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Electronic Musician

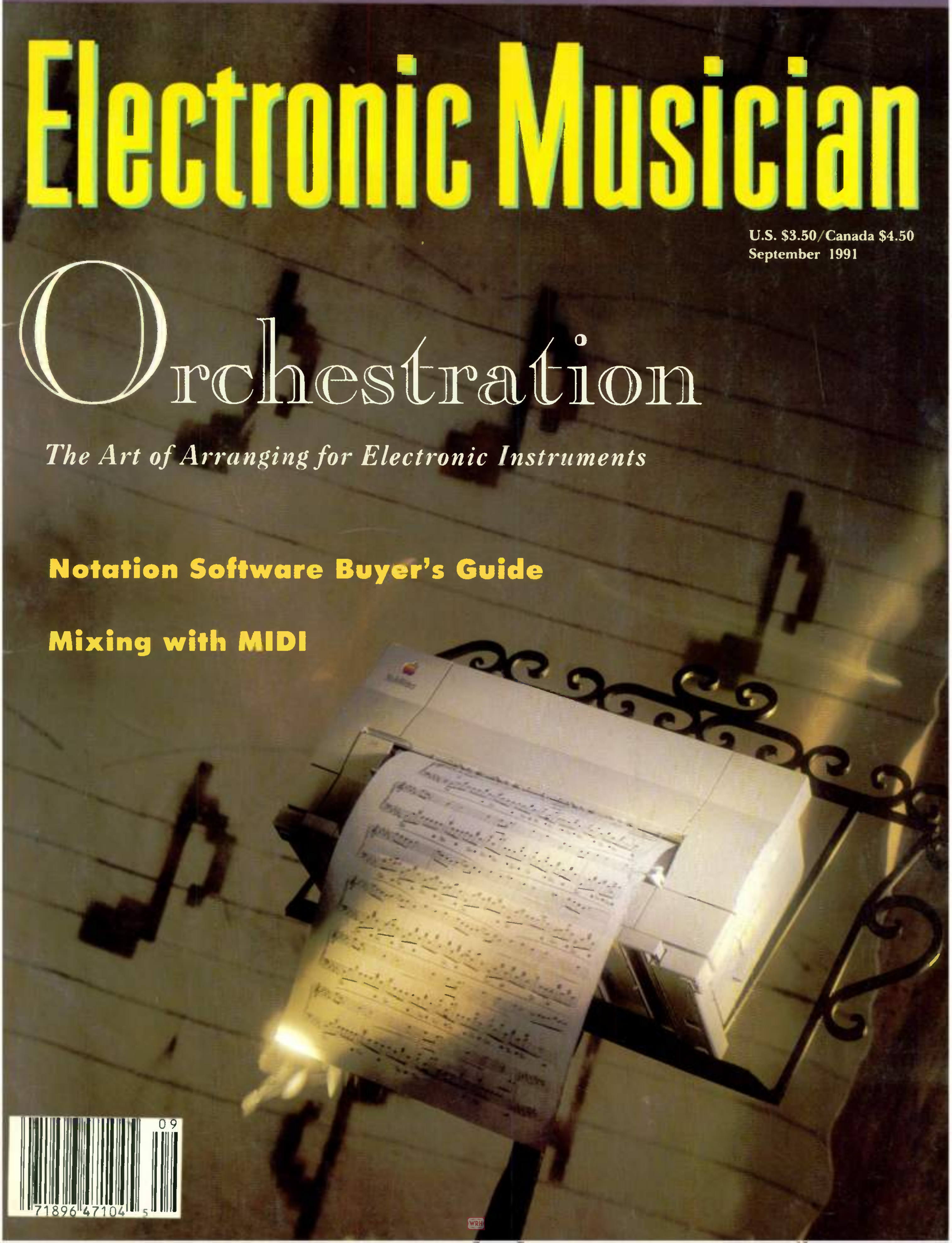
U.S. $3.50/Canada $4.50
September 1991

Orchestration

The Art of Arranging for Electronic Instruments

Notation Software Buyer's Guide

Mixing with MIDI



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TEC 1991 NOMINEE

MixerMixer ($299) combines all main outputs & AUX sends from each mixer — even headphone outputs. Three CR-1604s give you 48 line inputs, 18 mic inputs, 12 stereo returns, 24 channel patch/ direct outputs and 3 stereo sub-masters! Optional long-throw **Remote Fader** ($99) controls all mixer main outputs.

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Mono output. Summed L+R.

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Trim matches any signal, including instrument levels, −10 semi-pro and +4 pro gear.

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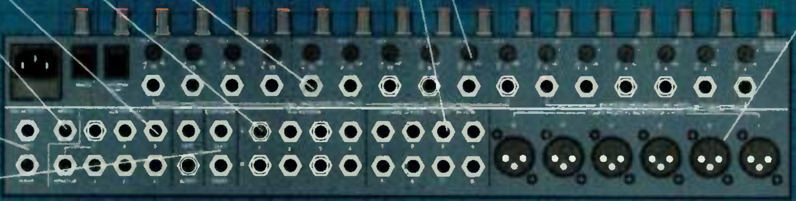
6 studio-grade, discrete, balanced transformerless mic preamps with astonishing specs & headroom: E.I.N = −129 dBm, 0.005% THD, +14dBv max input level. Plus...the $349 **XLR10 adds 10 more preamps** on chs. 7 to 16 (plus existing line inputs).



7 Sends. 4 knobs feed 7 separate outputs. Plenty of gain for special effects with center detent at unity gain. Use any 4 sends per ch.: Aux 1 switches to either pre (monitor) or post (effects). SHIFT changes AUX 3 & 4 to AUX 5 & 6.

3-band EQ. Musically useful frequencies: 80Hz (more real thump than 100), 2.5kHz (much better control than 1k), 12kHz (more sizzle than 10k).

Mute that's more than just a simple mute switch. Muted channels are UNassigned from the main left-right output and REassigned to outputs 3 & 4.

Stereo in-place solo maintains stereo perspective for ALL soloed channels & returns.

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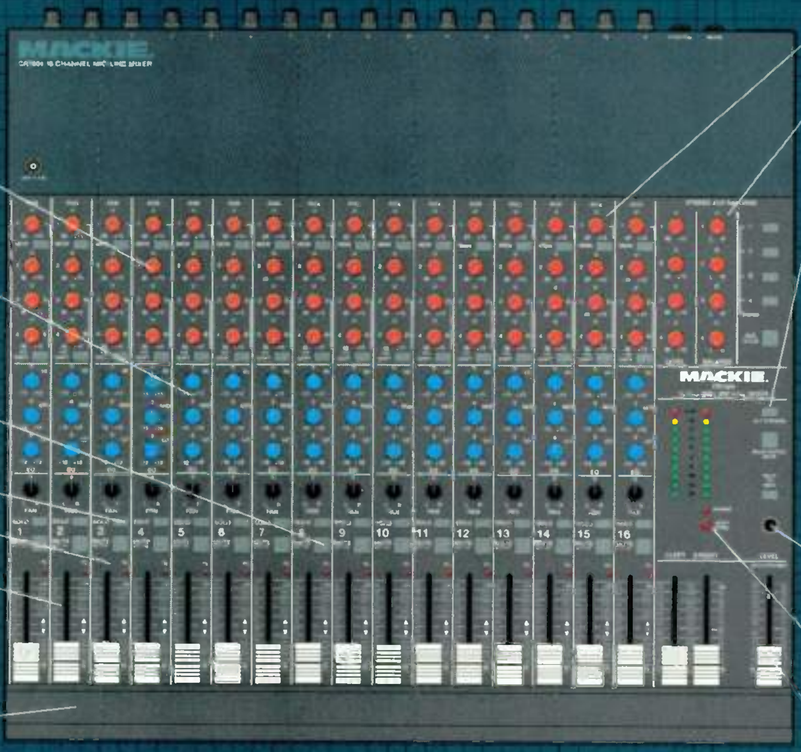
Inside: Less than **0.025%** THD 20-20kHz; **90dB** S/N ratio (ref +4dBu), super-quiet studio-grade mic preamps (E.I.N. —129dB @150-ohms)



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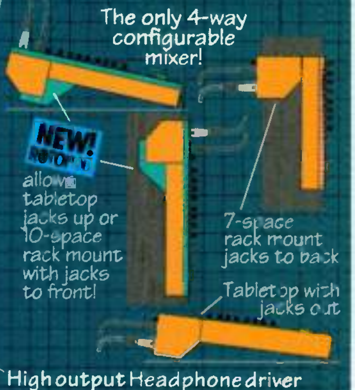

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Electronic Musician

AN ACT III PUBLICATION

SEPTEMBER 1991 VOL. 7, NO. 9

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Cover: Photo by Joshua Ets-Hokin. Special thanks to Apple Computer.

MIDI Reborn

After several years of dormancy, the MIDI specification springs back to life.

From its beginnings in 1983, MIDI was touted as an extensible specification. The original designers realized new applications would place increased demands upon the standard, so they allowed room to grow. Since then, we've seen several important additions, such as MIDI Time Code, the Sample Dump Standard, and Standard MIDI Files.

Now, a new round of extensions and proposals are being considered by the MIDI Manufacturer's Association (MMA) and the Japanese MIDI Standards Committee (JMSC), the two organizations overseeing the spec's development. General MIDI (discussed in last month's "MIDI for the Masses" article), Machine Control, and Show Control offer powerful new capabilities that could (and should) benefit all levels of MIDI users. In fact, part of the purpose of these new proposals is to increase the potential market for MIDI products to both the less sophisticated but much larger consumer arena, and to the state-of-the art world of recording and post-production studios.

At both extremes, the goal is the same: to make current products easier to use and more powerful. The proposals aim to increase functionality and make the technical details of system communication more invisible.

General MIDI is the simplest, yet most important addition to the spec in several years. Consisting primarily of standardized patch and drum-note assignment maps, its significance lies in its role as the final step towards the creation of foolproof, "plug-n-play," consumer-oriented MIDI systems. Other related additions, such as the Master Volume and Master Balance messages—which give global control over all multitimbral parts of a single instrument—make General MIDI even more powerful.

Current proposals for Machine Control and Show Control have more sophisticated goals. The hefty Machine Control proposal (initial drafts are as large as the original MIDI spec) seeks to incorporate into MIDI the machine-level transport controls found in synchronizers, as well as functions such as Track Select. This could standardize transport control of tape decks from sequencers, or even MIDI-equipped mixing consoles.

The main purpose of Show Control is to place theater-related equipment such as flash pots, lighting rigs, and hydraulic lifts under MIDI management. Current MIDI-operated lighting systems already use MIDI Note and Continuous Controller messages, but this proposal offers dedicated messages for theatrical applications.

All these extensions and proposals, as well as many others under consideration, point to an active specification promising many exciting new applications, products, and increased ease of use for all levels of MIDI users. More certainly can be done, even within the basic constraints of MIDI 1.0, but I applaud the two organizations' efforts to extend the spec. (For more information, contact the MMA at 5316 W. 57th St., Los Angeles, CA 90056; tel. [213] 649-6434.)

In many ways, MIDI is an "operating system" for electronic music and, as System 7 for the Mac and DOS 5 for the PC have shown, the immediate benefits of system-level change are important, but pale against the new foundation that is laid for future applications. I eagerly anticipate the day when everything in a music studio, including MIDI and digital audio signals and recording and video equipment, can be seamlessly connected and controlled from a single location. Until then, we're moving in the right direction.

Bob O'Donnell

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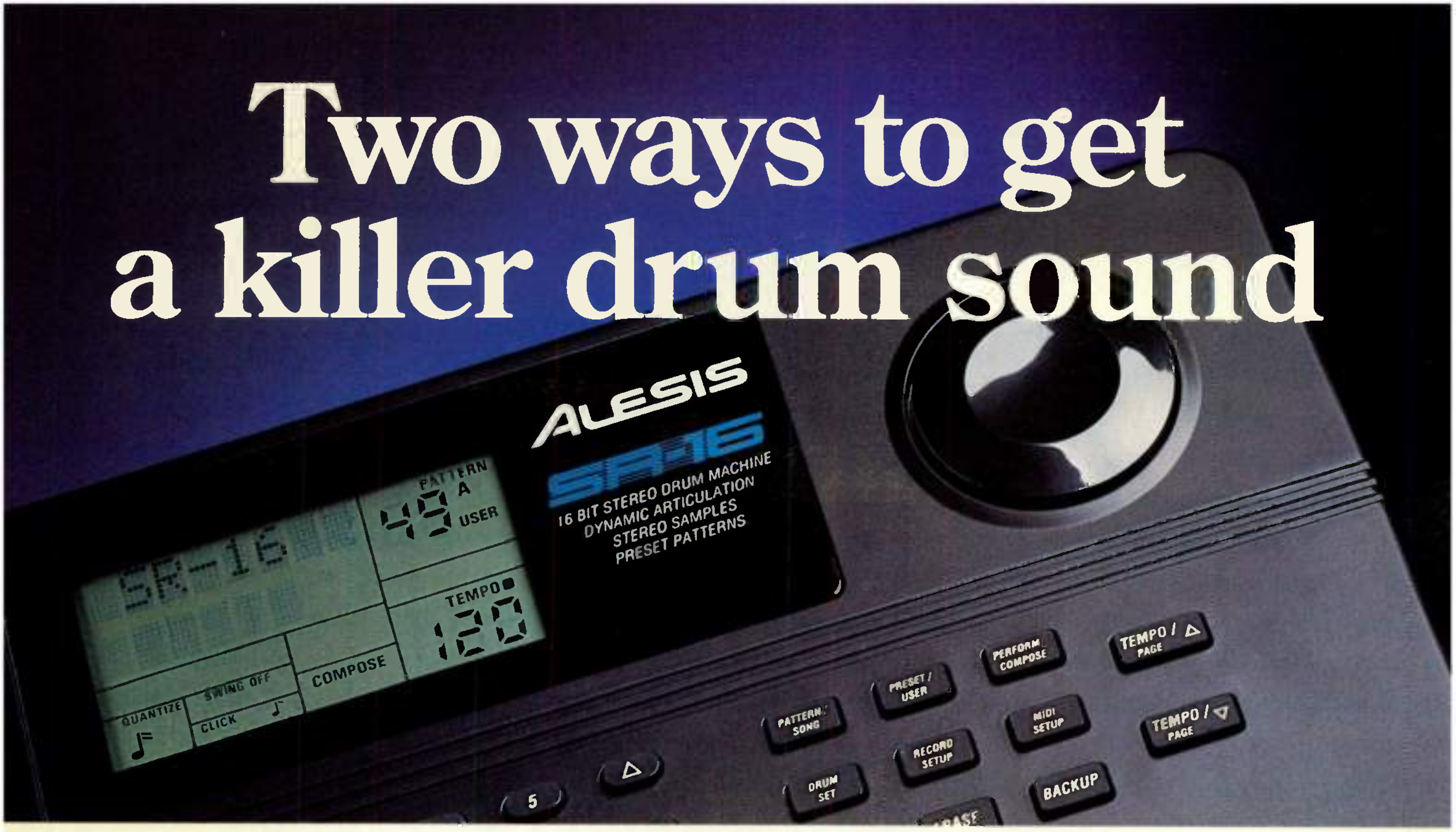
Also publishers of *Mix* magazine.

Consumer Audit BPA

Two ways to get a killer drum sound



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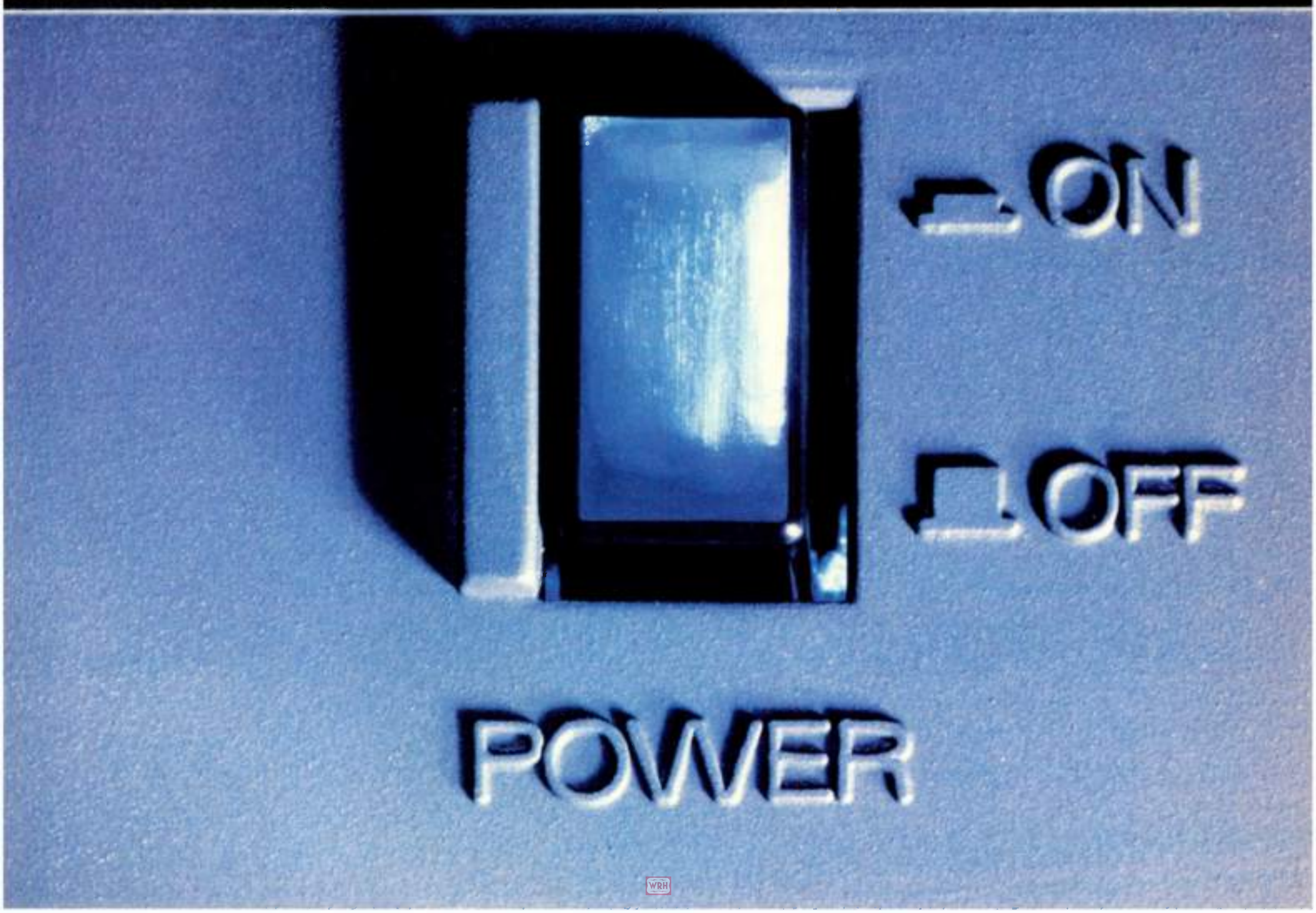


Studio M's Compu-mixer records MIDI volume levels and pan settings for internal and external parts.

If you'd like to produce professional quality music without first obtaining an advanced degree in rocket science, you've come to the right ad.

Introducing Studio M, a remarkable little powerhouse that integrates a sequencer, sound module and Compu-mixer into a versatile music production system. But just because Studio M is engineered to be incredibly simple to use doesn't mean it's simple-minded.

After all, in addition to its seamless integrated design and helpful displays, Studio M is 8-part multi-timbral and offers 30-voice polyphony. So you'll be able to play a lot of sounds at once. And you'll have lots of them to play with, since 220 sampled sounds are built in.

Call up the Timbre Edit display, and you'll have full control over envelopes, LFOs and multi-mode filters.

And whether we're talking about Studio M's great piano, drum and synth sounds—or any of its other sounds, for that matter—they're always easy to access. And even easier to edit. You'll also have built-in effects to add reverb, delay or chorus.

