoming sample and will open the appropriate edit window automatically.

The editor menu items let you set loops, display options, determine playback parameters, select up to six different sample "clips" from the clipboard, and the like. Ninety percent of the submenu items have keyboard equivalents. The sample menu has options for basic sample management, such as load, save, save as, delete, name, play, and info.

After loading a file, the edit window displays the entire waveform. The X axis represents time or sample length in computer words, and the Y axis shows amplitude in absolute units or as a percentage (these choices are made in the Options menu). The X-Y Read Out item in the Options menu provides constant coordinate displays for the mouse pointer as it navigates around the waveform.

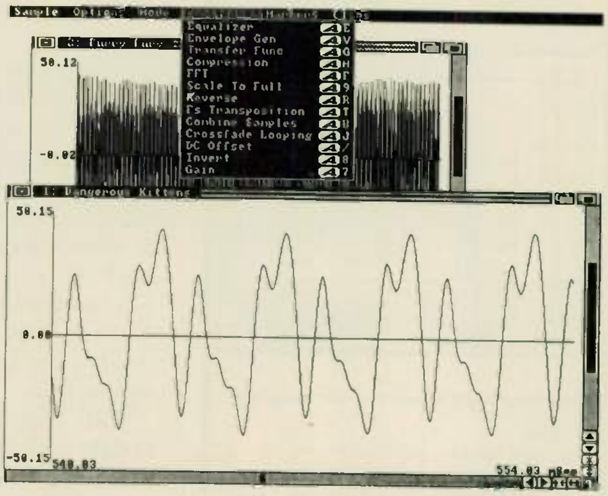
The edit windows offer independent gadgets for horizontal and vertical zoom and standard scroll bars for moving around a magnified wave. The size of the grab areas of the scroll bars scales automatically to reflect the proportion of the displayed segment relative to the entire wave size; however, another option lets you "lasso" a portion of the waveform with the mouse, which then fills the entire edit window. The Show Full menu item zooms out to the original full-wave display.

SOUNDING OFF

Sounds can be auditioned from the sampler itself (Remote mode, which features selectable MIDI channel and degree of transposition), or they can be previewed from the Amiga's internal digital synthesizer (Local mode). Although using Local mode yields a lower sound quality, it circumvents the time-consuming process of downloading the waveform from the computer's memory into the sampler after each edit. In either case, the Amiga's keyboard turns into a two-octave clavier.

THE ART OF LOOPOLOGY

Creating optimal loops is a primary function of any sample editor. In MIDI Sample Wrench, loop points are associated with *markers*—up to six vertical lines that can be placed at any point along the waveform's X axis. Markers can be set numerically by entering individual sample numbers, or by pointing to a place on the sample with the mouse.



Main screen with pull-down functions menu.

Either marker placement method is usable with Auto-locate, which forces a marker to the closest value adhering to a particular set of specifications, as set in the Auto-locate requester box. The

• SAMPLE WRENCH

Multiplier value forces the marker to the closest integer multiple of that value, Offset specifies how much of the beginning of the wave to exclude (for example, the attack transient), and the Zero Crossing button forces the marker to conform to a point in the waveform with zero amplitude. It would be nice to be able to invoke these suboptions directly. For example, to adjust the position of a marker placed with the mouse, you have to go back to the requester instead of just grabbing the marker and moving it.

MIDI Sample Wrench supports both sustain and release loops (assuming your sampler supports these). The Set Loops option brings up a requester box that associates the start and end points for both loops with the desired marker numbers. The loops can either be forward only or forward/backward, and the loop points can be adjusted by moving the associated markers. An Info option shows the sample position of any of the defined markers, the loop markers and type, file size, disk path, and backup method.

CLIPPING AND CRUNCHING

Traditional editing functions such as copy, cut, paste, and erase have the added ability to store six individually addressable "clips" in the program's clipboard. The clips can be defined with markers or the mouse and given names. All editing is performed on the active clip, which is selected from a list of defined clips.

MIDI Sample Wrench also contains some useful DSP (digital signal processing) features, although most take a while to do the necessary number-crunching after you have selected the desired parameters. Equalization functions include shelving, parametric, or lowpass/highpass mode. The cutoff or center frequency can range from 5 Hz to 95% of the Nyquist rate (i.e., half the sampling rate). Parametric and shelving modes allow up to ±20 dB of boost and cut. Unfortunately, only one frequency range can be manipulated in a single pass, and there is no graphic display.

Envelope Gen brings up a graphic window that lets you draw an amplitude envelope and impose it on a sample. The Transfer function allows you to draw a graph depicting the relationship between the output and input.

A compressor option offers settings for threshold, ratio, attack and release. (A ratio of 10:1, for example, will turn a 10 dB increase into a 1 dB increase; a ratio of 0.2:1 would create expansion of 1 dB for each 0.2 dB of change.) An FFT (Fast-Fourier Transform) doesn't actually alter the waveform, but does provide spectrum analysis via a pseudo-three-dimensional graph plotting frequency (x-axis), amplitude (y-axis), and time (z- axis). This can be handy for seeing the effects of equalization.

Scale To Full "normalizes" the waveform by increasing its peaks to the maximum possible short of clipping, thus increasing the output level. Reverse flips the wave front-to-back and Invert flips it top-to-bottom (i.e., a 180-degree phase change). Fs Transposition is a resampling process that allows samples created at one sample rate to be changed to a different sample, thus providing compatibility when transferring samples between samplers using different sample rates. Combine Samples mixes two waves; DC offset is occasionally useful for adjusting offset anomalies caused by the sampling process or a nonlinear Transfer process. And what sample editor would be complete without crossfade

Product Summary

PRODUCT:
MIDI Sample Wrench
TYPE:
Sample editor
PRICE:
$279; free upgrades to registered users of V. 1.0
HARDWARE REQUIREMENTS:
Amiga 500/1000/2000 (512K required, 1 MB RAM recommended)
FEATURES:
Compatible with 12- and 16-bit sample dump standard instruments, includes digital signal processing functions, multiple windows
MANUFACTURER:
dissidents
730 Dawes Ave.
Utica, NY 13502
tel. (315) 797-0343

Interface 8
Functions 8
Documentation 8
8 OVERALL

looping? MIDI Sample Wrench provides sampling enthusiasts with a variety of options for smoothing and blending waveforms at their loop points.

THE WRENCHING CONCLUSION

The program operates as specified, and all the required functions are present. What's more, the manual is decent and includes both reference and tutorial sections. My only complaint is that some functions, such as EQ, don't exploit graphics as much as I would like. And the less said about the copy-protection scheme, the better! (*But we'll mention it anyway because the manufacturer has modified the copy-protection scheme. You can send your master disk and $30 to dissidents, and the company will return a custom-compiled, unprotected disk that displays your name when it boots.—SO*) Nonetheless, MIDI Sample Wrench is a fairly solid offering that finally brings relatively sophisticated sample editing to the Amiga.

Jeff Burger *vacillates between bragging about his Amiga and throwing it out the window. His credits include music composition/production, computer graphics/programming, video production, sales/marketing, and technical writing. He is currently associate editor at EQ magazine.*

Music:
Voice of the Cheese, Part 2

By Robert Carlberg

Last month we printed the first half of the "I Wanna Be a Cheesehead" readers' reviews. Here's the best of the rest.

Life as a musician is not for the thin-skinned, as you will see. Last July, we offered readers the chance to be a reviewer and some music the chance for exposure it otherwise might not receive. This month we continue with reviews of our readers by our readers. (For additional commentary on the "cheesehead" phenomenon, see the "Back Page" this issue on page 146.)

From Mark Ingram of Purdy, MO:
In a sincere attempt to be fair, I have listened to these tapes almost exclusively for the last six weeks. They still defy classification, and oh, how I hunger for Mozart!

In replying to the **EM** "Cheesehead" offer, I was hoping to receive examples of classical or jazz music demonstrating state-of-the-art electronics, and large-

Ingram, Mark E.