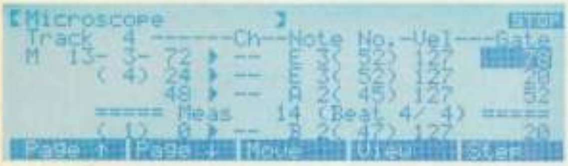
You can forget about MIDI channels with Studio M. To sequence: Pick a track, choose a sound and go.

Then there's Studio M's 16-track sequencer, which will allow you to choose the sequencing style—pattern, linear or loop—and then control up to 136 internal and external parts. When you combine this with Studio M's more than generous 50,000-note memory, we're obviously talking about major composing power.

When you're ready to mix, any adjustments you make using the 8-fader Compu-mixer are stored to provide a powerful auto mix-down function—as you monitor every fader movement on the LCD.

Each song will load along with its proper sounds.

Need more features? To sync to tape players with ease, Studio M offers Tape Sync II. For preserving your creations, the built-in disk drive stores up to 150,000 notes on a single disk. And for adding the newest sounds as your composing needs call for them, Studio M offers two ROM card slots for optional Roland sound cards.



Details, details, details: Studio M's microscope helps you make even the smallest changes to your music.

All of which makes our Studio M the ideal tool for music production at all levels. It serves as a stand-alone music studio or as the core of a larger MIDI system. Providing, of course, that you're up to speed on the function we've highlighted to the left.

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Our readers speak their minds on MIDI, rant at reviews, and offer excellent miking tips.

HUZZAHS

Just picked up the July 1991 EM and was greatly satisfied with the entire issue. I've never seen a magazine that covered all my interests as well as yours. Case in point is the desktop-audio-meets-desktop-video articles. I appreciate your covering these powerful new tools for the independent communicator. Huzzahs to your staff.

Andrew Schmit
Chicago, IL

SPECIFYING MIDI

I commend Peter Freeman and Bob Moog (June 1991 "Letters" and "Back Page," respectively) for expressing their views on the responsive inadequacies of today's electronic instruments.

As a computer systems engineer, I also believe the current MIDI specification defines enough resolution to represent realistic expressive capabilities.

The lack of quality real-time control is more evident in controllers than sound modules. It is a sin for a manufacturer to implement 7-bit Pitch Bend. The MIDI specification requires them to transmit the other byte anyway, so they're not gaining any throughput.

Some companies take pride in their work. These manufacturers should be applauded for their thorough MIDI implementations. They believe in us, the customers, to produce creative and truly moving music.

John Norvell
Tulsa, OK

MORE OR LESS SUBJECTIVE?

I thank Bob Moog for his illuminating "Back Page." The call for revisions to the existing standard—or a new one altogether—have lacked statistical support. Dr. Moog's contention is well-founded; the MIDI spec itself has enough resolution to do the job.

Rather than rehash the same arguments about the fabled MIDI 2.0, your magazine would do well to take Dr. Moog's ball and run with it. In your product reviews, tell us about the MIDI implementation. In past reviews, the only references to the MIDI response capability of a unit were subjective. While opinions are important, a combination of fact and opinion would be ideal. Let's see charts, graphs, tables, and details about where the unit excels and where it falls short. Then, tell us what it means to the performance of the product. If performance matters, this approach will help consumers make well-informed buying decisions and thus influence manufacturers to design better instruments.

Aaron Kneile
Des Moines, IA

We readers cannot afford to make incorrect purchase decisions. Your obligation is to take a critical view of all new product offerings, mercilessly pointing out their deficiencies no matter how many color advertising pages their manufacturers buy from you. If equipment makers believe in their products, they should have no problem with this approach.

We want bold, subjective product comparisons, not only tables of specifications. Many intangible characteristics separate a workhorse from a pain in the neck, and many of us skip to the final paragraphs of your reviews in search of the writer's subjective, emotional response to a product.

We do not appreciate loyal users of a maker's gear reviewing that gear. Their predisposition to like these products leads to puff-piece reviews. We don't appreciate techno-nerd writers who are more concerned about modulation routings and MIDI implementation than the musicality of an instrument. We want to know about noise, crosstalk among outputs, or any technical deficiency that will become clear when the product is used professionally.

It's your job to narrow the field to the worthiest candidates, so that our precious equipment dollars can be well spent. Have courage! Have opinions! Have a heart, would ya?

One Who Presumes to
Speak for Many
Hollywood, CA

Aaron and One Who Presumes—In reviewing any product, the goal is always to balance objective information with subjective commentary. We know many EM *readers are*

DAVID POVILAITIS

tech junkies, so we include as much relevant technical information as possible. At the same time, we incorporate subjective commentaries throughout the review to help give you a feel for what it's like to work with a given product. How much of each type of information is required to produce a balanced review differs from person to person, as your two letters demonstrate. With regard to how much space we devote to each article, we generally prefer to cover more products with smaller reviews, rather than produce the ultimate review on only two or three products a month.

One of the ways we have increased our product coverage is through buyer's guides. In a buyer's guide, we give an overview of a particular type of product and include a chart (like the one to which One refers) that lists all available products within the category and the features we determine to be most important to the typical user. Buyer's guides are not intended to be reviews, however. They are meant to provide you with the information you need to begin your decision-making process.—Bob O'D.

MIKING ALTERNATIVES

As an engineer and producer, I'd like to point out a few alternative perspectives on Jenny and Bruce Bartlett's article "Production Tips for Your Home Studio" in the May 1991 EM.

Omni mics: If leakage is not a major concern, an omni-directional mic is a better choice than a cardioid mic. For example, when placed at a semi-closed distance, an omni often picks up a greater portion of an instrument's natural body and produces a more natural, representative blend. Omni mics often eliminate "popping" (particularly when you use a condenser mic). In addition, leakage from cardioid mics often produces an "off-axis" coloration that can be even more offensive than an omni mic's more uniform off-axis frequency response. When following the basic three-to-one rule of close-miking technology (three units of distance between sound sources for every one unit of distance between the mic and its sound source), omnis produce outrageous results.

Acoustic guitar: Instead of placing a cardioid mic in front of an acoustic guitar's (bass resonator) hole, try placing a quality condenser omni or cardioid about eight to twelve inches in front of, and slightly below, the instrument, facing upwards at a point between the hole and the base of the neck. A full-range condenser really pulls through.

Sax: Close-miking a brass or reed instrument's bell produces an uneven instrument response. Upper sounds of most keypad instruments radiate from the open holes, while lower notes just blast out through the bell. Pulling back a bit, or placing the mic uniformly between the keys and the bell, generally results in a full, uniform pickup.

Mic-technique basics: The Bartletts advise readers to "damp your recording room with fiberglass insulation at least six inches thick, covered with muslin or burlap." This hails to the recording era of the 1970s. I call this the "sound sucker" era, whereby all room reflections are absorbed, leaving reverberation strictly to effects devices.

Modern recording studios and properly designed home recording environments may favor a slightly more reflective acoustic environment that combines both absorbtive and diffuse-reflected sound. Such a room often yields a sound that is slightly more live and in keeping with acoustic environments of the 1990s.

David Miles Huber
Seattle, WA

Bruce Bartlett responds: I edited and contributed to David Huber's excellent book, The Microphone Manual, *so he's giving me some of my own advice.*

Omni mics have the benefits mentioned and generally give the best performance for the price. Compared to omnis, however, cardioid mics reject background noises and room reverb, which are real problems in home studios. The article offered tips for home studios, many of which have inadequate soundproofing or acoustic treatment.

On miking acoustic guitars, I did not recommend placing a cardioid mic in front of the sound hole, but eighteen inches away, where the boomy effect doesn't exist. Huber's suggestion—miking closer, below, and partly toward the neck—works well, too.

I also didn't recommend miking the sax at the bell, but about eighteen inches away, where the sound is more natural.

Regarding room damping, I suggested starting with a six-foot-square patch of damping material. The room still will be pretty live. You can add more damping, a little at a time, until your recordings sound as dead or live as you wish. I've made many recordings with close-miking in rooms without any damping. The recordings sounded great, and the musicians enjoyed playing in the live environment.

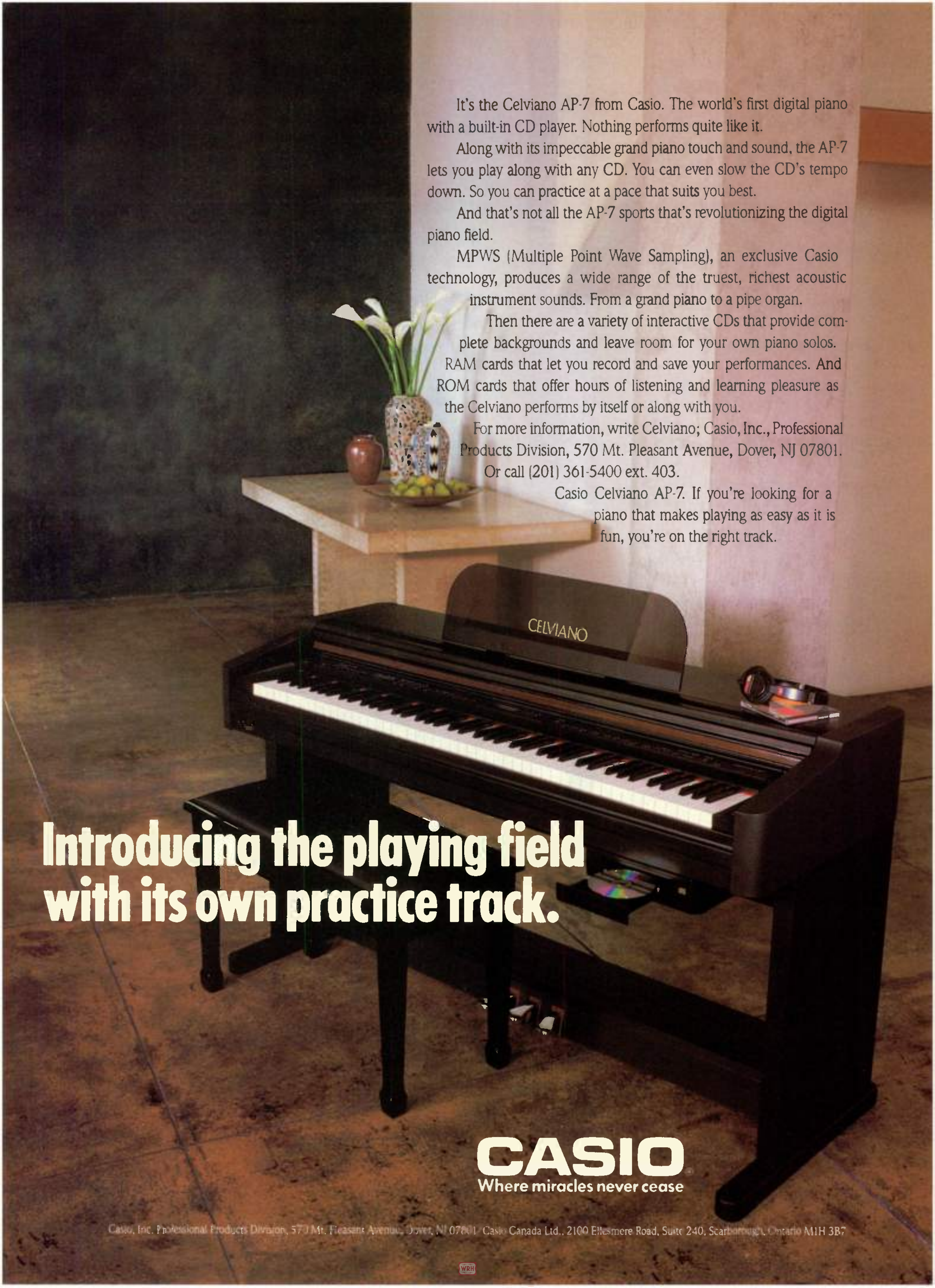
I'd like to correct an error that crept in during publication. The article said, "Generally, you should mike more distantly for recording than for sound reinforcement so that each mic picks up more room acoustics and background noises." "Background noises" should read "a more natural timbre."

Also, Figure 2 showed a microphone that was angled down at the sound hole several inches away, labeled "bassy." This mic actually should have been drawn close to the sound hole, where the sound really is bassy.

As Huber will agree, miking is an art as well as a science; there is no one right way to mic anything.

ERROR LOG

August 1991, "MIDI For the Masses," p.26. In the editing process, a sentence was added that said the JMSC had approved General MIDI. In fact, the JMSC is still considering the proposal.
August 1991, "Audio-Technica Headset Mics," p. 104: The opening photo shows the ATM71 mic, not the Pro 8.
July 1991, "What's New," p. 22: The address for the Metsan Corp. is PO Box 681272, Schaumburg, IL 60168.
July 1991, "NewTek Video Toaster," p. 99: The professional video system shown in Fig. 5 was assembled by the EM staff, based upon information provided by NewTek, and replaced a similar diagram provided by the author. According to the author, the prices listed are too low for professional production gear.
July 1991, "Opcode Cue 3.0," p. 106: The correct address for Opcode Systems is 3641 Haven Dr., Menlo Park, CA 94025-1010. ●



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WRH

WHAT'S NEW

Our Labor Day parade features multitrack hard disk recording for the Mac, a harmony machine for vocalists, and a MIDI-automated audio mixer.

DigiTech's The Vocalist

DIGITAL RECORDERS

Digidesign is shipping **Pro Tools** ($6,000), its long-awaited, integrated, multitrack hard disk recorder/MIDI sequencer/automated audio mixer/DSP system for the Macintosh. The base system provides four channels of recording and editing and includes *Pro Deck* recording and mixing software, *Pro Edit* timeline-based graphic editing software, a 1-U rack-mount audio interface, and a NuBus DSP card. The software supports unlimited non-linear "virtual" tracks (digital audio or MIDI tracks), non-destructive graphic editing, crossfades between edits, slipping of tracks and regions, and continuous SMPTE synchronization and resolve on all channels.

MIDI recording features include multitrack linear and loop recording, punch-in/out, pattern-based editing, and event editing and filtering. Real-time automated mixing (with optional external fader control) includes snapshot and dynamic automation, instant update, and event-level editing. Signal-processing modules include multiband parametric EQ and real-time digital effects such as delay and chorusing.

The audio interface provides four channels of balanced, +4 dBm analog I/O and two channels of AES/EBU or SP/DIF digital I/O. The system supports sampling rates of 44.1 and 48 kHz.

System expansion up to sixteen tracks (in increments of four) is expected to be available in the fourth quarter of 1991. Expansion also will require Digidesign's System Accelerator card (price and details to be announced) to provide a direct path from the audio cards to the hard disk drives.

Digidesign
1360 Willow Rd. #101
Menlo Park, CA 94025
tel. (415) 688-0600

SIGNAL PROCESSORS

DigiTech is shipping **The Vocalist** ($849.95), a harmony processor designed for the human voice. The unit provides 5-part diatonic harmonies that are said to be free of "chipmunk" high notes. In addition to manual and MIDI-controlled pitch correction, The Vocalist offers a vocoder mode; a built-in, 1-octave keyboard for selecting harmonies; an onboard synth that provides a cue-in tone and aids in harmony editing; programmable volume on each voice; and 128 factory presets and 128 user-definable programs, each with four variations. Song mode lets you program key and chord changes and step through them in real time via a footswitch or The Vocalist's control panel. Programmable portamento adjusts each voice's up and down slide between notes, and the programmable vibrato algorithm offers speed, depth, and attack controls. The unit has extensive manual and real-time MIDI control, de-essing, and programmable (±) tune on each voice.

DigiTech
5639 Riley Lane
Salt Lake City, UT 84107
tel. (801) 268-8400

Roland's **RSP-550 Stereo Signal Processor** ($1,295) provides true stereo processing and a wide variety of effects, including reverb, multitap delay, enhancer, EQ, flanging, phase-shifting, three types of chorusing, multiband pitch-shifting, rotating speaker effects, and vocoding. The unit features 39 effects algorithms and 160 user memory locations and supports up to five simultaneous effects. Reverb algorithms include hall, room, plate, and normal and reverse gated reverb. Delays range from single-line to a multitapped delay that features up to 8 taps, each with a maximum delay of 2,700 ms. A Tempo Delay function syncs to MIDI Clock, or you can tap in the delay time with a footswitch. In addition to Chorus, Stereo Chorus, and Space Chorus algorithms, you get chorus and pitch-shifting effects that let you assign separate chorus and pitch-shifting parameters to individual frequency bands. All four pitch shifters in the Quad Pitch Shift algorithm can be controlled via MIDI. The vocoder features a hold mode that lets you speak into a mic and hold the character of the sound without speaking into the mic again. The Rotary effect includes independent high/low horn speed, high/low rotor speed, and horn and rotor rise/fall time. The patch parameters can be changed in real time via MIDI Continuous Controllers, and the unit supports Program Change commands and SysEx bulk dumps. Independent, 16-bit A/D and D/A converters are provided for each channel, with a sampling rate of 48 kHz. The manufacturer claims a dynamic range of 95 dB, frequency response of 10 Hz to 21 kHz, and THD ≤0.02%.

Roland Corporation
7200 Dominion Circle
Los Angeles, CA 90040-3647
tel. (213) 685-5141

The Zoom **9030 Multi-instrument Processor** ($749) provides up to six simultaneous effects, combining digital effects with analog compression and distortion. The 16-bit effects unit includes more than 50 preset programs and 99 user-programmable memory locations. The half-rackspace 9030 has MIDI In/Out and remote footswitch jacks and provides mono input and stereo outputs. Zoom also released the **8050 Multi-function Foot Controller** ($249), a pedal board that features five Patch pedals and two Bank pedals, Control and Bypass pedals, and a MIDI Out jack. The 8050 has a custom Zoom remote port for control of Zoom 9010 and 9030 multi-effects processors.

Zoom
385 Oyster Point Blvd., Suite 7
South San Francisco, CA 94080
tel. (415) 873-5885

SOFTWARE

Recording Studio Professional ($195.95) offers sound recording and editing for IBM PC-compatibles using the Creative Labs Sound Blaster sample-playback card. Recorded sounds can be manipulated with cut-and-paste editing, digital parametric EQ (with assignable center frequency, bandwidth, and shelving), time compression/expansion, and pitch shifting. The software requires a graphic display card, a mouse, a hard disk, 640 KB of RAM, and the Sound Blaster.

Turtle Beach Systems
PO Box 5074
York, PA 17405
tel. (717) 843-6916

MIDI AUTOMATION

J.L. Cooper introduced **MixMaster** ($499.95), an 8×1/dual 4×2, MIDI-automated line mixer/submixer. A 1U rack-mount unit, MixMaster has eight ¼-inch, unbalanced VCA inputs and outputs, a mix output, and a stereo mix input. Setups of control parameters are saved in non-volatile memory, using a Learn button. The audio circuitry and dbx 2150 VCAs are the same as in

Zoom 9030 Effects Processor and 8050 Foot Controller

the company's MAGI II console automation system. The system can be controlled from a sequencer, MIDI fader unit, or other MIDI control source.

J.L. Cooper Electronics
13478 Beach Ave.
Marina del Rey, CA 90292
tel. (213) 306-4131

ACCESSORIES

Brüel & Kjaer's **Acoustic Pressure Equalizer** ($125) is an attachment for B&K's omnidirectional 4003 and 4006 mics. The APE functions as a passive acoustic spectral and directional equalizer. It uses diffraction to modify the soundfield near the mic diaphragm, which changes the frequency and polar response of the soundfield. This lets you create new mic characteristics with different spectral and spatial responses and increases directivity at higher frequencies.

Brüel & Kjaer Instruments
185 Forest St.
Marlborough, MA 01752
tel. (508) 481-7000

Brüel & Kjaer Acoustic Pressure Equalizer

Mackie Designs has released its first peripherals for the Mackie CR-1604 mixer. The **XLR-10** ($349) mounts directly to the CR-1604's bottom and adds ten discrete, phantom-powered mic preamps, giving the mixer a total of sixteen mic inputs. The **MixerMixer** ($299) is an active combiner that lets you combine three CR-1604s without using up inputs. Mackie's **Remote Fader** ($99) offers long-throw (100 mm) master fader for the MixerMixer, controlling the master outputs for all attached CR-1604s. The individual mixers' L/R main-output faders become submasters. The **RotoPod** bracket ($39) lets you rotate the CR-1604's removable connector pod so that the jack panel is on the same plane with the main controls. Mackie also offers a 39-cable **Cordpack** ($69) for connecting up to three CR-1604s with a MixerMixer.

Mackie Designs
16130 Woodinville-Redmond Rd., N.E., #2
Woodinville, WA 98072
tel. (800) 258-6883
or (206) 487-4333

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RACKS

SKB is shipping lightweight, **ATA-type effects racks** (ranging $139.95/2U rack to $239.95/12U rack) with flush-mounted rack rails so front panels aren't obscured. The front and rear lids are deep, protecting panel knobs and are attached with gasketed, stainless steel twist latches to prevent moisture and dust from entering. To prevent rail-stripping, equipment mounts on the rails with military aircraft A/N-type, case-hardened nuts and bolts. The racks are predrilled for mounting internal shelves. Racks are available in even sizes from two to twelve rack units.

SKB
3906 Sandshell Dr.
Fort Worth, TX 76137
tel. (817) 847-5400

AMPLIFIERS

Primarily aimed at guitarists, the **Elan Peacemaker X-175** tube power amp comes in a mono version that has a handle mounted to the top cover and in mono and stereo, 4U rack-mount versions (stereo $995; mono $549). The stereo version is a pair of independent, mono, 150W power amps that share a common faceplate. Each mono amp has two 6550 and two 12AX7 tubes; power supply; fan; volume, color, and presence controls; standby switch; and lighted power switch. The manufacturer claims frequency response of 20 Hz to 22 kHz (±0.5 dB) and 2.0% THD.

Metalhead Electronics
5707 Cahuenga Blvd.
North Hollywood, CA 91601
tel. (800) 966-3526
or (818) 980-1975

SOUNDS

Patchman Music announced two volumes of **Yamaha SY77 patches**. Volume I ($25 plus $1 s/h) contains 64 mostly single-Element patches designed to permit maximum polyphony. Volume II ($29.95 plus $1 s/h) features 64 patches designed for use with a MIDI wind controller or Yamaha Breath Controller. The latter volume includes assorted acoustic simulations, synth leads, and jazz section layers.

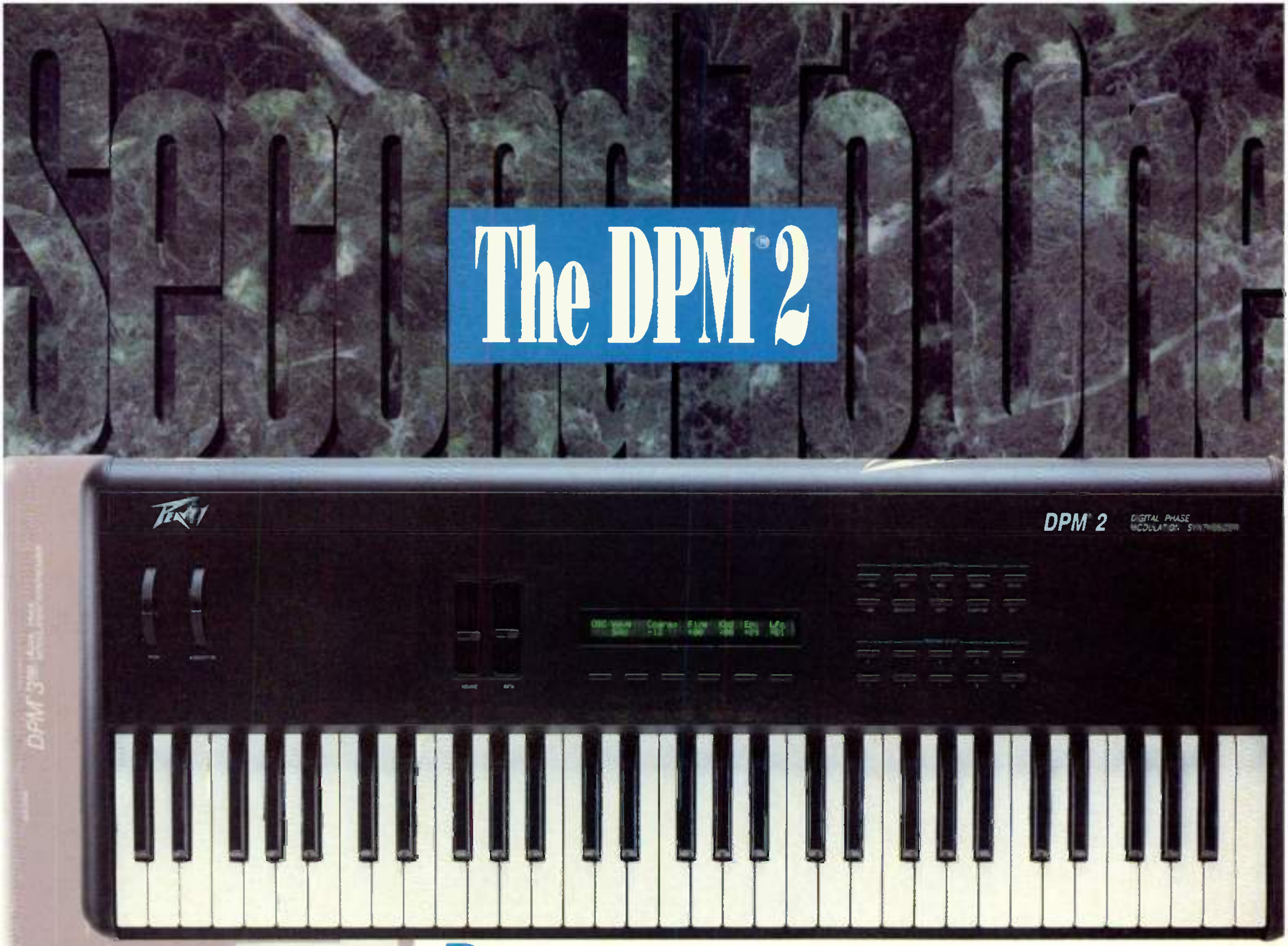
Patchman Music
2043 Mars Ave.
Lakewood, OH 44107
tel. (216) 221-8887

Electro Acoustics released a 10-disk set of **analog synthesizer samples for Ensoniq EPS and EPS 16 Plus** samplers ($89.95/set). The source synths, including an ARP 2600, are patched to produce non-imitative sounds of an avant-garde or "new music" character. All Instruments are programmed in stereo and implement velocity, pressure, and mod wheel. Each set includes 24 different waveforms in 48 layers and sixteen instrument files. A disk will load into an unexpanded EPS.

MacBeat (distributor)
1314 Rufina Circle, Suite A4
Santa Fe, NM 87501
tel. (800) 622-2328
or (505) 473-4929

REV UP

T.C. Electronic (tel. [805] 373-1828) has re-released its **Stereo Chorus/Flanger** effects pedal ($374). The SCF pedal offers chorusing, flanging, and



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385 Oyster Point Boulevard, Suite 7, South San Francisco, CA 94080

pitch modulation and has a built-in preamp, onboard power supply, and heavy-duty, die-cast housing. The manufacturer claims 100 dB dynamic range and 20 Hz to 20 kHz frequency response. Phase inversion of the processed signal between the left and right outputs results in a widened soundstage...Dr. T's (tel. [617] 455-1454) released ***Beyond 2.0*** ($319; upgrades $40) for the Macintosh. New features include support for Apple's *MIDI Manager* and Mark of the Unicorn's MIDI Time Piece; independent/multiple-track looping; scrolling event list with real-time editing; random velocity events; a scale editor that lets you create and save your own scales; a SysEx editing window; "fit time"; and more...True Image (tel. [619] 480-8961) announced version 2.0 of ***MacSpeakerz*** ($249), a loudspeaker design application for the Macintosh that uses the mathematical models of Thiele and Small to analyze vented and closed loudspeaker systems. Version 2.0 includes an array of interactive loudspeaker calculators, each of which is dedicated to a particular aspect of speaker design. Calculator types include rectangular and trapezoidal box; first-, second-, and third-order Butterworth crossover; first-order series crossover with adjustable damping; resonance compensator; inductance compensator; and tweeter attenuator. The expanded user's guide includes a chapter on crossover design.

T.C. Electronic Stereo Chorus/Flanger

Data Compression and the New Audio Formats

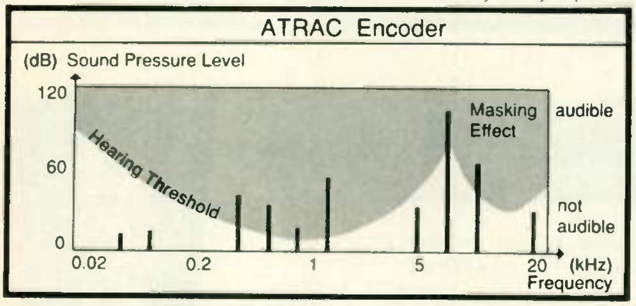
Consumer sound puts the Big Squeeze on audio data.

By Gary Hall

This seems to be the Year of Data Compression for audio and video. New formats for consumer audio are putting the focus on technologies that can squeeze large amounts of data into smaller storage space.

In the audio world, recent announcements of two new consumer formats for digital audio have made the subject particularly relevant. Phillips of Holland plans to replace the venerable cassette with a digital cousin known as "DCC" (Digital Compact Cassette). Sony, meanwhile, is throwing its weight behind something it calls MD, for "Mini Disc."

Courtesy of Sony Corporation



ATRAC data compression analyzes an audio signal into its component parts, then eliminates frequency components that are below the threshold of hearing.

Though very different, both formats depend on audio data compression. Without it, neither system would work. What is audio data compression? Why is it important? How does it work? How *well* does it work?

So-called "CD-quality" audio is sampled 44,100 times per second (in stereo), with a resolution of sixteen bits. This works out to a data rate of 172 kilobytes per second, or 10 megabytes per minute. High data-density is the biggest obstacle to designing digital audio systems that are smaller, cheaper, and more robust than the current CD and R-DAT formats.

For the Mini Disc, Sony chose a compression technique known as Adaptive Transform Acoustic Coding (ATRAC). This has been around for some time but only recently became practical for real-time use. Sony's proposal to use it for low-cost consumer products is a vivid statement of progress in DSP.

ATRAC is an analysis-resynthesis technique. An ATRAC processor gathers digitized audio into blocks of 1,024 samples, corresponding to about 20 milliseconds of sound. It then performs a spectral analysis on each block, yielding a description of the frequency and level of each component.

The next stage depends heavily on research into human hearing. Our ears are sensitive, but not infinitely so. To be heard, a sound must exceed a certain level, known as the *threshold of hearing*. The threshold is different in different frequency ranges. A strong frequency component also can *mask* softer sounds that are close in frequency.

An ATRAC processor determines the threshold that applies to each frequency component and compares it to the actual level. Components below the threshold are dropped, reducing audio data by a factor of about five.

During playback, the processor sums together sine waves at the frequencies and levels specified in the analysis and outputs sound at the standard 16-bit resolution and 44.1 kHz sample rate.

The benefits for consumer audio have been pointed out, but the implications for electronic music are no less exciting. For one thing, samplers and wavetable synths could use data compression to multiply their sound storage.

But the most interesting part is the

implications of analysis and resynthesis. With a sound broken into its individual frequency components, huge possibilities exist for creative processing. An analysis-resynthesis-based sampler potentially could transpose a sample without changing length or avoid munchkinization by altering the balance of components to preserve the resonances of an instrument or voice.

High data-density is the biggest obstacle to digital audio systems that are smaller and more robust.

How well does ATRAC work? It's been said there's no free lunch, and that applies to data compression. The sound quality of ATRAC-encoded recordings impressed critics in early tests, but it's obvious the process removes information from the signal.

In recording and sound-production it is common to process a signal in various ways, such as equalization. These processes can bring out sound components that may have been masked in the original. Processing of ATRAC-encoded sound could reveal the absence of expected features or distortions introduced in the compression process. Dubbing of ATRAC'd material also could degrade sound information.

It also appears to me that ATRAC coding will distort time information. The 20-millisecond blocks imply that the sound can change only 50 times a second. However, transients and stereo imaging information happen in much shorter intervals. I wonder whether ATRAC-coded recordings will be as pleasing to listeners once they have lived with them for a while.

Questions aside, audio data-compression will soon be a fact of life for professionals and consumers. It's time to start thinking about the implications for our music. ⊕

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Electronic Orchestration

Bring your music to life by applying these essential musical principles.

By René Salm

A well-arranged composition is music to the ears. Average melodies can sound extraordinary when an optimum mix of musical elements is achieved. The sad fact is good melodies seldom survive poor arrangements.

Navigating sonic landscapes is particularly dangerous for electronic musicians, due to the overwhelming number of options available. A collection of multitimbral synthesizers with built-in effects combined with a powerful software sequencer is an arrangement disaster waiting to happen.

The challenge for today's composers is to organize, manage, and control a wealth of resources. The best way to meet this challenge is to apply old-fashioned concepts of orchestration and arrangement. However, electronic music redefines the sound of the instrument and its projection into an acoustic space. Recognizing the impact of electronic instruments on basic orchestration principles is the first step to creating better compositions and arrangements.

WHAT'S ORCHESTRATION?

My musical dictionary defines orchestration as "the art of employing instruments in various combinations." This definition is quite clear regarding acoustic ensembles, but the nature of electronic music makes it hard to define an electronic "instrument." A *patch* is the obvious answer, but it doesn't take into account expressive manipulation of an electronic sound. This requires some type of controller, such as a keyboard or percussion pad.

An electronic instrument therefore must be defined as a patch *plus* the performance values of a particular controller. This fact enlarges the parameters of electronic instruments, because an identical sound played on diverse controllers becomes, in effect, a new

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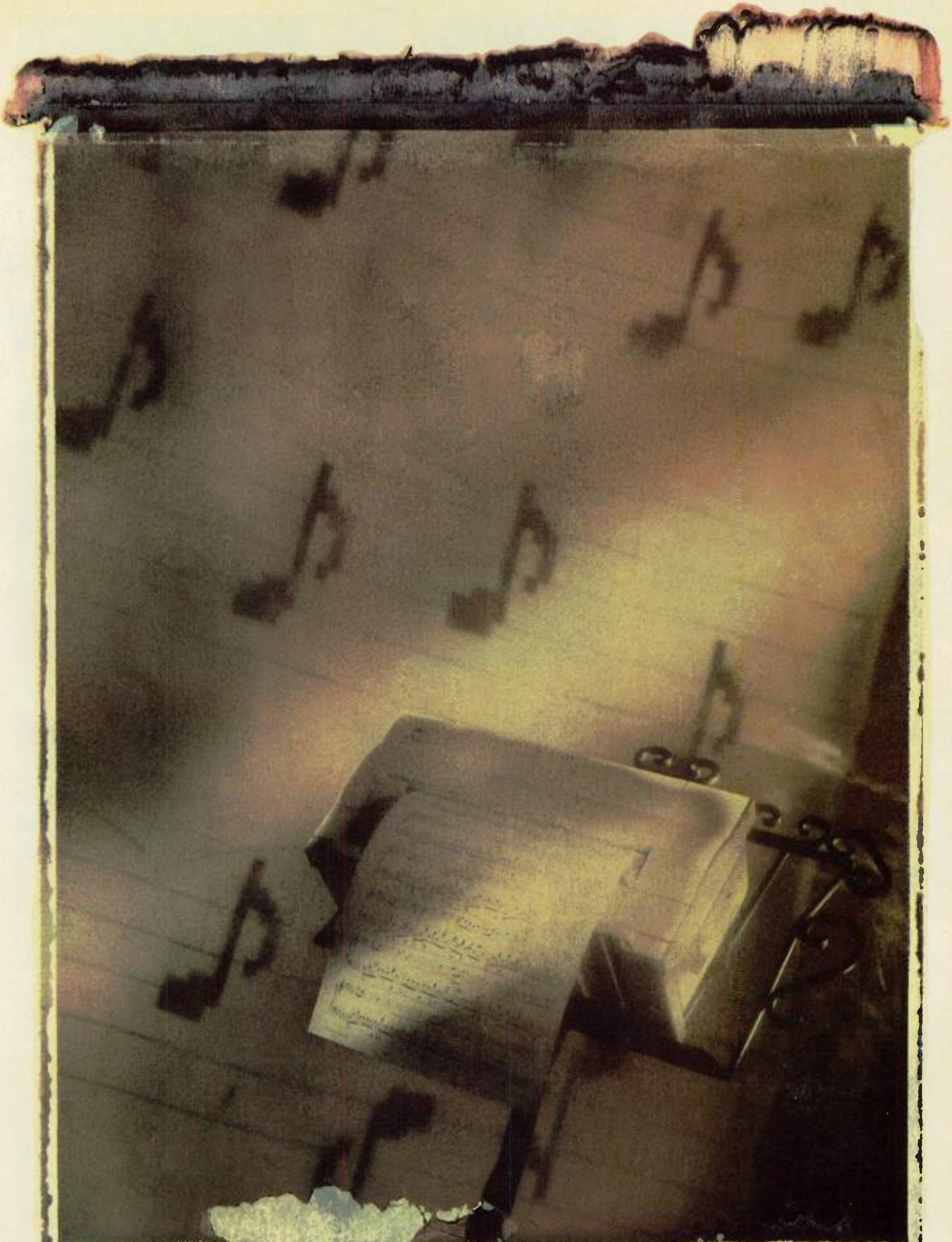

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YAMAHA

sound. For example, a grand piano patch performed on a keyboard controller undertakes a completely different sonic entity when performed on a wind controller. The orchestrator must account for this idiosyncracy of the electronic instrument when choosing tone colors for a particular arrangement.

P-A-T ORCHESTRATION

The three basic elements distinguishing both acoustic and electronic sounds are pitch, amplitude, and timbre. Understanding how these three elements affect orchestration is essential for creating well-arranged music.

PITCH

One of the underlying principles of orchestration is that sounds compete with each other within the audio spectrum. Usually, if a trumpet and flute sound the same pitch, the flute is inaudible. But if the trumpet drops an octave or two, the flute emerges. Adding to the problem of competing instruments is a limited audio spectrum. For example, an 88-key piano spreads from 27 Hz to approximately 4,096 Hz (four octaves above middle C). Even the average 60-year-old can hear up to 8,000 Hz, and a young person's range peaks around 16,000 Hz.

In addition, most musically useful sounds have fairly complex overtones which, in combination with other sounds, diminishes clarity. When a note is played on the piano, one pitch is perceived, but that pitch's many overtones give the sound its timbre. (For more on the relationship between overtones and timbre, see "Harmonics: The Basics of Sound" and "On Sound" in the February 1990 **EM**.)

It's easy to overload regions of the audio spectrum if pitches and patches are indiscriminately piled upon each other. This temptation increases with the growing polyphonic and multitimbral capabilities of today's instruments.

Orchestrators must assign sounds to certain areas of the audio spectrum where they can maintain proper dispersion and clarity. To accomplish this, keep in mind the three basic regions of the spectrum: treble, mid-range, and bass. An axiom of orchestration derived from this principle is: A full and balanced effect requires all three regions of the audio

MIDI CONTROLLERS

As part of the language of MIDI, electronic orchestrators have a number of controllers at their disposal. It is important to understand their relationship to acoustical counterparts. Topping the list are Velocity, Pressure, and Volume. Synth programmers often assign overall volume (not to be confused with MIDI controller 7) as an attribute of velocity. In other words, the speed at which a key is pressed determines how loud it is. However, in some cases it is more effective to use pressure to control the dynamics of a passage. Velocity works better to adjust a sound's attack. Many synths and samplers allow you to create this modulation routing. You can also experiment with velocity switching. For example, if you have two synths, one with a good slow brass and another with a good attack brass, you can assign them to the same MIDI channel and adjust the velocity cutoff of each.