Schneebeli, Scot

Willoughby, Brian

scale MIDI sequencing at its best. Unfortunately, this was not to be. What came in the mail were three tapes, *Lux Aeterna/Astralingua* and *Coal Aberrations* by **Henry E. Schneider** and collaborators, and *Crossway of Dreams* by **Alesh Jarmar**. In short, this stuff lacks many of the structural elements that generally define music in the first place: tempo, meter, exposition and recapitulation of theme, variation, chord progression, and, in the case of much of *Coal Aberrations,* even chords in the strictest sense of the word (at least three simultaneous notes). *Aberrations* is more accurately described as a track of "sound effects"—cosmic in its reach, but definitely *not* in its grasp.

Interestingly, Schneider et. al. and Jarmar both use sampled thunderstorms and rain showers for a quite *literal* rendering of the phenomena; but they (and the intended listener) should go back to one of their best known *metaphorical* antecedents, the *Pastoral* (6th) symphony of Beethoven. The contrast is marked, the wonderful musicality of the latter overwhelming.

Likewise with Jarmar's synthesized sounds of wind blowing—a nice touch, but the anemic musical setting makes it a pale imitation of a section of Richard Strauss's *Don Quixote.* This should be familiar territory to Jarmar, described as a world-acclaimed, classically trained organist and composer. His academic credits and film score awards are impressive indeed, but if there is keyboard and compositional virtuosity here, it is obscured by the gauziness of deeply layered synth sounds.

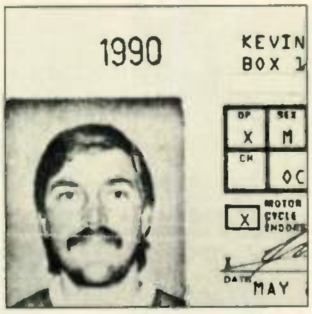
In fairness to Jarmar, his creations are structurally disciplined enough to be called music, and "Unknown Pilgrim" has a pensive, haunting tunefulness that makes it the best of the lot. Unfortunately, it used a percussive sound something like a mix of marimba and organ, and, as with the other titles, I was left distracted and ultimately bored.

Apparently, I am going to have to be on something a lot stronger than light beer to enjoy new age music, "progressive" or otherwise. So now, it's back to Mozart.

From Scot Schneebeli of Sarasota, FL:
John M. Bennett, *The Blur* (Luna Bisonte Prods). Yeech! Well, okay, so I was expecting something a bit more musical. It

does carry a disclaimer saying it's avant-garde poetry performed with music. So with that in mind, I listened a second time. Yeech. I thought of polyester beatniks with plastic pocket protectors who had done way too much acid and their thoughts were left even more disjointed than usual. That combined with the "music," which was akin to a mass of MIDI equipment having a gestalt glitch, left everything to be desired.

From Brian Willoughby of Redmond, WA: "Still Fallin'," "Shine, Shine," and "Much Too Much," from **Ken Neagle** (Audio Lab Recording Studio, 2521 West La Palma Ave., Suite P, Anaheim, CA). Listening to this tape reminded me of a theory that I have: Given the right software and enough data, a powerful computer could generate all of today's pop music. Expensive MIDI equipment could be justified by the savings the record companies would see after humans stopped using their studios and wearing things out. What would pop music be without lyrics, you're saying? Well, the Commodore 64 has a speech synth that can sing, so just put it under MIDI control with a few words from the database, and produce commercial/jingle-type singing—sort of a wimpy Max Headroom. One caveat: Because all the computer data would come from existing music, you wouldn't hear any original ideas or unique approaches; but that's what's missing from this nine-minute tape, anyway.



Domino, Kevin

From Kevin Domino of Eden Prairie, MN: Artistic performances can be thought of as consisting of distinct elements. For instance, the visuals in a movie can be broken down to costuming, staging, lighting—whatever the producer needs to make the appropriate emotional impact. As artists we strive for a performance that is so seamless the individual elements are not noticeable in themselves; but if any of these parts are missing or of relatively substandard quality, the performance is lacking and the message is muddled. A reviewer's job is to point out the parts of a performance that are noticeable.

First, we have a four-song demo from the **Lost Boyz**. This tape will help them secure work in a club circuit. Mainstream rock; nothing unusual. They are a maturing rock band, close but not quite there. The tape exemplifies the common dilemma of no clear vision, nor a shared common idea; it needs one person in control of the final production to scrutinize those details interfering with the message. Some examples: What a good, in-control drummer! But the killer snare on the last tune is definitely needed in the others, especially the first. Some string parts don't follow the rules of good voicing and, occasionally, clumsily go to octaves or unisons with the vocals at strange times. The bass volume is virtually nonexistent. The lead guitar solos are lick- or riff-based rather than played with the continuity of the phrases being considered. These are minor points, though, that can be fixed. Good basic tunes, with good variety. High level of technical playing. With good management and a clearer focus, the Lost Boyz from Ft. Collins, CO, could be contenders.

The next tape is a project by **Hani Habashi**, *The Ugly Ones*. The note with it recommended listening to it entirely at one sitting "to grasp the concept" of his parody of Top 40 radio. One or two things caught my ear, such as the phrases "My hair wants to die," or "I like rubber." On "Water in the Hat," I was almost comfortable with the way it was developing, but it never got anywhere before ending. The other fourteen songs, of half-speed drum machine and overprocessed voices with incessant synth ostinati, were tough to listen to. I longed for anything to grab hold of emotionally or sonically. Maybe it's because I am reminded of tapes I did when younger, or maybe it's that this music never earned the trust of my ears.

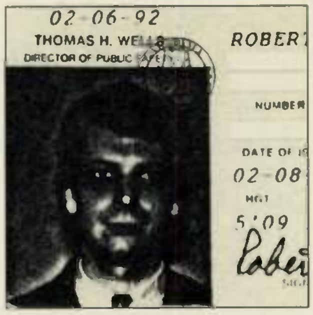
From Craig Mealer of Northport, AL:
Voyage to the Silicon Jungle (Composer/Performer: **Frank Sledge**, 2145 Oglesby Ave., Winter Park, FL 32789). One tip if you're making this trip: The voyage is short, and you get to the jungle right away. Jungle it is indeed—imagine the intrepid listener, bravely hacking his way through the dense underbrush of repetitive sync-bass passages and midrange slush—wondering, waiting, hoping—is there a point to this? Will I ever get out?

Okay, so it's not *all* bad. Some ideas are here, some feeling, it just never shows the full realization one would expect from "instrumental new wave jazz" or, as the composer says in his presskit, "Mind Movie Music." I don't know about you, but I prefer movies with a plot, a climax, and a resolution; even if they're only in my mind.

Music for Synthesizer Orchestra (Composer/Performer: **Glenn Meade**, 734 South Lyman, Oak Park, IL 60304). Mr. Meade has an impressive list of credentials—he graduated from excellent schools, has written numerous compositions, and has had many of those works performed. The presskit description of his production facilities gives the impression that this is a well-recorded tape with high-quality instrumentation.

I really wanted to review the *Music for Synthesizer Orchestra*, I really did. Unfortunately, I kept falling asleep during the opening of the first track and not awakening until the tape turned itself off. If you *really* like classical music, maybe you'll be able to listen to this straight through, without a nap.

From Phillip Schawillie of E. Rochester, NY: **Alexandros** (c/o Alexandros Audio-Visual Research Inc., 78 Fifth Ave. P.H., New York, NY 10011). Reviewed: *Antithesis* and "Next to Nothing," which is an excerpt from *Parallel Universe*. These cassettes represent Volumes I and IV of the twelve-part "Future Classics," a work in progress for the synthesizer. Alexandros intends these works to propel him to ranks of solo synth artists such as Vangelis, Kitaro, Tomita, and Brian Eno. Does his music justify this? Not entirely, but it shows potential.



Mealer, Robert

Antithesis, Alexandros's first full-length release, shows him emulating a wide range of styles, from the Kitaro-like beginning of "Life Synthesis" to the more classically influenced title track. Regardless of style, his strength is his counterpoint, whether it be melodic, textural, or timbral. "Antithesis," the most unique track on the album, has an interaction of tonalities reminiscent of Stockhausen meeting Wagner. "Hypothesis," the opening track, uses a more conventional approach, first stating the theme with solo piano, then adding a string countermelody, and, finally, replacing the strings with a bank of synthesizers. It, too, is an album highlight.

The biggest problem is when he deemphasizes the counterpoint in favor of straight-ahead themes and prominent rock percussion. In tracks like "Return to the Unknown," "Continuous Past," and "Vital Force," the drum machine is clearly in evidence. Moreover, the unimaginative patterns and weak production give the percussion the sound quality of a 1970s home electronic organ.