It's also a good idea to distinguish between MIDI Volume (controller 7), and the dynamics generated with velocity. Use MIDI Volume to initially balance each instrument and create overall level adjustments and velocity (or pressure, if you've programmed a sound that way) to bring out particular notes. In other words, MIDI Volume covers general dynamic markings, and velocity works for individual accents. You can also use MIDI-programmed EQs to add emphasis to certain notes or certain instruments. (For more on these ideas, see "Fundamentals of MIDI Mixing" on p. 58). Every orchestral instrument has an acoustical foundation or fundamental tone that sets up the overtone content for that instrument. EQing for that pitch and its subsequent overtones adds a great deal of realism to a synth.—*Ron Reaser* spectrum sounding at once.

AMPLITUDE

Although a literal definition of amplitude is *loudness*, orchestrators are more concerned with "perceived loudness," which is relative. Listening to music at high decibel levels for an extended period acclimates the ears to the loudness until the volume is (relatively) comfortable. If a soft passage suddenly appears, an audience may barely notice it. The contrary also is true: A long, soft performance followed by an abrupt volume increase may leave an audience feeling as if their eardrums exploded. It's an orchestrator's responsibility to balance perceived loudness and softness to achieve the optimum musical effect.

There are several ways to increase perceived loudness without increasing volume. Raising the pitch or shortening a sound's attack are two effective methods. Also, limiters and compressors make a sound seem louder by decreasing the volume of its highest transients. The main body of the sound is proportionately louder and often cuts through a thick instrumental mix (see "Pumping Gain: Understanding Dynamics Processors" in the March 1991 EM).

Using amplitude effectively in orchestration requires enough dynamic range to permit an ideal contrast between loud and soft, as well as intermediate gradations of loudness. A symphony orchestra in a well-designed acoustic space posts a phenomenal range of approximately 70 decibels between its loudest and softest sounds. Synthesists must seek to maximize dynamic range relative to the listener.

The human ear's sensitivity to amplitude is greatest within a frequency range of 1,000 to 4,000 Hz (roughly the top two octaves of a piano). Within this fairly high range, the human ear perceives incredibly faint tones. Sensitivity falls off progressively for lower tones and rapidly for higher ones. Remember when you score for the treble region that this area takes little

ORCHESTRAL WRITING

If you use orchestral timbres in your music, you should know something about traditional orchestration. Countless books are dedicated to the subject, but here are a few basic facts:

(1) The orchestra is divided into four "choirs," each with various instruments: strings, woodwinds, brass, percussion. The vocal choir can also be added as a fifth category. These choirs offer very effective timbral contrast.

(2) Substantial doubling occurs between instruments of the various choirs to form countless timbral combinations.

(3) Sometimes musicians are grouped into a "section" (violins I and II, violas, soprano voices, etc.). These groups can be subdivided, and individual performers soloed or grouped into "chamber combinations" for effects where smaller masses of sound are contrasted with larger masses.

The synthesist can apply these principles to his or her medium by dividing sounds into timbral "choirs" or "groupings" for the purpose of contrast; doubling and layering sounds for added timbral variety; and "modularizing" or breaking down sounds into simpler components to form effective contrasts between small and large masses of sound.

A working knowledge of the history of the orchestra helps produce a convincing product. For example, the Oberheim Fat String sound is lush and full but completely out of place when compared to the string sounds produced by an orchestra performing Beethoven. Similarly, the TX81Z brass sounds may be fine for Schubert, but tinny and pale for the music of Bruckner. Like most acoustical instruments, each MIDI synth has its strengths and weaknesses. A good orchestrator makes the most of both.—*René Salm and Ron Reaser*

power to grab a listener's attention.

At the other end of the amplitude scale is the phenomenon known as *masking*, where a loud tone obscures softer ones occurring on, or near, the same pitch. The effect is more apparent on soft, higher-pitched tones, so masking is more pronounced in the bass region, where overtones are the strongest. Because of this, loud bass notes require lots of vertical space in an arrangement.(Vertical space refers to the intervals between notes in a chord.)

An old principle of classical orchestration is to distribute the sounds of a chord according to the harmonic series with the widest spaces toward the bass end. This particular distribution produces a full, blended sound, as the various fundamentals and their overtones reinforce each other. In this case, masking is avoided through the judicious choice of amplitude levels and timbres.

TIMBRE

My trusty dictionary defines timbre as "the character of a sound, as distinct from its pitch; hence the quality of sound that distinguishes one instrument from another." The perception of character requires attention to a sound's attack, envelope, and overtone structure.

Timbre is especially important in orchestration because an instrument's character prompts you to include it in an arrangement. Another basic principle of orchestration is maintaining interest through contrasting timbres, such as thin sounds against thick, fast attacks against slow, and bright sounds against dark.

Unlike pitch or amplitude, timbre is a subjective term. The lack of an objective measurement forces the orchestrator to break down the qualities of sound to accomodate the definition and placement of timbre in the work.

Attack. Sounds are easily constrasted on the basis of a short attack (percussive, plucked, explosive) or long attack (any type of sound with an initial rising amplitude envelope). Human ears are highly discriminating in this respect; a minute difference in the speed of the attack of two sounds renders an effective contrast.

Human ears are so sharp that attack transients occurring in the first few milliseconds of a sound are critical to our perception of that sound (a transient is a brief, non-sustaining overtone). **Fig. 1** shows a graphic representation of the enormously complex attack portion of a violin tone, representing the bite of the bow into the string.

Electronic instruments create contrasting timbres by attaching varied and complex attacks to base sounds. Sampling technology made this easy, and Roland uses the concept as the basis for their L/A instruments.

Thickness. Timbres can be contrasted on the basis of those with many overtones (thick, rich, colorful) and those with few overtones (thin, clear). Examples of thick timbres are gongs, timpani, and cymbals. The flute is a classic thin timbre.

A thick timbre also must include some element of sustain. A complex set of overtones with a short duration won't survive the attack portion of the sound; it produces instead a dry, distinctive timbre, much like a woodblock or marimba.

A thin timbre has relatively few overtones. These "transparent" timbres are useful when clarity of sound is important. Thin timbres don't have to contrast directly with thick ones. In fact, such opposition is rather lopsided. A more musical result is attained when two or more thin timbres are combined. This method also works for building thick timbres, as it adds maximum flexibility to your sounds. Simply break down thicker sounds into two or more thinner components, and use them as building blocks.

Pitch vs. Unpitched. Pitch may be considered a parameter of timbre, as pitch depends on a certain regularity in the overtone structure. Also, note that many unpitched sounds actually possess some sense of pitch. The howling of wind reveals a fundamental that appears and disappears, sometimes moving higher, sometimes lower. Therefore, unpitched sounds can be

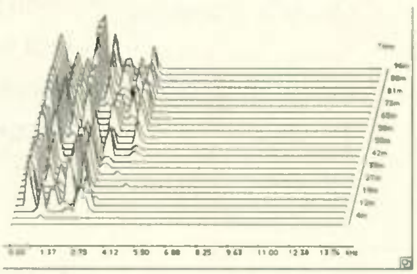

FIG. 1: A harmonic analysis of the attack of a violin tone, showing the first few harmonics. Notice how each harmonic has amplitude envelopes far more complex than typical ADSRs.

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layered with pitched ones to evoke timbre thickness, color, or atmospheric effect. An unpitched timbre with a short or definite attack (drums, etc.) gives a pitched sound rhythmic definition and percussiveness. Truly unpitched sounds often have a formant (a sharp amplitude peak in the frequency spectrum) that, for orchestrational purposes, places them in the treble, midrange, or bass regions.

PLACING SOUNDS

One of an orchestrators most important tasks is determining where to place sounds to achieve an appropriate balance within the audio spectrum. Obviously a thicker sound takes more space than a thin sound, so a work with many notes (fast-running scales, arpeggios, many notes sounding at the same time, etc.) may require timbres with less overtones and clean attacks.

Use thicker sounds if a piece has ample space between chord notes (theorists call this *open position*), because there's more vertical room for each note. Conversely, a crowd of notes stacked up in a small area of the audio spectrum (a *closed position*) requires thin sounds or the passage may resemble mud, not music (see **Fig. 2**).

Low notes have multiple strong overtones, which often makes them thicker than high notes. Except when plucked (pizzicato strings, etc.), low notes also move slower, last longer, and require more vertical clearance than high notes. Consequently, it's

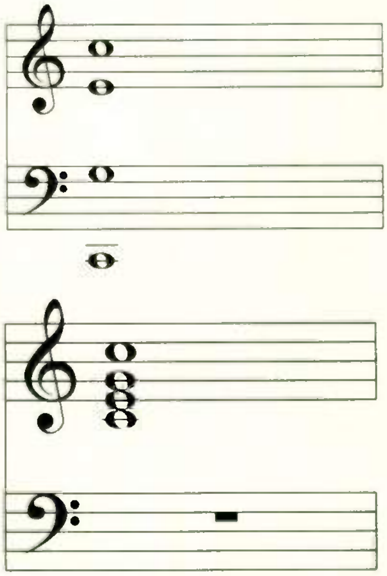

FIG. 2: The same chord shown in an open position on the top and a closed position on the bottom.

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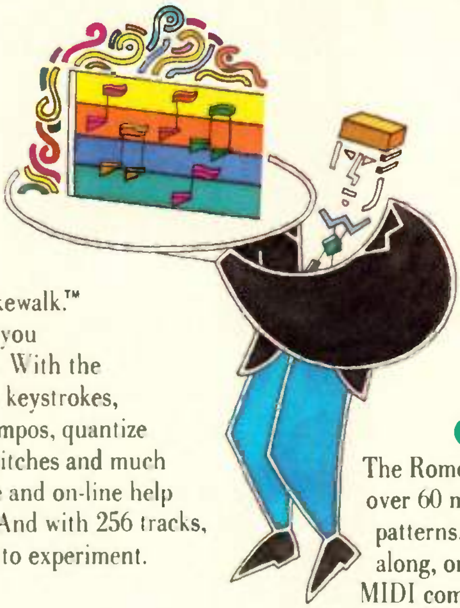

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much easier to muddy up the low end of the audio spectrum.

Conversely, the higher the note, the thinner the sound. As a result, high notes take up relatively little vertical space. Color, punch, and presence can be evoked safely by doubling tones at the octave, third, fifth, sixth, etc. Also, multiple patches can be assigned to high notes without immediate fear of creating musical pea soup.

FERTILE GROUND

Foreground, middleground, and background comprise the electronic orchestrator's bread-and-butter. They permit differentiation of musical sounds, while maintaining clarity, interest, and balance in the overall composition.

These spatial dimensions are well-suited to electronic orchestration because the accessible sounds of one electronic instrument can represent the equivalent output of a group of acoustic musicians. Symphonists must tailor the music to the instruments at hand. However, synthesists can tailor instruments to the demands of the music. Therefore, synthesists relegate hard work that traditionally went into arranging to a "pre-orchestrational" stage: sound design and editing.

The Foreground. The foreground typically consists of a single-line melody, motive, or riff. It also can be defined as anything that stands out in a listener's perception, whether this is accomplished through contrast in timbre, pitch, articulation, or increased amplitude.

Motion moves an element into the foreground. Simply playing a melody in sixteenth or eighth notes, while the remaining parts move in quarter notes, effectively divides the music into foreground and background (other things being equal). One exception to this rule is a florid accompaniment, where background material consists of subdued arpeggios or similar patterns.

Articulation refers to phrasing, embellishments, accents, and characteristics of attack and decay such as staccato and legato. An articulated line always receives more attention than an unarticulated one. In small doses, effects like vibrato and pitch bend enhance articulation.

Increasing relative volume of an instrument obviously pushes it forward, as does adding partials (such as opening up a filter). In fact, a brighter sound often seems louder than a sound with less partials. Shortening the attack of a sound gives it more bite and generally adds percussiveness.

Soloing is another obvious, but often overlooked, method for bringing elements to the foreground. Voice doubling, which can occur at the unison, octave, and/or other intervals, also works. Although pitch shifters create this effect, they are less effective than actual doubling, where timbres, volume levels, panning, and articulation can be individually altered. Other effects, such as chorus, reverb, and tremolo, thicken sounds and aid articulation. A powerful, but underused technique called *heterophony* (see sidebar, "Heterophony," for more) goes beyond voice doubling and effects.

Finally, other things being equal, a higher-pitched sound maintains the foreground more effectively than a lower-pitched one. This is why the top notes of a chord are heard distinctly, and melodies are usually placed on top of homophonic textures of melody plus accompaniment.

The Middleground. Often music consists only of foreground and background material. But when present, the middleground consists of secondary melodic material or motivic fragments that possess horizontal motion but are less obtrusive than foreground material. Examples include contrapuntal melodies, ostinato figures, and motives punctuating foreground material.

The bass line generally falls into the middleground (rather than the background), because it is generally strong enough to support the harmonies above it.

For maximum clarity and effectiveness, the middleground and foreground should contrast as much as possible. If you use a legato line for foreground material, consider a staccato articulation for the middleground.

The Background. The least obtrusive of all sections, the background generally exhibits little horizontal motion, but maintains an extended vertical dimension. Consequently, it often includes sustained tones or chordal "filler" and plays an important harmonic role. Devices used to create a foreground (volume, brightness, articulation) generally are subdued when treating the background.

Be careful with the background, because a heavy scoring hand damages the clarity of the overall composition. With bass, one sustained note at a time may work just fine, while the treble may require three or more notes to provide needed emphasis.

Mixing the background notes in two (or all three) registers effectively furnishes a bed for foreground material and virtually guarantees a full sound if clarity is maintained.

CLASSIFYING SOUNDS

Every synthesist should organize their vast collection of sounds for easy access and use. Much like attempting to describe timbre, classifying sound is subjective and depends on one's working methods. Nevertheless, a general and useful way to organize sounds is by the qualities of timbre that have been discussed: attack, thickness, instrumental timbre, bright sounds, dark sounds, pitched sounds, and unpitched sounds. Of course, some sounds can be classified in more than one category.

Change useless preset names like Zoot String, Bible Pad, or Schluba. These only confuse the creative process. Drop redundancies (the word "string" is superfluous in a string bank),

HETEROPHONY

Though scary sounding (heterophonic means "different-voiced"), this term refers to an often overlooked technique of combining a melody with simultaneous variations. Heterophony is a great way to underline a melody and bring it to the foreground. It also provides an alternative to voice doubling or the use of effects like chorusing. As shown in Fig. 3, the technique involves splitting up a single melody into a number of parts.

Heterophony energizes creative juices because there's almost no limit to the number of simultaneous variations to which a given melody can be subjected. If done properly, the resulting texture is almost always interesting, musical, and effective. It's also highly colorful and particularly well-suited to the electronic musician because of such diverse possibilities as independent panning of the linked melodic strands and elaborate echo effects. The technique is most effective when the melodic strands have contrasting timbres.

Heterophony should not be confused with *hocketing* or *pointillism*, two other terms for musical textures. Hocketing refers to the distribution of a melodic line among two voices in such a way that when one voice sounds, the other is silent. Hocketing has one melodic line, whereas heterophony always has at least two strands. Pointillism, on the other hand, is a texture where pitches are sounded in various timbres, and largely in linear isolation from one another, rather than as melodies.

Heterophony can effectively underline a melody and bring it to the foreground. It also manages to combine foreground and middleground elements, requiring only a little background material to create a rich, exciting texture. Use it!—*René Salm*

FIG. 3: Heterophonic textures split up melodies into at least two strands. This works best with contrasting timbres.

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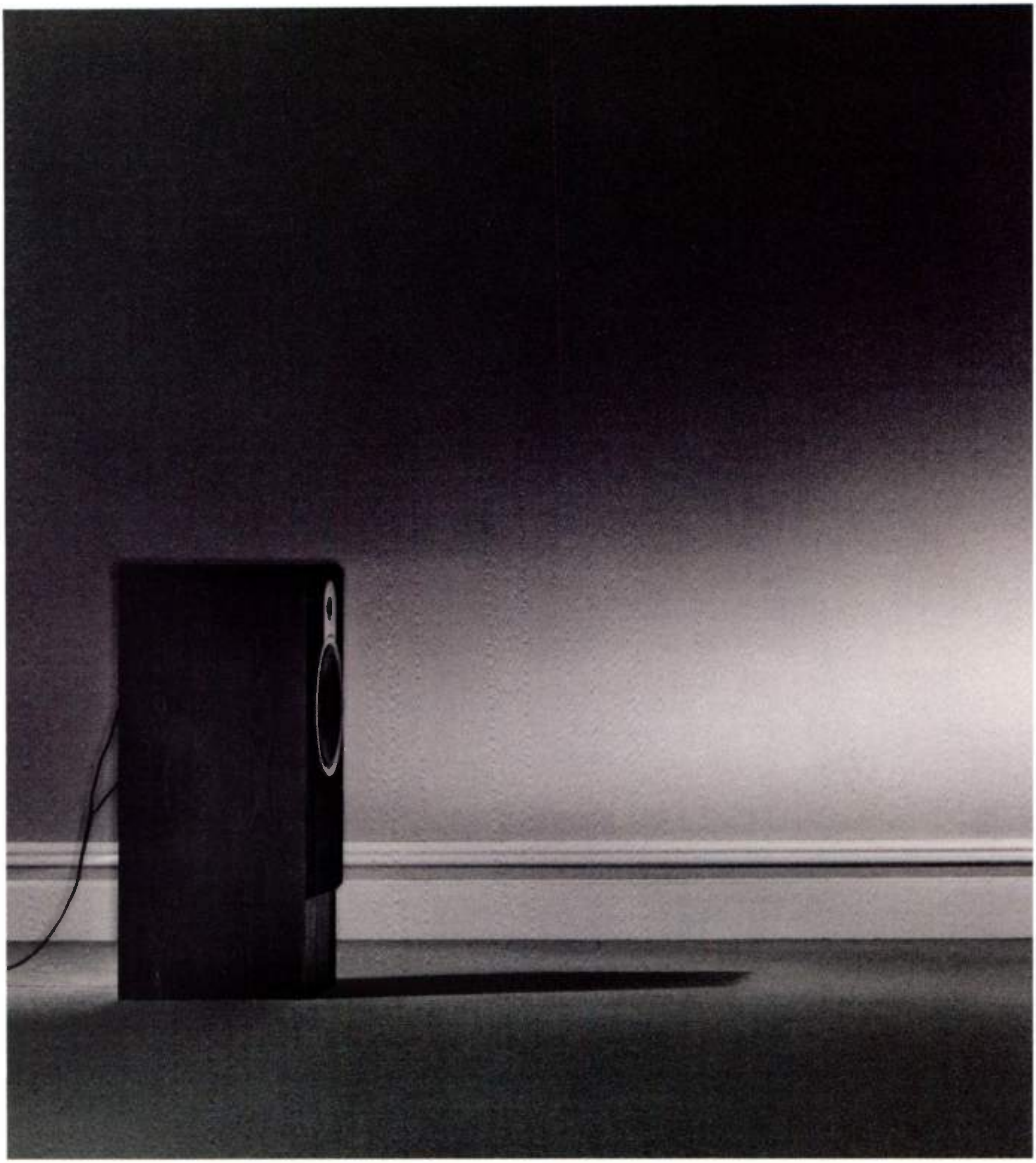

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IEC Tape Specifications				
Tape Selector Position				IEC II High (CrO2)
			Unit	XLII-S 100
Physical Properties	Backing Material-			Dual Surface Tensilized Polyester Base-Film
	Tape Width		mm	3.81
	Coating Thickness		μm	4.5
	Total Thickness		μm	11.0
	Yield Strength		N	5.5
	Breaking Strength		N	10.5
Magnetic Properties	Magnetic Material		—	Black Magnetite
	Coercivity		kA/m (Oe)	58(720)
	Retentivity		mT(G)	190(1900)
	Squareness		—	0.90
Electro-acoustic Properties	Optimum Bias		dB	+1.0
	Sensitivity	315 Hz	dB	+0.5
		6,300 Hz	dB	+1.5
		12,500 Hz	dB	+3.0
		16,000 Hz	dB	+4.5
	Output Uniformity at 315 Hz		dB	0.3
	Output Uniformity at 8 kHz		VU	0.4
	Dynamic Range	315 Hz	dB	65.5
		10 khZ	dB	57.5
	MOL (Maximum Output Level)	315 Hz	dB	+6.0
		10,000 Hz	dB	-2.0
	AC Bias Noise		dB	-59.5
	Erasing Effect		dB	70
	Print-Through		dB	51
All features and specifications are subject to change without notice. (Specifications are measured with reference to: IEC Type II Reference Tape U564W in accordance with established IEC procedures.)				