If the four-part "Next to Nothing" track on *Parallel Universe* is a valid indication, Alexandros has improved his approach to percussion. Instead of straight rock rhythms there are a variety of Latin and shuffle tempos, using Steve Thornton (known for his work with Miles Davis) for embellishment. Alexandros enhances the lush orchestral techniques of the prior album by incorporating some classic lead lines and chord progressions of jazz fusion. "The Logical Confusion/Necessarily Not" borrows from the Chick Corea/George Duke/Jan Hammer school of lead synth, and "All Beyond a Thought" is reminiscent of the Mahavishnu Orchestra. If Alexandros can maintain the quality of "Next to Nothing" in future releases, he may yet become the next synth superstar. Mean-

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while, the sheer magnitude and variety of his output is interesting and fun to watch.

From Kevin East of Herndon, VA:
Mark A. Vickness, *Markus* (Markus Productions, 1810 Euclid Ave. #18, Berkeley, CA 94709). Review 1 (long version): Berkeley's North Side is a vibrant heterogeny of shabby students, bemused merchants, and intriguing street types anchored by La Val's Pizzeria and Good Time Jazz club. Hovering over this hamlet of college town eclecticism is a lonely Mark Vickness, apparently ignorant of the pulsating rhythms that surround him. Vickness's debut tape, *Markus*, glides on a sheen of squeaky clean production. Synth tones ring clearly, HR-16 ride cymbals and tuned agogo bells glisten brightly, wooden flutes breathe a husky resonance, all underscored by a smooth and deliciously rich bass. This is slick, pro-quality sound here.

However, the music is better suited for, ah, Chevrolet ads? Imagine Wagner and McCartney collaborating on *Respiratory Complaints of the Humpbacked Whales.* Imagine Bambi and Thumper frolicking through a floral menagerie of color and light, then hunkering down over the Dow Jones. Imagine your worst fears about new age dominating the airwaves.

On several occasions, Vickness starts to explore some subtle rhythmic patterns, displaying a deft touch for matching bass and percussion. Both "City" and "Island" are introduced by uptempo beats, which are sadly smothered by gooey synth horn ensembles and a host of soloing vehicles. Sadder still, the solos rarely rise above the shameless noodling you and I smugly pass off as new age on our CZ-101s. Vickness claims classical training and inspiration, and indeed some of the melodic counterpoints indicate a flair for compositional drama. For the most part, however, the songs have a tendency to mercilessly crush any glimmer of liveliness under a mush of digital pap. It would seem that instead of overviewing the North Side from an apartment window, Mark might benefit from regular soirees into the streets.

Review 2 (short version): This is shameless, digital poo-poo.

Robert Carlberg *would like to thank those who participated in this experiment. Send music for review in the regular column to PO Box 16211, Seattle, WA 98116.*

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If You Can't Say Something Nice...

Getting criticized is never fun, but it's part of life. If you're going to go public, you have to be prepared for the worst as well as the best.

By Craig Anderton

I had some real misgivings about running Robert Carlberg's column this month; our amateur reviewers seemed like a pretty merciless bunch. Did running these reviews serve any *constructive* purpose?

Most people would agree that negative reviews are, like it or not, a part of life. Many musicians had mercilessly taken Robert to task for *not* reviewing their tapes, and we certainly get enough people slamming the magazine or its authors for one reason or another. Yet I don't like to perpetuate that vicious cycle of criticism. I'm not saying that negative criticism should be excised, but rather, that it be handled as diplomatically as possible.

Or should it? I've received some very negative reviews over the years. While painful at the time, those comments often helped make me a better musician not just by offering some honest criticisms, but by stiffening my resolve to make better music. And frankly, some of it gave me a bit of an "I'll show them..." attitude that kept me going when door after door was slammed in my face. Rejection is part of the real world, and the sooner we get acclimated to it, the better for our psyche.

Also consider that reviews can tell more about the reviewer than the record (a point made by Robert rather hilariously in the July 1989 issue). Artists assume that when readers read a negative review, the reviewer's word is taken as gospel. Not so—many readers conclude it's the reviewer who has a problem, not the artist, and let the *reviewer* have it with both barrels.

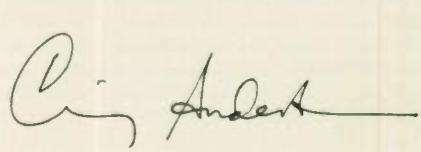
Interestingly, though, it seemed that many cheeseheads *wanted* to like the music, but couldn't—maybe a lot of the music being released to the public should stay private. Item: When I first got involved with the magazine, I was awed by the volume of tapes coming in for review. I thought I'd get to hear all kinds of music too innovative and inventive for the major labels to touch ...wrong. Much of it was "vanity press" music; sure, there were exceptions, but for every Michael Gilbert there were a dozen unlistenable tapes. When Tim Tully joined the magazine, he came to the same conclusions independently, as did Nancy Woodruff, our office manager. Maybe the phrase "If you don't have something nice to say, don't say it at all" also applies to tapes: If your music doesn't have something to say, maybe you're best off playing it for your friends and leaving it at that.

When I mentioned some of my reservations to Robert, he replied "I think the reader reviews were fascinating, but these guys got *much* tougher than I ever had the [guts] to try. What does this tell me? A) They agree with the need for a little constructive criticism. B) There's a lot of music out there that isn't wonderful. Will there be some hurt feelings resulting from the 'Voice of the Cheese' columns? Undoubtedly. Who's to blame? The reviewers? You? Me? The magazine? No—I think it's the artists. Anybody who sends out music that isn't fully developed deserves to be reviewed mercilessly. Anything less would be dishonest."

Although I wish people would be a bit more understanding (see the December 1989 "Back Page"), honesty *is* crucial. One reason I don't edit Robert is because he covers music by some people who are my friends; I won't use my position to influence the column's content. To censor the "cheeseheads" would be wrong, too. They wrote their reviews in good faith, so it's our obligation to follow through on our experiment, and let the chips fall where they may.

One chip has already fallen from Jeanne Zanussi, one of our esteemed copy editors. Her comments: "A bunch of arrogant, pretentious snobs get a chance to get into a magazine and then spout off about their philosophies. After reading, I couldn't tell you anything at all about any of the music on these tapes because no one really talked about it." Other staffers found the column entertaining and not to be taken so seriously.

I'm concerned about hurt feelings—I know from personal experience how difficult it is to see one's hard work dismissed in a cavalier fashion. I used to be emotionally devastated when I got bad reviews; now I grudgingly recognize it's just part of the deal. Yet the end result of bruised egos can be positive. Groups from the Beatles to David Bowie to Kiss were totally panned at one point, only to end up laughing all the way to the bank. If you're going to "go public," you'd better have a pretty thick skin. Welcome to the club, fellow cheeseheads and amateur musicians.



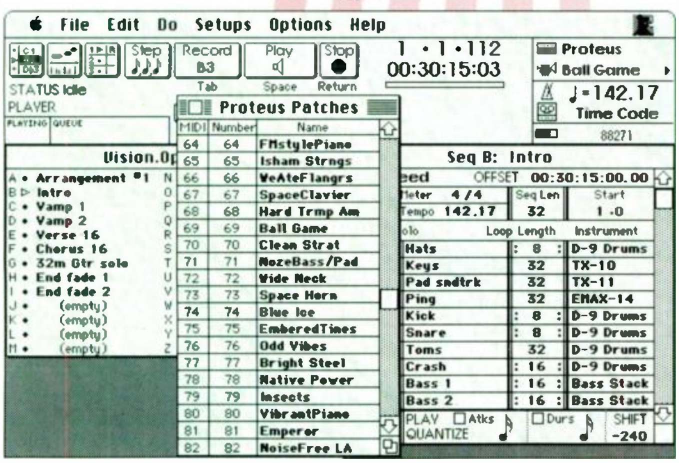
Two New Views

Introducing Two New Macintosh Music Software Achievements: Galaxy, Opcode's Universal Librarian, and Update 1.1 of Vision.

We've created a landmark connection between Vision and our Librarians. Here's how it works. Patch names from our new librarians are automatically transferred to Vision's instrument set-up. Then you can choose patches by name—not number. Choose the name, hear the sound. And when you change a patch name in the bank, the name changes in Vision too.