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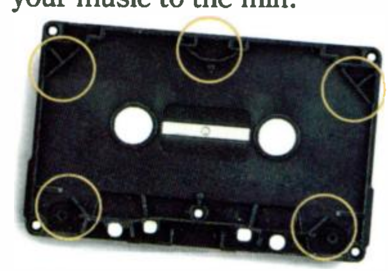
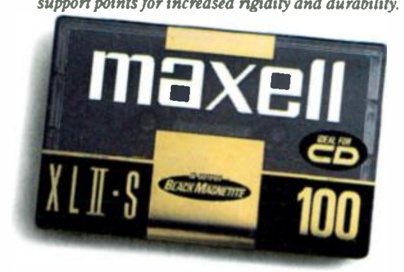


XLII-S high resonance-damping cassette shell has five support points for increased rigidity and durability.



TEN PRINCIPLES OF ELECTRONIC ORCHESTRATION

- Use the whole acoustic range: treble, midrange, and bass.
- Use the entire amplitude range from soft to loud.
- Be colorful: Use all the main timbral groups.
- Use a wide variety of controllers to input data.
- Use the principle of foreground, middleground, and background.
- Create variety through contrast: Alternate acoustic regions, amplitude levels, and timbral groups in the foreground, middleground, and background; occasionally solo one acoustic region and/or timbre.
- Be transparent: Never overload any acoustic region—treble, midrange, or bass.
- Be economical: Say what you have to say with the least possible means.
- Creatively use the acoustic space surrounding the listener (sound placement, panning, and motion).
- Use effects, but only when musically necessary.

and use abbreviations that aptly describe a patch (Horn +5th, Fast Ding, etc.). However, complex or multilayered patches often are easier to identify by a colorful "handle," such as Jungle Chirps.

Highly distinctive patches have great novelty appeal but don't necessarily wear well. As a rule of thumb, the more distinct the patch, the less often it should be heard.

Beware that a sound's complexity diminishes its flexibility. A sample of the L.A. Philharmonic playing a fortissimo chord loses definition and character as it is moved up and down the keyboard. In general, the more complex the patch, the narrower its acoustic range.

Finally, complex sounds push the limits of clarity. It *is* possible to create quality-sounding, effective, and complex layered patches, but it takes extra care. To insure clarity within a complex sound structure, use the same principles required for good orchestration: Spread tones over the acoustic spectrum, provide each tone sufficient vertical distance, and judiciously mix distinct timbres.

THE FINAL NOTE

Despite the many explanations of various principles, orchestration has few hard and fast rules. The principles and concepts offered in this article are only suggestions. In the final analysis, your ear is the judge. No two people orchestrate the same way, so every synthesist eventually develops their own sound.

René Salm *is presently composing a concerto for piano and synthesizers and preparing a solo electronic performance of Bach's* Goldberg Variations. *He hails from Eugene, Oregon.*

EM Guide to Notation Software

The heritage of music publishing is passed to your computer.

By Christopher Yavelow

More than 600 years have been invested into the development of conventional music notation. Despite this fact, music software developers initially excluded conventional notation as the ideal representation of musical data for computers and attempted to foist new systems, such as alphanumeric event lists and "piano roll" bar graphs, in its place.

However, it's not easy to ignore six centuries of musical language. Music publishers and engravers, copyists, orchestrators, arrangers, and performers do not communicate via printed event lists and bar graphs. The development of music notation software marries the computer's unlimited editing power to the symbols that preserve music through the ages.

There are many good arguments for pursuing the field of music notation by computer. Due to repetition and modified repetition, cutting and pasting easily arranges musical information that is reusable within the same piece. Automatic proofreading of parameters such as rhythm and instrumental range is simplified by the computer, and interactive "proofhearing" and ease of editing are obvious advantages for composers and orchestrators. In the music publishing industry, computers make it possible for several people to work on the same job. A well-devised notation software program substitutes for manual dexterity and, to a certain extent, musical knowledge.

The development of music notation software was stimulated by the introduction of Adobe Systems' printer-independent PostScript music font, *Sonata*, designed by Cleo Huggins in 1986. Amazingly, no standardized, computer-based musical symbol set existed until the release of the Adobe font. Today, over two dozen PostScript

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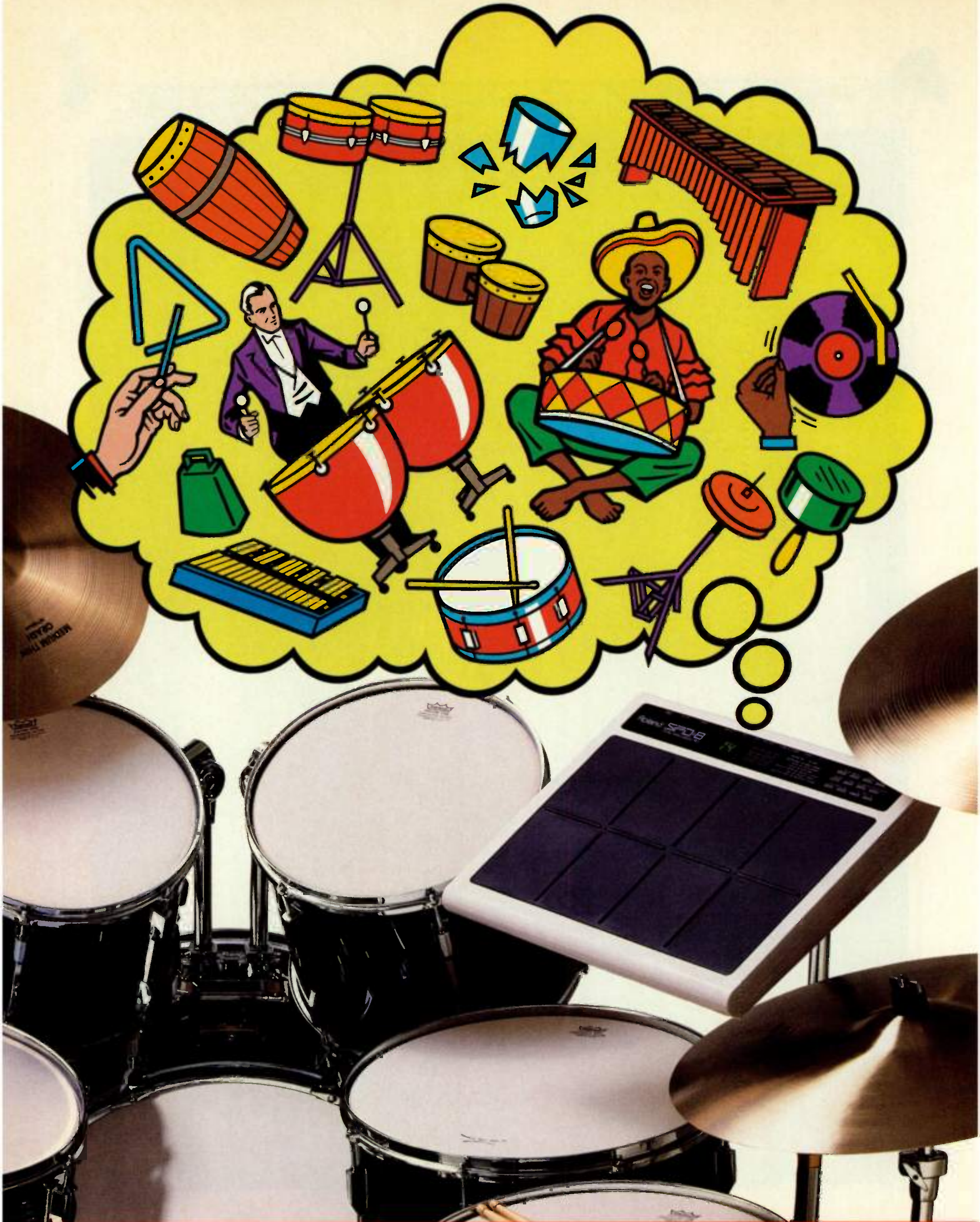

Roland SPD-8

EM GUIDE TO NOTATION SOFTWARE

Manufacturer/Product	Requirements	Viewing Modes	Max. # of Staves	User Preferences	Rhythmic Resolution	Optional Noteheads
MACINTOSH						
Cerl Sound Group Lime 0.6f	2 MB	page, limited screen	64	yes	128th note	diamond, X, slash, grace, cue, other
Coda Finale 2.6.1	1 MB (2 MB and hard disk recommended)	page, scroll, screen	128	yes	4,096th note	diamond, X, slash, grace, cue, other
Coda MusicProse 2.1	1 MB (2 MB and hard disk recommended)	page, scroll, screen	32	no	128th note	slash, grace, other
Electronic Arts Deluxe Music Construction Set 2.5	512 KB	page, scroll, screen	32	no	32nd note	n/a
Great Wave ConcertWare MIDI 5.0	1 MB	scroll	8	no	32nd note	diamond, X, slash, grace, cue
Mark of the Unicorn Professional Composer 2.3m	1 MB	page, scroll	40	no	128th note	diamond, X, slash, limited grace
Music Publisher Music Publisher 2.5.2	1 MB (2 MB recommended)	page, screen	100	no	128th note	diamond, X, slash, grace, cue
Passport Encore 2.1	1 MB (2 MB recommended)	page, scroll, screen	64	no	128th note	diamond, X, slash, grace
Passport NoteWriter II 2.6	1 MB	page, screen	40	yes	64th note	diamond, X, limited slash, grace, cue, other
Pygraphics Music Writer 1.1	2 MB	page, scroll, limited screen	40	no	64th note	diamond, X, slash, grace, other
PC						
alla breve Musicad 1.1	640 KB, graphics card, hard disk, mouse	page, zoom screen	50	yes	128th note	n/a
Coda Finale 2.0	2 MB, hard disk, *Windows 3.0*, mouse	page, scroll, screen	128	yes	128th note	diamond, X, slash, grace, cue, other
Dr. T's Copyist Apprentice 2.0	640 KB	scroll, screen	16	no	64th note	diamond, X, grace, cue
Dr. T's Copyist DTP 2.0	640 KB	scroll, screen	16	yes	64th note	diamond, X, grace, cue, other
Erato Music Manuscripter 1.0	640 KB, (4 MB EMS recommended), math coprocessor, digitizer tablet	page, screen, scroll	64	yes	128th note	diamond, X, slash, grace, cue, other
Grandmaster MusicEase 3.0	512 KB (640 KB recommended), graphics card	page, scroll, zoom screen	unlimited	yes	64th note	X, grace, cue
Passport Encore 2.1	2 MB, mouse	page, scroll	64	yes	128th note	diamond, X, slash, grace
Passport Score 3.06	640 KB, hard disk (mouse and math coprocessor recommended)	page, screen	32	yes	256th note	diamond, X, slash, grace, cue, other
Pygraphics Music Writer 0.9e	640 KB, graphics card, mouse	scroll, limited screen	40	no	64th note	diamond, X, slash, grace, other
SongWright SongWright V 5.0	320 KB, graphics card	screen	unlimited	yes	32nd note	diamond, X, slash, grace, cue, other
Teach Services Laser Music Processor 3.2	512 KB, graphics card	screen	unlimited	limited	32nd note	n/a
Temporal Acuity MusicPrinter Plus 4.1	640 KB, (hard disk recommended), graphics card	page, scroll, screen	42	no	64th note	diamond, X, slash, grace, cue, other
Theme Software Theme 3.5	640 KB (hard disk recommended), graphics card	page, scroll, screen	40	no	128th note	diamond, X, slash, grace, cue, other
thoughtprocessors The Note Processor 2.2	640 KB, (hard disk and mouse recommended), graphics card	page, scroll, screen	50	yes	256th note	diamond, X, slash, grace, cue, other
AMIGA						
Dr. T's Copyist Apprentice 1.65	1 MB	scroll, screen	16	no	64th note	diamond, X, grace, cue
Dr. T's Copyist DTP 1.65	1 MB	scroll, screen	16	yes	64th note	diamond, X, grace, cue, other
Electronic Arts Deluxe Music Construction Set 1.2	512 KB	scroll	48	no	32nd note	n/a
ATARI						
C-Lab Notator 3.1	1 MB	page, scroll	32	yes	96th note	diamond, X, grace, cue, other
Dr. T's Copyist Apprentice 1.60	1 MB	scroll, screen	16	no	64th note	diamond, X, grace, cue
Dr. T's Copyist DTP 1.60	1 MB	scroll, screen	16	yes	64th note	diamond, X, grace, cue, other
Hybrid Arts EZ Score Plus 1.1	1 MB	scroll	3	yes	128th note	diamond, X, other

music fonts are available for creating professional-looking music.

APPLICATIONS

There are two fundamental reasons for using music notation software: the manipulation of preexisting music and the creation of new music. The first category includes publishing, engraving, copying, arranging, and orchestration. The second category includes composition and musical realization.

Notation software performs a variety of tasks: (1) Production of a final legible copy for publication or distribution; (2) Extraction of parts required for performance; (3) Creation of new versions of existing works in different keys; (4) Rapid creation of a new arrangement via software tools; (5) Easy musical idea manipulation via software tools; (6) Creation of an electronic realization as an instant reference "Polaroid" (as they say in Hollywood), or for "proofhearing" and; (7) Creation of an electronic realization as an end in itself.

The order of this list is not arbitrary. It progresses from tasks concerned primarily with form (the graphical elements, or "look") to those concerned with content (the musical interpretation of symbols).

EM GUIDE TO NOTATION SOFTWARE (CONTINUED)

Manufacturer/Product	Clefs	Mixed Key Signatures	Complex Meters	Chord Symbols	Cross-Staff Beaming	Tuplet Range	Music Input
MACINTOSH							
Cerl Sound Group Lime 0.6f	G, F, alto, tenor, baritone, small, perc, octave	no	no	text	yes	64:64	RTM[1], STM[2], M[5]
Coda Finale 2.6.1	G, F, alto, tenor, baritone, small, perc, octave	yes	yes	fretboard, text	yes	unlimited	RTM[1], STM[2], IMF[3], CK[4], M[5]
Coda MusicProse 2.1	G, F, alto, tenor, baritone, small, perc, octave	no	no	fretboard, text	no	99:99	RTM[1], STM[2], IMF[3], CK[4], M[5]
Electronic Arts Deluxe Music Construction Set 2.5	G, F, alto, tenor	no	no	fretboard	no	3, 5	STM[2], CK[4], M[5]
Great Wave ConcertWare MIDI 5.0	G, F, alto, tenor, octave	no	no	fretboard, text	no	3	RTM[1], STM[2], IMF[3], CK[4], M[5]
Mark of the Unicorn Professional Composer 2.3m	G, F, alto, tenor, perc, octave	no	no	text	no	unlimited	STM[2], CK[4], M[5]
Music Publisher Music Publisher 2.5.2	G, F, alto, tenor, limited baritone, small, perc	no	yes	n/a	yes	99:99	RTM[1], STM[2], 9
Passport Encore 2.1	G, F, alto, tenor, perc, octave	no	no	text	no	16:16	RTM[1], STM[2], IMF[3], CK[4], M[5]
Passport NoteWriter II 2.6	G, F, alto, tenor, baritone, small, perc	no	yes	fretboard	yes	unlimited	CK[4], M[5]
Pygraphics Music Writer 1.1	G, F, alto, tenor, perc, octave	no	no	text	no	3, 5, 7, 9	RTM[1], STM[2], IMF[3], CK[4], M[5]
PC							
alla breve Musicad 1.1	G, F, alto, octave	yes	yes	fretboard, text	yes	unlimited	RTM[1], IMF[3], CK[4], M[5]
Coda Finale 2.0	G, F, alto, tenor, baritone, small, perc, octave	yes	yes	fretboard, text	yes	unlimited	RTM[1], STM[2], IMF[3], CK[4], M[5]
Dr. T's Copyist Apprentice 2.0	G, F, alto, tenor, baritone, small, perc	yes	yes	fretboard, text	yes	unlimited	IMF[3], CK[4]
Dr. T's Copyist DTP 2.0	G, F, alto, tenor, baritone, small, perc	yes	yes	fretboard, text	yes	unlimited	IMF[3], CK[4]
Erato Music Manuscripter 1.0	G, F, alto, tenor, baritone, small, perc, octave	yes	limited	text	yes	unlimited	STM[2], CK[4], 10
Grandmaster MusicEase 3.0	G, F, alto, tenor	yes	yes	fretboard	no	unlimited	RTM[1], STM[2], IMF[3], CK[4]
Passport Encore 2.1	G, F, alto, tenor, baritone, octave	no	yes	fretboard	yes	16:16	RTM[1], STM[2], IMF[3], CK[4], M[5]
Passport Score 3.06	G, F, alto, tenor, baritone, small, perc, octave	yes	yes	fretboard	yes	99:99	STM[2], IMF[3], CK[4]
Pygraphics Music Writer 0.9e	G, F, alto, tenor, perc, octave	no	no	text	no	3, 5, 7, 9	RTM[1], STM[2], IMF[3], CK[4], M[5]
SongWright SongWright V 5.0	G, F, alto, tenor, baritone	no	no	fretboard	no	3	RTM[1], STM[2], IMF[3], CK[4]
Teach Services Laser Music Processor 3.2	G, F, alto, tenor	yes	yes	text	yes	unlimited	RTM[1], STM[2], IMF[3], CK[4], M[5]
Temporal Acuity MusicPrinter Plus 4.1	G, F, alto, tenor, baritone, small, perc, octave	yes	yes	fretboard, text	yes	unlimited	RTM[1], STM[2], IMF[3], CK[4], M[5]
Theme Software Theme 3.5	G, F, alto, tenor, baritone, perc, octave	no	yes	text	yes	31:31	RTM[1], IMF[3], CK[4]
thoughtprocessors The Note Processor 2.2	G, F, alto, tenor, baritone, small, perc, octave	yes	yes	fretboard, text	limited	unlimited	RTM[1], IMF[3], CK[4], M[5]
AMIGA							
Dr. T's Copyist Apprentice 1.65	G, F, alto, tenor, baritone, perc	yes	yes	fretboard, text	no	unlimited	IMF[3], CK[4]
Dr. T's Copyist DTP 1.65	G, F, alto, tenor, baritone, perc	yes	yes	fretboard, text	no	unlimited	IMF[3], CK[4]
Electronic Arts Deluxe Music Construction Set 1.2	G, F, alto, tenor	no	no	fretboard	no	3, 5	STM[2], CK[4]
ATARI							
C-Lab Notator 3.1	G, F, alto, tenor, baritone, perc, octave	yes	no	fretboard, text	no	15:15	RTM[1], STM[2], IMF[3], CK[4]
Dr. T's Copyist Apprentice 1.60	G, F, alto, tenor, baritone, perc	yes	yes	fretboard, text	no	unlimited	IMF[3], CK[4]
Dr. T's Copyist DTP 1.60	G, F, alto, tenor, baritone, perc	yes	yes	fretboard, text	no	unlimited	IMF[3], CK[4]
Hybrid Arts EZ Score Plus 1.1	G, F	yes	yes	fretboard, text	no	99:99	limited RTM[1], STM[2], CK[4], M[5]

FOOTNOTES

RTM[1] = real-time MIDI, STM[2] = step-time MIDI, IMF[3] = import MIDI files, CK[4] = computer keyboard, M[5] = mouse, L[6] = lyrics, AT[7] = annotative text, OT[8] = other text, 9 = includes presto keyboard, 10 = includes digitizer tablet

READING THE CHART

To assist you in choosing a notation package, the chart on pp. 47 to 49 lists some of the more important features to consider. Beware of simply picking the program with the most features (unless you happen to be an engraver). In most cases, the more features a program has, the harder it is to master and the slower it runs. Also, don't pay for features you'll never use. Determine what you need in a notation program and purchase accordingly.

SYSTEM REQUIREMENTS

This is the absolute minimum firepower needed to run the software. Remember that it is in the manufacturer's interest to state that a program runs on a minimum system, even if it takes ten minutes to copy four bars of music. Try the software on the computer you plan to use, and bring a stop watch.

Beware of companies that claim a hard disk is not required. Notation files are often 10 to 100 times larger than MIDI sequencer files of the same data. While a single-page lead sheet may be generated with a floppy drive-based system, larger files require a hard disk.

VIEWING MODES

Some programs require viewing and editing music as if it were a virtual *page* on the computer screen. Other programs take the *scroll* approach: The music appears on one continuous staff that might be 20 or 30 feet long in the real world. The more elegant programs allow switching between page view and

Text Input	PostScript	Transposition	Part Extraction	MIDI Channels per Stave	MIDI Expression Marks	MIDI Editing	Export MIDI Files	Price
L[6], AT[7], OT[8]	yes	clef, mode	yes	2	limited	no	no	$160
L[6], AT[7], OT[8]	yes	chromatic, diatonic, enharmon c, mode	yes	4	yes	yes	yes	$749
L[6], limited AT[7], OT[8]	yes	chromatic, diatonic	yes	1	no	no	yes	$399
L[6], AT[7]	yes	chromatic, limited diatonic	no	1	no	no	no	$129
L[6], AT[7], OT[8]	yes	chromatic	no	1	yes	yes	yes	$189
L[6], limited AT[7], OT[8]	yes	chromatic, diatonic, enharmonic, limited mode	limited	1	no	no	no	$495
L[6], AT[7]	yes	chromatic, clef, mode	yes	4	yes	no	no	$495[9]
L[6], AT[7], OT[8]	yes	chromatic, enharmonic	limited	4	no	yes	yes	$595
L[6], limited AT[7]	yes	chromatic, limited enharmonic	no	4	no	no	no	$595
L[6], AT[7], OT[8]	yes	chromatic, diatonic, enharmonic	yes	1	yes	limited	yes	$595
L[6], AT[7]	yes	chromatic	yes	16	no	no	yes	$295
L[6], AT[7], OT[8]	yes	chromatic, diatonic, enharmonic, mode	yes	4	yes	yes	yes	$749
L[6], AT[7], OT[8]	no	n/a	no	16	no	no	yes	$149
L[6], AT[7], OT[8]	yes	chromatic	yes	16	no	no	yes	$450
L[6], AT[7], OT[8]	no	chromatic, diatonic, enharmonic, mode	yes	2	yes	no	no	$1,050[10]
L[6], AT[7]	no	chromatic, diatonic, enharmonic, mode	yes	1	no	no	yes	$295
L[6], AT[7], OT[8]	yes	chromatic, enharmonic	limited	4	no	yes	yes	$595
L[6], AT[7], OT[8]	yes	chromatic, diatonic, enharmonic	yes	1	no	no	no	$995
L[6], AT[7], OT[8]	yes	chromatic, diatonic, enharmoni	yes	1	yes	limited	yes	$595
L[6], AT[7], OT[8]	no	chromatic, diatonic, enharmonic, mode	yes	2	no	no	yes	$99
L[6], AT[7]	no	limited chromatic	no	16	no	no	yes	$99
L[6], AT[7], OT[8]	no	chromatic, clef, diatonic, enharmonic, mode	yes	16	yes	yes	yes	$595
L[6], AT[7], OT[8]	no	chromatic, clef, diatonic, mode	yes	1	limited	no	yes	$395
L[6], AT[7], OT[8]	yes	chromatic, clef, diatonic, enharmonic, mode	yes	16	yes	limited	yes	$295
L[6], AT[7], OT[8]	no	n/a	no	16	no	no	no	$129
L[6], AT[7], OT[8]	yes	chromatic	yes	16	no	no	yes	$349
L[6], AT[7]	yes	chromatic, limited diatonic	no	1	no	no	no	$99
L[6], AT[7], OT[8]	no	chromatic, diatonic, enharmonic mode	yes	16	yes	yes	yes	$699
L[6], AT[7], OT[8]	no	n/a	no	16	no	no	no	$149
L[6], AT[7], OT[8]	yes	chromatic	yes	16	no	no	yes	$399
L[6], AT[7], OT[8]	no	chromatic	yes	16	no	no	no	$149

scroll view, with no restrictions on editing in either domain. *Screen* means the image displayed on the computer monitor is identical (or extremely close) to what appears on the printout (WYSIWYG for you computer types).

MAXIMUM NUMBER OF STAVES

There is a correlation between price and the number of staves a program supports, so determining your needs is important. Notating symphonies requires more than eight staves, but seldom more than 32. However, if notation software is used to drive synthesizers, one stave for each of the sixteen MIDI channels is necessary. If your MIDI interface supports multiple channels, you may want as many as 64 staves.

USER PREFERENCES

Many professional engravers have cultivated a style to the extent that a single manuscript page is easily identified as, for example, a Schirmer, Henle, or Hal Leonard edition. Some programs permit user customization by allowing adjustment of variables such as beam angle and thickness, dot offset, and stem width. This sort of control adds a good deal to the price and many kilobytes to the size of a notation program.

In many ways, a music notation program functions like page-layout software; as with those packages, it's important to have complete positioning control over each of the elements that make up a page. The ability to move individual elements such as expression marks, accidentals, and beams often makes the difference between professional-looking music and, "not-quite-professional" results.

RHYTHMIC RESOLUTION

The available symbol set is extremely important. If the program doesn't allow 128th notes, and suddenly you need to write one, you're out of luck. The chart displays the minimum note value the program can generate.

OPTIONAL NOTEHEADS

Having the option of using diamond noteheads is important if you're notating string harmonics. Likewise, guitar notation may require slashes and percussion transcription Xs. If you include cues in your score, you need the ability to intermix notehead sizes on the same staff. This is another area to ensure a program meets your needs.

CLEFS

Clefs are yet another individual case. Chopin composed most of his *oeuvre* using treble (G) and bass (F) clefs exclusively.

MIXED KEY SIGNATURES

Most notation programs support standard circle-of-fifths key signatures. But if you need to mix sharps and flats in a signature, make sure the software allows it. Also, if you change key mid-song, not every program provides for key signature cancellations (a bank including natural signs indicates which accidentals in the previous key are no longer in effect).

COMPLEX METERS

Most programs provide for standard time signatures. If complex (2 + 4/8), fractional (3.5/4), or multiple (different number of beats but with coinciding barlines) meter signatures are characteristic of your musical language, software options are limited. Provisions for alternating signatures (6/8 + 3/4) such as Leonard Bernstein used in "America" are almost non-existent, but there are obvious work-arounds if the software allows them (change the meter signature at every bar, or use triplets).

CHORD SYMBOLS

Many programs now offer the option of placing chord symbols over staves. Some offer text symbols only, others provide fretboard notation, and some allow both. The best implementation of chord symbols automatically transposes them when the music is transposed to a new key. Some programs allow specification of a unique MIDI channel for chord symbols to play back on, complete with voicing options. If your software lacks a chord symbol option, you sometimes can get around this by creating a lyric line above the staff and using one of the available chord-symbol fonts.

CROSS-STAFF BEAMING

Diagonal beaming is now the standard for notation programs, so don't settle for anything less. For notating more complex keyboard music, however, you also should look for beams that can cross staves.

TUPLET RANGE

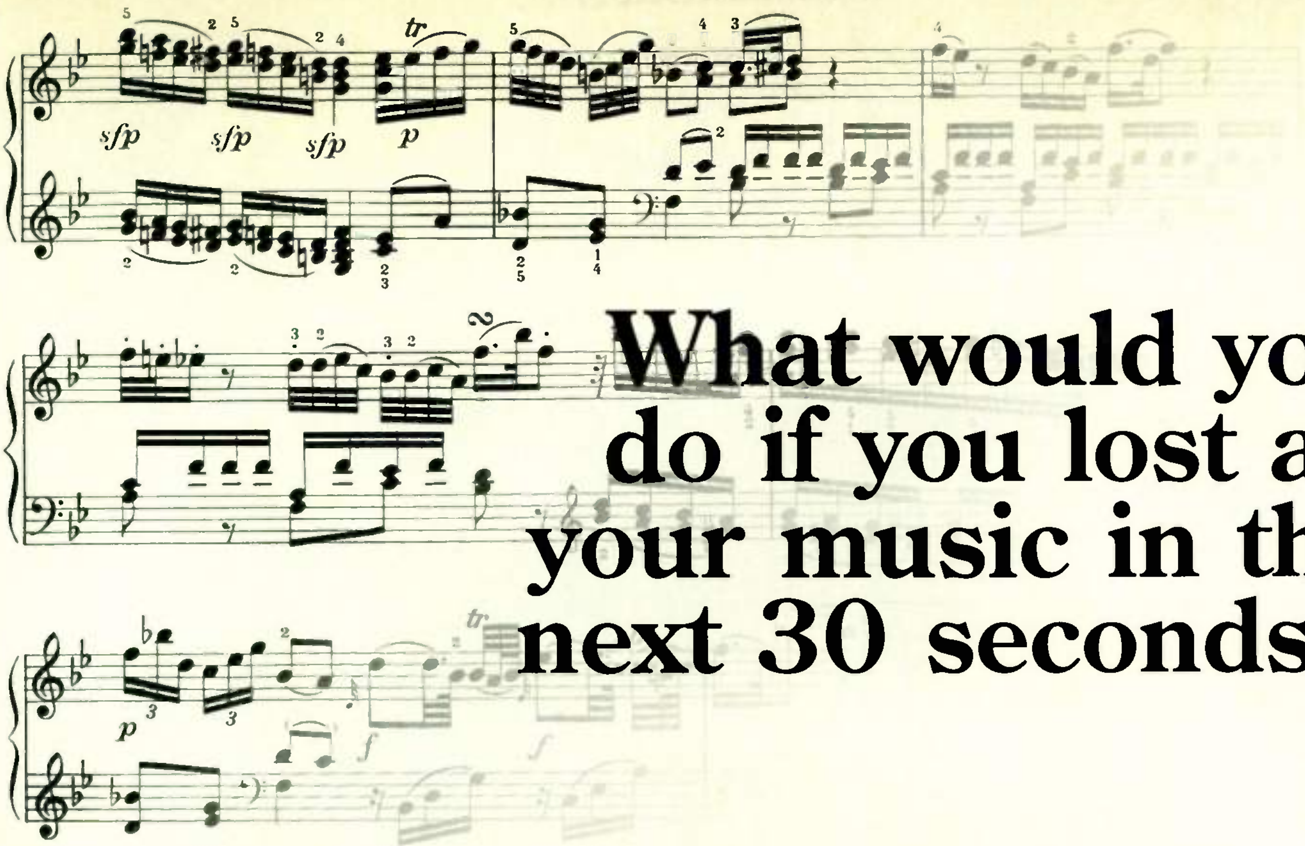
Notation programmers often labor under the false assumption that no one needs to notate any tuplet greater than a triplet. Some provide for triplets and quintuplets at best. The ratio in this column indicates the maximum tuplet size. With this in mind, "99 : 99" means any tuplet up to 99 in the time of 99 is notatable, including, for example, 87 in the time of 91. A lone "3," "5," or "7" indicates a limitation to triplets, quintuplets, or septuplets, respectively.

MUSIC INPUT

Fast and efficient data entry is the most difficult and important task notation programs fulfill. In many cases you'll spend more time entering the music than anything else, so it's important to make the process easy and offer as many choices as possible. Many programs use MIDI as their main form of input, but the type of MIDI implementation varies widely.

Real-time MIDI entry, which allows you to simply play (as if into a sequencer), is not a trivial programming feat. You should worship the software's programmers if the program permits it. From a programmer's standpoint, real-time entry is easiest if the software supplies a metronome click to play to.

Step-time MIDI entry refers to specifying a note duration by way of computer keyboard or mouse, then playing the note or chord on a MIDI keyboard to specify pitch. One feature to look for with step-time entry is the ability to "lock in" a duration. With this feature, if you need to enter 100 sixteenth notes in a row, you don't have to manually specify the duration before you play each note.



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LIST OF MANUFACTURERS

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or (708) 524-9441

Cerl Sound Group
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252 ERL
103 S. Mathews
Urbana, IL 61801-2977
tel. (217) 244-4298

C-Lab
PO Box 750
Nevada City, CA 95959
tel. (916) 265-6484
or (415) 738-1633 (tech line)

Coda Music Software
1401 E. 79th St.
Minneapolis, MN 55425-1126
tel. (612) 854-1288

Dr. T's Music Software
100 Crescent Rd., Suite 1B
Needham, MA 02194
tel. (617) 455-1454

Electronic Arts
1820 Gateway Dr.
San Mateo, CA 94404
tel. (415) 571-7171

Erato Software
PO Box 526278
Salt Lake City, UT 84152-6278
tel. (801) 328-0500

Grandmaster, Inc.
PO Box 2567
Spokane, WA 99220-2567
tel. (509) 747-6773

Great Wave Software
5353 Scotts Valley Dr.
Scotts Valley, CA 95066
tel. (408) 438-1990

Hybrid Arts, Inc.
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Culver City, CA 90232
tel. (213) 841-0340

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Cambridge, MA 02142
tel. (617) 576-2760

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Santa Cruz, CA 95062
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Bellevue, WA 98005
tel. (800) 426-2673
or (206) 462-1007

Theme Software Company
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Charlottesville, VA 22906
tel. (804) 971-5963

thoughtprocessors
584 Bergen St.
Brooklyn, NY 11238
tel. (718) 857-2860

Importing MIDI files is an important feature for a notation program if you are involved in MIDI sequencing. You may have hundreds of completed sequences you want printed in notation, and a program's ability to import Standard MIDI Files and convert them to notation saves endless data reentry.

Most non-MIDI input strategies are some variation on the theme of selecting a rhythmic value from an onscreen palette or specifying a duration with the computer keyboard and then clicking the note at its desired location with a mouse. Some systems turn the computer keyboard into a veritable musical keyboard by remapping the keys, while others position notes with cursor arrows instead of a mouse.

TEXT INPUT

The degree of *lyric* input flexibility you require involves determining how many verses you normally need and whether fonts should be mixed (most foreign language translations are in italics beneath the non-italicized original lyric). Another important feature is the ability to "bind" syllables to notes, so reformatting a system of music automatically reformats the lyrics as well.

Annotative text refers to just about anything besides lyrics. A good example is stage directions in an operatic work. Make sure it is possible to "attach" text to particular measures, staves, or systems, so that reformatting your music moves the annotative text along with the specific music it relates to.

Other text includes such automatically placed items as headers, footers, and page numbers. These can be added after printout, but their omission from the software is a good indication of a lazy programmer and should make you suspicious about other aspects of the program.

POSTSCRIPT

An important consideration is whether the notation software prints PostScript. PostScript, a page-description language developed by Adobe Systems, offers many advantages. The main one is printer independence. Any printer that understands PostScript can print your files at its highest resolution. Other advantages include a wealth of PostScript music fonts (32 and counting) and the capability to scale output to any possible reduction or enlargement. TrueType is a new type system developed by Apple and Microsoft that offers many of the same benefits for both Macintosh and IBM computers. Some PC programs offer high resolution output on non-PostScript laser printers.

TRANSPOSITION

The option to transpose music is important for two reasons. First, you may wish to print out music in a different key to accommodate the vocal range of a singer. Second, some instances of repeated musical phrases within a piece may warrant a new transposition. There are many ways material can be transposed: chromatically, diatonically, enharmonically, modally, or by user-defined scale. Such custom pitch-remapping allows, among other things, rapid changing of a passage from major to minor or Lydian mode.

An easy way to check if the software transposes correctly is to move a passage from the key of C to C#. The third and seventh degrees of the scale should renotate at B# and E# accordingly, *not* C and F natural.

PART EXTRACTION

If you need orchestral parts on a regular basis, examine part-extraction capabilities carefully. The best implementation is for the software to create separate parts files automatically. This allows you to go in and tweak them if necessary (to assure page turns are humanly possible, etc.). Some programs create parts directly from the score file without allowing any control over their formatting. This is fine if the program does it correctly, but most don't. Instrumental parts usually "concatenate" groups of rests into one long multi-measure rest with a large number over it. The software should be intelligent enough to break these multi-measure rests at tempo, meter, and key changes, as well as other places you indicate in the master score.

Two useful features are the automatic placement of the instrument's name in the upper left corner of the part (where it belongs) and the ability to auto-reformat parts as they are output, whether to a file or directly to the printer. The notes in instrumental parts are expected to be tightly spaced to require as few page turns as possible. The program should take care of this for you.

MIDI CHANNELS PER STAVE

The ability to assign separate MIDI channels to staves—and even separate voices on the same stave—turns your notation software into a veritable MIDI sequencer and facilitates proofhearing your masterpiece. The option to initialize each staff with a patch change message saves a great deal of setup time during playback of notation files. Having the music scroll by during playback often is regarded as a courtesy feature, but it actually is quite important. When listening to a piece of music without a computer, you probably flip score pages as the playback progresses. It helps when the computer does this for you onscreen, because it never gets lost.

MIDI EXPRESSION MARKS

Programs with the most sophisticated MIDI implementation include the ability to assign MIDI messages that affect playback to expression marks. For example, a ritard on the page can slow down the rate of MIDI clocks, or a forte can affect the velocity level of notes being played. Those who want an accurate playback of what they write will find this feature invaluable.

MIDI EDITING

If you intend to use your notation software as a sequencer (and not just for simple proofhearing), it should provide some of the MIDI data-editing features offered by dedicated MIDI sequencers. Don't assume that because a program allows MIDI input it retains velocity, controller, or tempo data. Most limit MIDI data capture to pitch. Others capture all associated MIDI data but don't allow editing.

EXPORT MIDI FILES

Importing MIDI Files is important as a data-entry method. However, if you want to share your work with someone owning a different notation program, or you want to do more work with the MIDI data in your favorite sequencer, the ability to export MIDI Files is essential.

THE BOTTOM LINE

Music publishers often invest a five-figure sum in the engraving, printing, and distribution of each musical composition they publish. Many current notation packages are well on their way to eliminating these expenses. Soon publishers may stop weighing the odds about the chances of recovering their investment in a publication: They can store all works upon a disk and print them as the need and orders arise. Furthermore, it is increasingly common for a composer to supply music already notated on disk, which represents further savings for publishers.

As more notation files of musical works are distributed via telecommunications, the long-awaited concept of "publication on demand" may become a reality.

Christopher Yavelow *is an award-winning computer-assisted composer, author, and consultant living in Hollywood. His book,* The Macworld Macintosh Music and Sound Bible, *will be published by IDG Books Worldwide this fall.*

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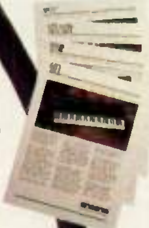

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THE TECHNOLOGY THAT PERFORMS

By Dan Phillips

Fundamentals of MIDI Mixing

How to keep your sequencer gainfully employed.

TROY THOMAS

MIDI sequencing has spoiled me. I can't imagine not being able to tweak any aspect of a musical phrase until it's just right. But when I'm ready to mix, that level of control disappears. With my easy-to-use-but-utterly-unautomated analog mixer, it's back to doing it all by hand and hoping it works.

That need not be the case.

MIDI can control levels, pan, EQ, and effects as well as notes and wheel movements. With a sequencer's editing and overdubbing capabilities, difficult fader moves may be recorded and edited to perfection. You can lavish attention on each track, recording one or two faders at a time. In performance, mix automation can bring the sophistication of the studio—auto-panning, dynamic effects, and so on—to the stage.

THE BASICS

There are two ways to mix with MIDI. The first uses MIDI Continuous Controllers to change parameters such as volume and panning of electronic instruments. All the mixer does is combine the preconditioned output signals. I call this "pure" MIDI mixing.

The second way is to use MIDI to control external hardware, a method I call "MIDI-controlled" mixing. MIDI-controllable mixers start at about $600, offering anything from simple channel muting to full automation of every function. You can add gain control to an existing mixer for similar cost. Some effects processors also offer one or two channels of gain control, as discussed later.