"The undisputed leader in Editor/Librarian technologies." Keyboard Magazine

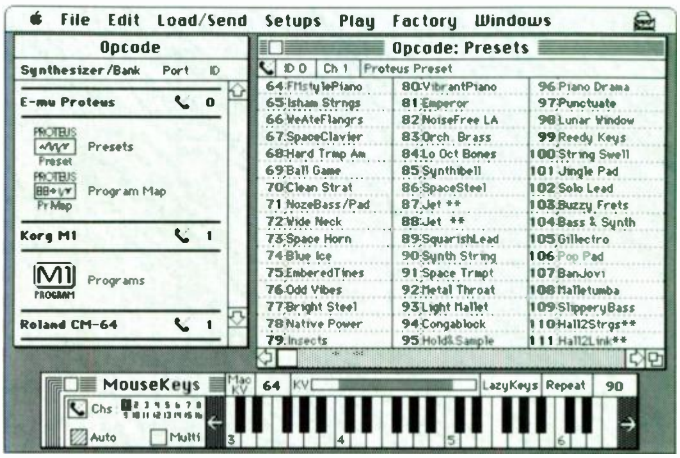
Galaxy™ stores data from any MIDI equipment with System Exclusive capabilities. We support over 70 MIDI devices. But you can easily create your own file type with our simple language called PatchTalk.™ And Galaxy is a full featured Opcode Librarian, with Patch Factory™ for random patch generation, and you can get and send single patches, banks, or Bundles. As always, we don't compromise.



New update 1.1 of Vision

"Vision is loaded." Keyboard "the best sequencer I've had a chance to use..." Mix

And the reviewers haven't even seen the 1.1 update. Editing windows scroll on playback. You can import note names from Opcode Editors or type them in; perfect for drum sounds or samples. Select or move a note in graphic or list editing—you hear the note. Edit MIDI parameters with the new pencil and exponential curve tools. Create MIDI mixes with automated, moving faders. Tap Tempo on record or playback, or sync Vision to live music with our updated Studio 3 SMPTE/MIDI interface!



New universal librarian: Galaxy

"Expect to be amazed." Craig Anderton, Electronic Musician

To see Vision and Galaxy, visit your Opcode dealer today. Vision is the winner of Keyboard Magazine's 1989 Readers' Poll for "Software Innovation of the Year."

Call us at (415) 369-8131 for a free brochure. Demo disks of Vision are $10.

Opcode systems, inc.

Opcode Systems, Inc.
3641 Haven Drive, Suite A
Menlo Park, CA 94025-1010

Vision™

Small Monitor. Big Difference.

In the recording business, little things can often make big differences. Studio monitors, highly sophisticated critical listening devices, are certainly no exception. Our Control Series™ compact personal monitoring systems each provide the performance characteristics demanded in today's recording environments.

Take our Control 5™ for example. You get power handling capacity of 175 watts, outstanding dynamic range, smooth frequency response and excellent clarity and imaging. This high power, low distortion system is housed in a non-resonant polypropylene structural foam enclosure.

Today you can find Control 1's in home studios and midi workstations; Control 5's in major recording and teleproduction facilities; Control 10's in foreground and background systems, corporate boardrooms and teleconferencing facilities. And the two-way horn loaded Control 12SR, a logical extension of the technology, in sound reinforcement applications from supper clubs and discotheques to small tour sound systems. Control Series meets such diverse applications because they are, above all else, powerfully honest.

Control Series. Compact high performance monitors designed to meet a broad range of fixed and mobile applications.

Versatility, the Other Advantage.

Designed to accommodate a wide variety of specialized mounting brackets, Control Series monitors can go virtually anywhere. On the console, on the wall, on the ceiling, in a rack, on a tripod, keyboard or mic stand. Control 10's and 12SR's even come with a built-in handle so they travel like a seasoned professional.

Next time you're looking for a super compact high performance loudspeaker system, remember Control Series then go see your JBL dealer. Look at the specs, then listen to the big difference.

JBL

JBL Professional
8500 Balboa Boulevard, Northridge, CA 91329 USA
H A Harman International Company