Each method has advantages and disadvantages. Best of all, you can use them together; they aren't mutually exclusive. Pure MIDI mixing works with any MIDI instrument that receives velocity or MIDI Volume (Controller 7). With a few of these instruments and a sequencer, you have what you need to get started in MIDI mixing. On the other hand, MIDI Volume has no effect on guitarists and vocalists. (If anyone has a MIDI version of these, let me know.) MIDI mixing with tape tracks and non-MIDI instruments requires a MIDI-controlled mixer or gain-control unit.

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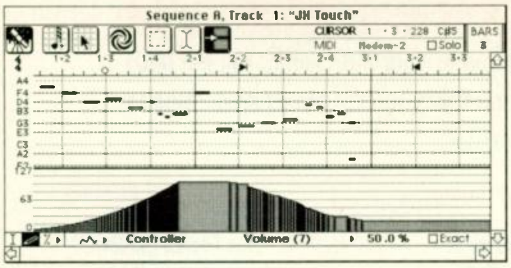

FIG. 1: Sequencers such as Opcode's *Vision* let you enter and edit controller data by drawing in graphic curves. This screen shows a smooth fade-in and fade-out of MIDI Volume.

"PURE" MIDI MIXING

If you're familiar with only one Continuous Controller, it should be 7. That's MIDI Volume, the most important ingredient of pure MIDI mixing. It works in conjunction with your synth's master volume control, and you can use it to create smooth crescendos and decrescendos, as shown in **Fig. 1**.

MIDI Volume control is especially important for multitimbral synths. These units may combine eight or more "instruments" into two or four outputs. Most of these instruments respond to MIDI Volume on each channel, so you can get inside the machine and do your mixing there.

But MIDI Volume has some drawbacks. For starters, it can diminish your signal-to-noise ratio. Most modern synths accomplish their internal volume settings digitally. In the digital domain, reducing overall volume lowers resolution. At one-fourth maximum volume, a 16-bit machine functions with only fourteen bits.

Lowering the volume of the synth means lowering the level going into the main mixer. This also reduces signal-to-noise, because the mixers input noise remains constant. These fidelity losses may or may not be significant for you. For production demos or jingle work, for example, these effects may not present any problems. If you are mixing a record, or otherwise concerned with maintaining pristine audio, you can solo each instrument and make your own judgments.

Some effects, such as compression and distortion, are sensitive to input level, so that changing the volume of an instrument going into them changes the timbre as well. For example, feeding a low-level signal to an overdrive section can turn a squealing Jimi Hendrix guitar into a whimper. The solution is to modulate the output of the effect, as opposed to the volume of the instrument. In most synths, MIDI Volume changes the level of the sound before it reaches the onboard effects. If your instrument does not let you control post-effect level, you might have to perform that volume change by hand. (For other types of volume control problems, see the sidebar "MIDI Compression.")

PANNING AND EFFECTS

Level is an important part of mixing, but not the only one—and it isn't the limit of MIDI's possibilities, either. It is, however, the only mixing-oriented controller with anything close to universal implementation. For other mixing functions, you'll need to dig into the specifics of your synths.

The MIDI spec defines Controller 10 as Pan. If your instruments respond to this controller, you're all set; 0 is full left, and 127 is full right (see **Fig. 2**). Many synths don't respond automatically to MIDI Pan, but don't let that stop you. If your synth lets you modulate pan, simply assign it a MIDI source such as Mod Wheel. (If you can actually set it to be modulated by Controller 10, so much the better.) Record this into your sequencer to automate dynamic panning, or to set up different positions for the instruments in a multitimbral module.

If your synth offers a pan parameter but no official modulation path, there's another possibility: System Exclusive. Many synths send parameter changes as SysEx messages. Most sequencers record these as they would any other MIDI event. (Some sequencers can even edit this data, but you may need to study the manual to determine which byte is the actual value.) This is a good trick to keep in mind, since it lets you use MIDI to do almost anything you can do from the front panel.

Equalizing is another task you may want to do on the synth itself. Outboard EQs are useful for fine-tuning an electronic timbre, but the onboard filters found on most synths are sufficient for many tasks. If your synth has basic lowpass filters, for instance, you can adjust the brightness of the sound to fit in with the rest of the mix. If it has complex, multi-mode

MIDI COMPRESSION

For signals with wildly varying levels, a simple volume control isn't an adequate solution. If your synth program pops in and out of the mix no matter where its level is set,try using compression.

An analog compressor operates on signal level, but in MIDI you can deal with MIDI Volume or with velocity. (Be sure that the synth patch uses velocity to control level only. If it affects timbre, this trick could get you in trouble.)

Use your sequencer's scaling features to change velocity or volume of the offending track by 50%. This cuts the unevenness by half, compressing by a ratio of 2:1, but it makes the notes softer as well. To bring them back up to level, add a constant value. This brings the volume of the program back up, while preserving the compression (see Fig. 4). If your sequencer doesn't let you scale by a percentage, try limiting the maximum and minimum velocities.

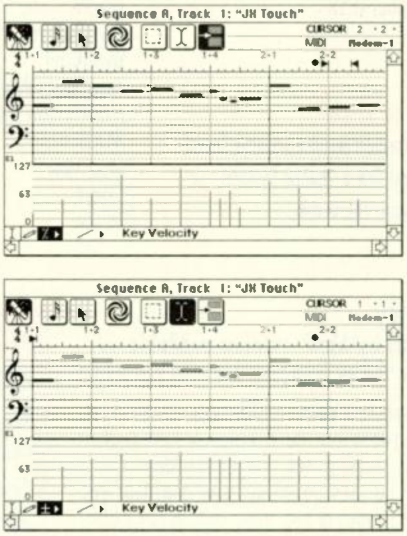

FIG. 4: The top window shows pitches and durations of a musical phrase in piano-roll style. The bottom window shows compressed velocities from the same phrase in bar graph form.

● MIDI MIXING

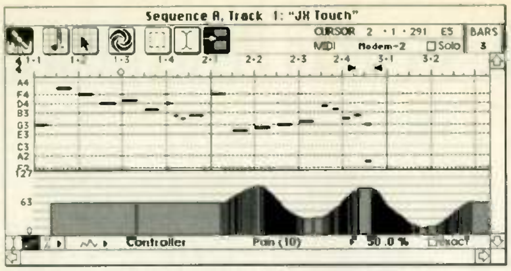

FIG. 2: In this example, the panning is stable for the first measure and then goes into a stereo tremolo effect. Graphic editing made this gesture easy to enter.

filters, more subtle effects are possible. Highpass filtering can remove body to make more space in the mix, for example. Highpass and lowpass filtering can combine to create a bandpass filter.

Many effects processors (both built-in and outboard) let you modulate one or more parameters in real time with MIDI controllers. You can use your sequencer to increase reverb at the end of a chorus or add delay for just the first few notes of the bridge (see "The Magic of Real-Time MIDI Control" in the August 1990 **EM**).

MIDI Controller 12 has been designated Effects Controller, but this new development is not implemented on most instruments. It is purposefully generic; the spec doesn't say which parameter (effect level? reverb time? chorus depth?) it should control. For now, assign controllers as described for panning.

Effects can be recalcitrant about real-time modulation. If there isn't an official way to control reverb time, perhaps the effects DSP can't do this in real time. Effects often "glitch" when their parameters are modulated, so pay attention to what's happening in the audio.

MIDI-CONTROLLED MIXING

"Pure" MIDI mixing is great, but it doesn't apply to recorded audio or non-MIDI instruments. It's still possible to control these, if you have a MIDI-controlled mixer. These are actual audio mixers that accept control of some or all of their functions via MIDI. Automatable parameters may include track-muting, volume, or all settings, including EQ and effects bussing.

MIDI-controlled mixers use "modulatable" gain blocks to control the level of incoming signals. Most designers use Voltage-Controlled Amplifiers (VCAs), some favor multiplying DACs, and one company (Yamaha, in their DMP7 and DMP11 mixers) actually digitizes analog audio and controls it in the digital domain. To change the volume of a track, the mixer sends a message to the gain module instead of moving a physical fader. Some companies sell banks of gain modules without the rest of the mixer. You can use these to retrofit MIDI control into existing boards.

MIDI-controlled mixing comes in two styles. *Snapshot automation* memorizes settings as a preset (or "scene"), like a program in a synth. These snapshots can be recalled by MIDI Program Changes, so that you can quickly reconfigure a board for different songs, or song sections. Most units include a crossfade time for each snapshot, so that the settings change from one configuration to another over some period of time.

More sophisticated boards combine snapshot automation with real-time control. Different MIDI controllers (or other MIDI messages) affect each parameter, for independent control. Some systems also let you match a parameter to a MIDI note number, with note velocity as the control signal. Even a small mixing board can have more than 128 variable parameters, and note velocity provides a convenient way to extend the number of controls.

You can use any multi-effect that has volume as a programmable parameter for automated mixing of one or two channels. Just create programs that are identical except for level, and switch between them. If volume is a modulatable parameter, so much the better; you can control it via MIDI to mix a single, complicated track, while you do the rest by hand.

Sequencers let you realize parts that you might never be able to play correctly—or even think of playing at all. Likewise, MIDI-controlled mixing can bring new ideas into your studio. *Transforming* is a popular volume-modulation effect that can turn a sustained pad into a catchy rhythmic pulse. The signal is repeatedly pulsed on and off, muted and unmuted, in time with the music. You can achieve this effect by writing percussive MIDI Volume

(continued on p. 116)

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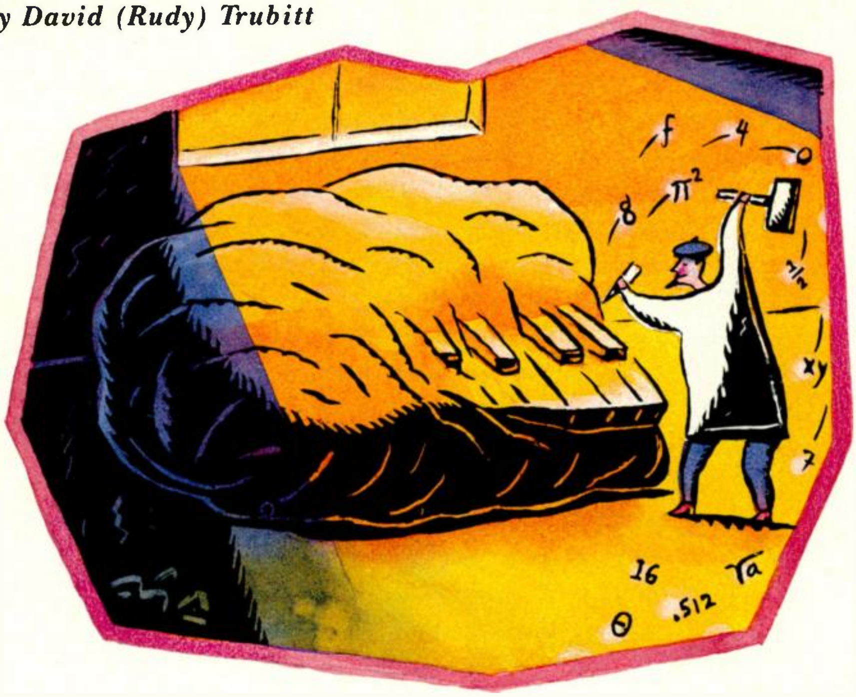
Physical Modeling

By David (Rudy) Trubitt



Advanced synthesis techniques place real instruments (in the form of mathematical models) inside the computer.

Few would deny the advantages of samplers and sample-playback instruments. Traditional synthesizers rarely reproduce an acoustic instrument as convincingly as a sample of the actual sound.

But sooner or later, the advantage becomes a limitation. Even with performance controls to add expression, the soul of the sound remains static. Samples with different articulations can help, but they still fail to provide insight into the nature of the instrument. *Piano* and *forte* horn samples won't get you a *mezzo* toot, no matter how you crossfade the two.

What's missing is a link to the physical characteristics of the original instrument. *Physical modeling* uses a mathematical description to bridge the gap between the sound of an instrument and the process by which that sound is produced. The idea of physical modeling is not new (related work dates back to the 1960s), but realizing it requires a massive amount of computation. In the past, this has been a major obstacle, but new DSP chips have made the needed horsepower available at reasonable cost.

To get an idea of the state of physical modeling, I paid a visit to Stanford University's Center for Computer Research in Music and Acoustics (CCRMA, pronounced "karma"), the birthplace of FM synthesis. CCRMA's physical modeling derives from waveguide synthesis, developed by Julius Smith.

A waveguide is a tube, much longer than it is wide, down which a wave travels. The tube resonates at frequencies related to its dimensions. If the waveguide bends, changes diameter, or intersects another waveguide, additional resonances are formed.

Waveguide synthesis uses mathematical models of waveguides and oscillators, which have behaviors consistent with that of their real-world counterparts. "Real" instruments can be simulated by a computer model of interconnected waveguides and oscillators.

Perry Cook, a CCRMA research associate, explains: "A clarinet, to a first approximation, is a cylindrical bore with a non-linear oscillator (the reed). If you don't mess it up with evil things like tone holes, you can actually do a pretty cheap (in terms of computation time) simulation of a clarinet."

Cheap compared to more elaborate waveguide models, that is. Despite advances in DSP performance, waveguide synthesis is rarely done in real time. Complex models require anywhere from two to ten minutes of high-speed computation to produce a single minute of music. Chips keep getting faster, though; what's in the lab today could be in your studio tomorrow or the day after.

CCRMA's researchers do most of their work on NeXT computers. Developers favor the NeXT for DSP work because of its built-in Motorola 56001, high-resolution graphic interface, and object-oriented environment for software development.

A unique aspect of CCRMA's work is the development of a graphic "workbench" for experimentation with physical models. **Fig. 1** shows the main screen of Perry Cook's SPASM (Singing Physical Articulatory Synthesis Model) program, which simulates the tone and articulation of a human singer.

An oscillator models the vocal chords (glottis) and provides a basic vocal waveform. Eight individual waveguide

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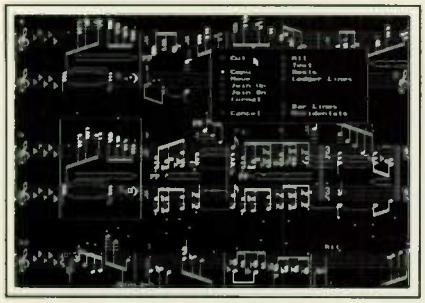

Main Score Editing Screen

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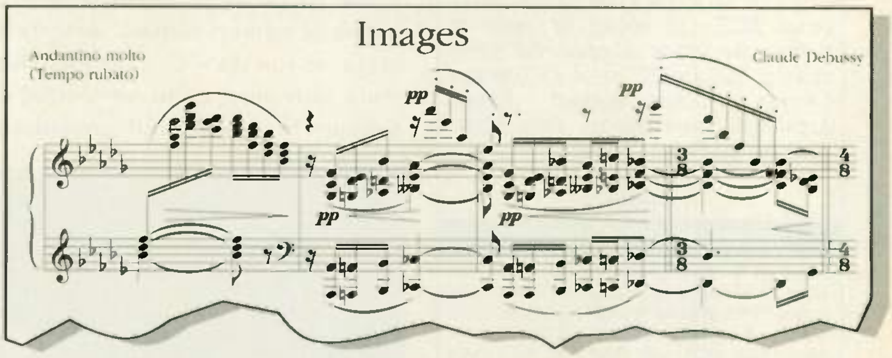

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chambers combine to model the complexities of the human vocal cavity. Each waveguide's diameter changes to create the basic vowel sounds, while a movable noise source creates sibilance and consonants. With the model in place, animating the vocal tract creates speech or song. SPASM sings with a classical style and can produce a wide range of vocal tones and articulations.

Perry Cook has created another workbench for brass instruments, which are well-suited to the waveguide model. "A brass instrument is the prime case of a waveguide," Cook explains, "because large parts of it are cylindrical, and it's extremely long compared to its width. To change notes you just add hunks of tubing, like patching in a new piece of waveguide."

A special oscillator (mass-spring-damper type) models lip position as a function of differential pressure. The model can simulate a variety of trills and falls by varying the tension of the lip. The timbres suggest the full range of the brass family, from tuba to piccolo trumpet.

Cook cites the advantages of waveguide synthesis: "You don't need a bunch of memory to stash these models in, as you do with a sampler. You really only need about one period of memory, because that's the round-trip delay time within the instrument. What is the memory of a trombone? It's the length of it."

CCRMA's technical director, Chris Chafe, also is a cellist, and he has devoted considerable energy to the modeling of bowed string instruments. The better your understanding of the instrument, the better your chance of making a convincing model.

Chafe describes his research with physicist Bob Schumacher: "Bob knew from studies he'd done that what distinguishes a real vibrating system from current synthesizers is unpredictability at a very micro-time level. Given a straight note, you get changes in the waveshape from one period to the next." It turns out that these irregularities are not random. Under analysis, their frequency turns out to be a subharmonic of the fundamental pitch.

"In a bowed string, the string alternates between sticking to the bow and flying back. A noise happens when the string is flying back. The fact that you're pulsing noise into a highly tuned resonator means that the previous noise reverberates and affects the noise that happens a few periods in the future. That's where the subharmonics come from." Subharmonics also exist in wind instruments, where higher pitches resonate at fractions of the instrument's total length, but still have

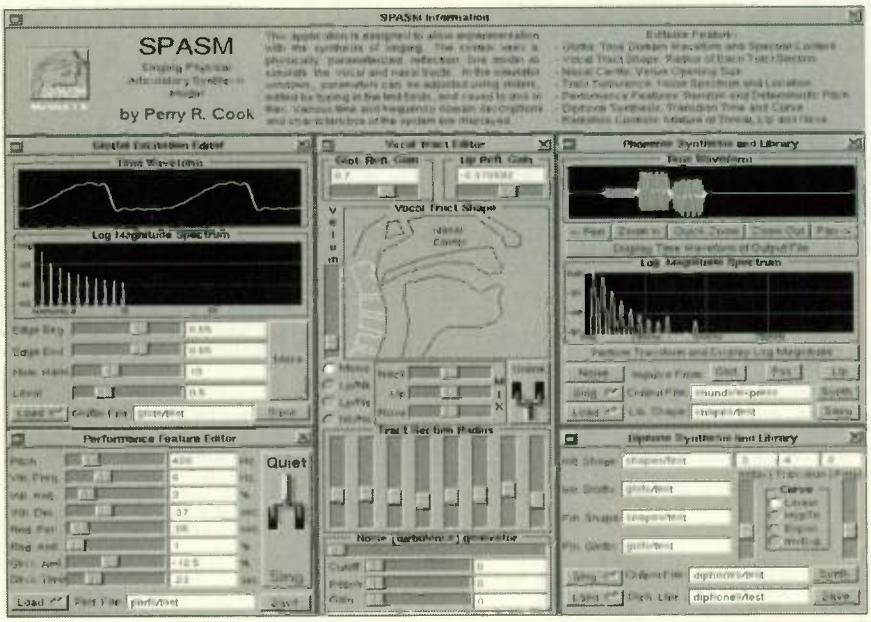

FIG. 1: The NeXT-based SPASM program typifies the "workbench" approach developed at CCRMA. It provides convenient testing of parameters in a complex model of a physical process.

to traverse the full length of the horn.

Adding synchronized noise improved the realism of the model. "Pulsing the noise during the slipping part of the bow sounds right and gives you subharmonics," Chafe continues. "You also don't need an extra control signal for the noise, like a finger on a 'noise' wheel, because it's there as part of the physical model."

Chafe tested his cello model with a familiar ritual, tuning up. "The hardest thing in the world was to make this thing stay in tune," he explains. "Every time I had a bug in my program, it would express itself as being slightly out of tune."

Lack of real-time control hampers the process of refining the model, one reason that CCRMA researchers favor the workbench approach. "Let's say I did a more complete cello," Chafe explains, "and wanted to hear how it responds to a different amount of rosin. It would be very tedious to run that experiment without a knob that said 'rosin coefficient' to turn until you got it right. It's all very intuitive when listening, but poking numbers into equations is not an easy way to learn."

Chafe is optimistic about the future of physical modeling, and he expects commercial applications of the technology to appear soon. "It's technically feasible now, and we'll see this stuff brought to market in short order, I'm sure. The benefits are pretty obvious to players who like to get expressive control. As of yet, no one has finished a piece of music using this approach. It's all kind of speculative, a lot of test tones. I think we're just at the point where there will start to be some interesting music made."

For further reading, start with "On the Oscillations of Musical Instruments," by McIntyre, Schumacher and Woodhouse, in the *Journal of the Acoustical Society of America*, November 1983, and "Musical Applications of Digital Waveguides," by Julius O. Smith, Stanford University Department of Music Report, STAN-M-39, 1987.

David (Rudy) Trubitt *looks forward to the availability of ever more expressive electronic instruments.*

PART ONE

VECTOR

PROGRAMMING WORKSHOP

Practical programming tips for Korg's Wavestation.

Not long ago, you could mention the word "vector" to the typical electronic musician and watch his or her eyes glaze over. "You mean like Vector Hugo? Or Vector Mature?" Things are a bit different today. Vector synthesis is an established method of sound production, supported with products from major manufacturers.

This two-part article explains how to program vector synthesizers. Part 1 examines the capabilities of Korg's Wavestation and its rack-mounted cousin, the Wavestation A/D. In Part 2, you'll learn how to program Yamaha's SY22 and its keyboardless relative, the TG33.

ORIGINS AND DEFINITIONS

Sequential Circuits introduced the first vector synthesizer, the Prophet VS, in 1986. According to Sequential founder and president Dave Smith, the idea was to build a synthesizer with as much sonic animation as possible. Sequential closed its doors in 1988, and Smith, Sequential co-founder John Bowen, and a group of other engineers eventually became a West Coast research and development center for Korg. The first product they introduced was the Korg Wavestation, which reached the market in mid-1990. Both the Wavestation and the recently released Wavestation A/D (which essentially is identical to the keyboard but adds analog inputs for vocoder functions and processing of external signals) are based on the vector synthesis concepts the group developed while creating the VS.

BY HOWARD MASSEY

FIG. 1: The Wavestation's joystick controls the balance of as many as four sound elements.

Webster's Dictionary defines a vector as "a quantity having both magnitude and direction." In other words, a vector is some kind of quantifiable movement. A vector can be described by how far it has strayed from its previous position (its magnitude) and by the direction of that movement. This description will make more sense if I point to a practical application.

Consider a volume slider. When you move it in one direction, you hear more audio signal. When you move it the other direction, you hear less. As you move the slider, you are in fact creating a vector, because you can specify both the direction of the movement (up or down) and the magnitude of the movement (that is, how much the slider has moved from its previous position).

Substitute a balance control for the volume slider. Moving it in one direction increases the level of one sound while decreasing the level of another. Placing it squarely in the middle gives an equal balance of both sounds. The movement of the slider determines the relative levels of the two sounds.

Suppose you want to blend four sounds, not just two. You need a 2-axis controller to specify the relative balance of sounds. The solution that the Sequential (and later Korg) engineers found is a joystick. Each position of the joystick corresponds to an exact blend of the four component sounds. **Fig. 1** shows the Wavestation's joystick controller. Each joystick "compass point" corresponds to one sound element.

The programmer uses the joystick both to define an envelope that determines the blend of sounds over time and for realtime mixing of timbres. In Yamaha vector synths, the vector control also can alter relative pitch. Thus, you can create a "detune" envelope or use detuning as a performance control.

PATCHES AND PERFORMANCES

The key to the Wavestation is its data organization. When you select different sounds on the instrument, you are actually playing what Korg calls Performances. **Fig. 2** shows the structure of a Wavestation Performance. When you press the EDIT soft key, you go to the EDIT PERFORMANCE screen (**Fig. 3a**), which displays the component Parts of the currently selected Performance.

A Performance can have as many as eight Parts. Each Part consists of a Patch with several defining parameters, accessed by pressing the DETAIL soft

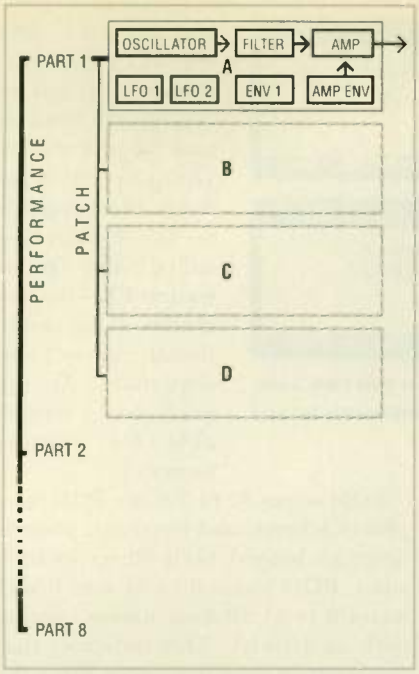

FIG. 2: The Wavestation architecture groups oscillators (actually a complete synth voice) into Patches. Patches are combined into Performances. Performances can be further combined into a Multiset for sequencing.

key from EDIT PERFORMANCE. From the PERFORMANCE PART DETAIL screen (**Fig. 3b**), the user can layer or split Patches across the keyboard, with individual settings for transposition, keyboard mode (polyphonic or monophonic), routing to onboard effects processors, and so on.

Each Patch has one, two, or four Voices, each with an oscillator, filter, and amplifier, two envelopes, and two LFOs. The Wavestation performs vector synthesis by dynamically controlling the relative levels of these Voices. With up to 32 Voices and eight vectors, a single Performance can produce a huge amount of timbral complexity.

Above the Performance level, the Wavestation has a single Multiset of up to sixteen Performances. This lets the instrument serve as a multitimbral tone generator with an external sequencer or controller.

Finally, the Wavestation includes two onboard effects processors. Choices include various reverb, delay, equalization, panning, flanging, phasing, pitch shifting, and distortion effects. A flexible bus structure lets individual Parts feed as many as four separate effects.

If you select a two- or four-oscillator Voice structure, the joystick lets you create a Mix Envelope that dynamically blends the oscillators over time. In addition, you can use the joystick to change timbre in real time. A brief editing session will show you how to construct Wavestation Performances and the role that vectors play.

EDITING

Like most synths, the Wavestation uses a temporary buffer for editing. When you edit a Performance, you work with a *copy* of the data, so you can edit without fear. (The Wavestation handles wave sequences differently. When you edit a wave sequence, you write directly to memory.)

Select any Performance and go to EDIT PERFORMANCE. The screen shows a listing of Patches used. (Most Performances contain at least two.) Using the cursor keys, go to the PATCH field for Parts 2 through 8 and

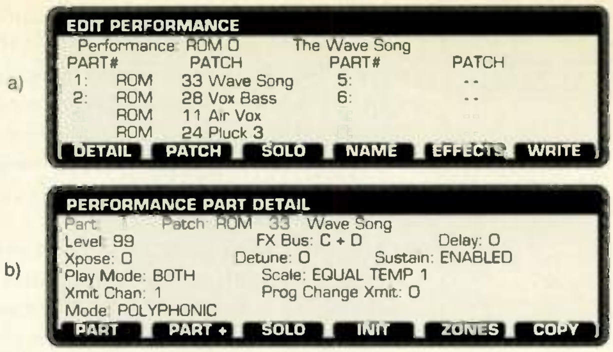

FIG. 3: A Wavestation Performance consists of from one to eight Parts. Level, transposition, keyboard zone, and so forth, are set individually for each Part.

including 64 wave sequences. The Wavestation A/D has an additional 119 waves and 32 sequences. (Korg recently announced an upgrade to the keyboard—called the Wavestation EX—that includes these additional waves and sequences. An upgrade board is available for existing owners.)

null each Part (Patch name "—"), using the data entry wheel or keypad.

Position the cursor over Part 1 and press the PATCH soft key, which brings you to the EDIT PATCH screen (**Fig. 4a**). Press the soft key labeled INIT to set all parameters to their basic values. Answer "YES" when the system asks if you are sure.

After initialization, the system returns to EDIT PATCH. The screen now shows a single-oscillator structure. You plan to do vector programming, so change the structure to four oscillators (leave Hard Sync off).

Bypass the effects section for now. It's usually best to add effects *after* building the basic sound. To disable effects, return to EDIT PERFORMANCE by hitting EXIT and pressing the EFFECTS soft key. Set both FX1 and FX2 to "No Effect."

LISTEN TO THE WAVES

Return to EDIT PERFORMANCE, and press the PATCH soft key, followed by WAVES. In the WAVES screen (**Fig. 4b**), you can see that the initialized Patch assigns ROM wave 32 (Soft EP) to all oscillators at the same tuning and at maximum level.

You can assign different waves to each oscillator (as well as alter the tuning and overall levels of each oscillator). Here's your chance to listen to each wave in its "raw," unprocessed glory, using the soft keys to mute the other oscillators. The Wavestation has a total of 428 waves,

ROM waves 32 to 396 are PCM samples of acoustic and electronic sounds. Some are looped, while others are transient. ROM waves 0 to 31 and RAM1 waves 0 to 31 all have names starting with an asterisk. This indicates that they contain wave sequences. From the factory, all RAM2 waves contain blank wave sequences, and these yield no sound until edited.

When you finish auditioning the waves and wave sequences, return the oscillator to its initial default, Soft EP (ROM 32). Unmute the other oscillators. For the moment, leave them playing Soft EP, but set each oscillator to a different pitch. This will illustrate how vectors work.

Using the cursor keys and data-entry controls, change the value of Semi for oscillators B, C, and D. Set them to +4, +7, and +12 respectively. This gives a major triad when you play a note.

EDITING THE MIX ENVELOPE

Next, construct a vector that arpeggiates this triad. Press the MIXEV soft key on the WAVES screen to access the EDIT MIX ENVELOPE display (**Fig. 5a**).

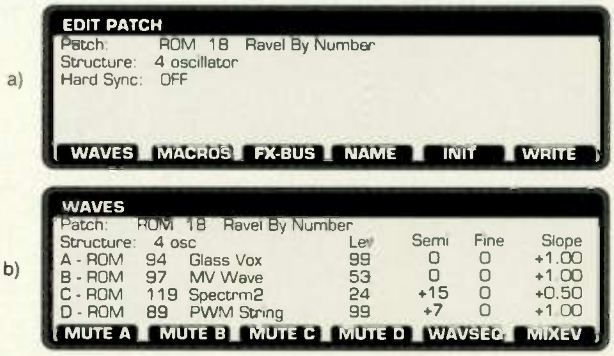

FIG. 4: A Patch uses from one to four oscillators, each of which uses a wave or wave sequence. The Slope parameter stretches or contracts intonation across the keyboard.

This screen is unique in that movement of the joystick actually changes the onscreen graphic (remember Etch-A-Sketch?).

Start with point 0. (Press the POINT soft key to change the current point.) Move the joystick to the A (9 o'clock) position. The little black square on the screen faithfully moves over, and the relative levels of A, B, C, and D oscillators change to 100%, 0%, 0%, and 0%, respectively.

Press the POINT key again to increment to point 1. This time, move the joystick to the B (12 o'clock) position. The relative levels change to 0%, 100%, 0%, and 0%. Set point 2 to the C (3 o'clock) position, point 3 to the D (6 o'clock) position, and, finally, point 4 back at A.

If you do this correctly, you'll have a diamond-shaped Mix Envelope. Now, set the Loop parameter to "0<->3" and leave Repts (for Repeats) at the default setting of "INF" (for infinite). Now play a note. What you hear is the Mix Envelope cycling around its first four points. Because the oscillators are playing the same wave at different pitches, you hear an arpeggio.

It's time to liven things up. Exit to the WAVES display and experiment with different waveforms for the four oscillators. When you find a timbral pattern you like, readjust values of Semi to unison, or select something more captivating than a simple major triad.

Notice that transient waves seldom work in a looped vector. Usually, they die by the second or third time through the loop.

Return to EDIT MIX ENVELOPE and adjust the time values for the various points. This alters the rhythm of arpeggiation. You also can use the joystick to readjust any or all points of the Mix Envelope. With just these two screens, you can develop unique and useful sounds quickly.

VECTOR MODULATION

To take things a step further, go to the EDIT MIX MOD screen (**Fig. 5b**) by pressing the MIXMOD soft key. Here you can select up to four modulation sources to apply to each axis of the vector. Modulation sources include note number, velocity, aftertouch, LFOs, envelopes, and MIDI Continuous Controllers.

The X-axis corresponds to left-right (A-C) motion, and the Y-axis is up-down (B-D). Each axis can have one or two modulation sources, and you can adjust the scaling for each. With little effort, you can establish complex and expressive patterns of vector motion.

At this point, you might want to write your new Performance and Patch to memory. Note that on the Wavestation, saving a Performance does *not* save any edited Patches in that Performance. You must save the altered Patches separately from the EDIT PATCH screen.

This may seem obvious, but the time will come when you tweak a Performance to perfection and then forget to save the edited Patches with it. When you recall that Performance, it will sound different because it uses the stored versions of the Patches. *Caveat programmer.*

WAVE SEQUENCING

Even without the vectors, the Wavestation can produce complex timbral changes or entire musical passages with the touch of a single note. Examples include ROM Performances a)

b)

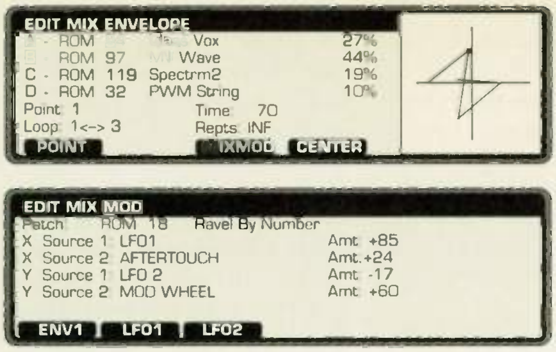

FIG. 5: The Mix Envelope is a two-dimensional envelope with four stages. Each point of the envelope corresponds to a particular blend of the four oscillators.

49 (Station Platform) and 0 (The Wave Song). The key to these Performances—indeed to many of the Wavestation's best sounds—lies in wave sequencing.

Wave sequences let the user produce long periods of arbitrarily changing sound. You can think of a wave sequence as an envelope of many segments, each with a different sound.

The wave sequence is a list that tells the Wavestation which of its onboard waves to play and in what order. For each wave, the user specifies duration in arbitrary units and pitch in semitones and cents. (In external sync mode, duration units become MIDI clocks.) Crossfade times also can be set, so that one wave changes into another slowly or rapidly.

In addition, a wave sequence can loop in either a forwards or forward-backward direction. You can modulate the starting point of the wave sequence with note number or velocity, and you can modulate the current position with aftertouch, LFO, envelope, or any MIDI Continuous Controller.

WAVE ANALYSIS

To understand how wave sequences work, analyze some of the existing ones, then create a few of your own. Let's look at the wave sequences that make StationPlatform (ROM 49) and The Wave Song (ROM 0) so effective.

Call up StationPlatform and press the EDIT soft key. The EDIT PERFORMANCE screen shows that this Performance uses two Patches, Vocalise (ROM 9) and WS Metal (ROM 6). Use the SOLO soft key to listen to each Patch. (Choose the Part to solo with the cursor up-down keys.) Your ears will tell you that Vocalise is slower and more subtle than WS Metal, so let's examine it first.

Place the cursor over Part 1 (Vocalise) and press the PATCH soft key to go to the EDIT PATCH screen. You can see that this Patch has only one oscillator. Press the WAVES soft key, and you'll discover that the single oscillator uses ROM wave 9 (*WSVoice).

Since *WSVoice is a wave sequence, a WAVESEQ soft key appears in the screen. (This doesn't happen unless at least one oscillator uses a wave sequence.) Press this key to go to the WAVE SEQUENCE screen (**Fig. 6a**), where you'll see a listing of all waves in this sequence.

Scroll through this list, using the cursor up-down keys. The *WSVoice sequence consists of four ROM waves ("OO" Vox, MV Wave, Voices, and Glass Vo), all with the same tuning and volume. The waves play in an infinite forward loop, so that the amplitude envelope determines the total length of the sound. The duration (Dur) of each wave is 46, and the crossfade time (Xfd) is the same. When crossfade time and duration are identical, the result is a smooth change from one timbre to the next.

Return to EDIT PERFORMANCE, position the cursor over Part 2 (WS Metal), and press the PATCH soft key. This Patch also uses only one oscillator. Press the WAVES soft key and you'll see that this oscillator uses ROM wave 6 (the wave sequence *WSMetal).

Now press WAVESEQ. You'll discover that *WSMetal uses nine waves (BellWave, followed by eight waves derived from the original Prophet VS). As with *WSVoice, all waves share the same tuning and volume level and play in an infinite forward loop. Note that the duration of each wave in this sequence is 24 and the crossfade time is 12. This accounts for the more rapid and abrupt change in timbre.

Pressing the UTILS soft key reveals another difference between this sequence and *WSVoice. The WAVE SEQUENCE UTILITIES screen (**Fig. 6b**)

a) b)

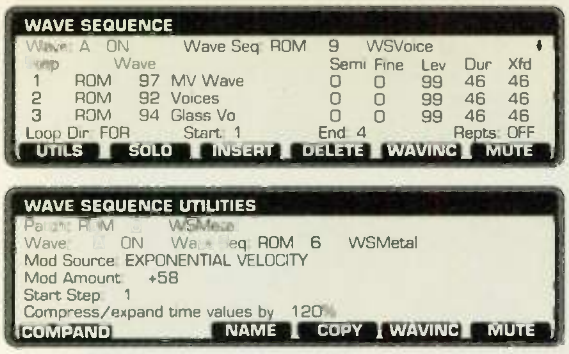

FIG. 6: Onscreen, the wave sequence is a list of individual waves with transposition, level, duration, and crossfade value for each. Utilities are provided for copying, stretching, and modulating the sequence in real time.

shows that key velocity modulates the start point of the sequence. This provides a way to dynamically alter timbre by starting the sound at a different point in its development.

THE SECOND WAVE

In contrast to StationPlatform, ROM Performance 0 (The Wave Song) shows off the rhythmic capabilities of wave sequencing. The EDIT PERFORMANCE screen reveals that the Performance has four Parts. By using the SOLO soft key, you can hear that the Wave Song Patch in Part 1 is responsible for the percussive quality of this sound.

Position the cursor over that Part and press the PATCH soft key. Again, only one oscillator is in use. Press the WAVES soft key to discover that this Patch uses the ROM wave sequence *WavSong. Press the WAVESEQ soft key to view the wave sequence. This sequence lists sixteen waves that play in an infinite forward loop, with all waves at maximum level and all except the last set to basic tuning.

Note that the crossfade time between each step is 0, and that most waves in this sequence are percussive transients. That's why this wave sequence sounds more like a sequence of notes. Unlike vectors, wave sequences retrigger playback of each wave every time around the loop.

SEQUENCE EDITING

By now, you're probably dying to edit these sequences or construct your own. If you've already attempted some edits, you have probably seen the stern admonition, "CANNOT WRITE TO ROM." As noted, there is no edit buffer for wave sequences; when you make a change, the Wavestation writes it to memory automatically. You can't write data to ROM, so you get an error message instead.

To alter a ROM wave sequence, you must copy it to a RAM slot. You can copy any sequence to any RAM location or copy specified steps of a sequence to any point in a RAM wave sequence from the WAVE SEQUENCE UTILITIES screen. This lets you assemble new wave sequences from components of existing ones. The RAM2 Bank comes with all blank wave sequences, so you have 32 empty slots to use for editing. When all the RAM2 slots are full, it's time to wheel out the old MIDI librarian.

Wave sequence editing is an easy task, similar to entering a sequence in single step mode. Using a combination of step-entry and cut-and-paste, you can create all sorts of new and useful wave sequences. As with any other type of creative work, the main thing is to dive in and gain some experience.

Remember that both modes of Wavestation synthesis—vectors and wave sequences—can interact. A single Patch contains up to four oscillators, each potentially with its own wave sequence. A vector can create shifting blends of wave sequences and sustaining or transient waves.

In addition, each oscillator has multiple envelopes and LFOs controlling discrete filters and amplifiers. The mind reels when you consider that a single Performance can use as many as eight Patches. The Wavestation's extensive digital effects section rounds out the picture of an instrument of extraordinary sonic power.

That does it for the Wavestation programming workshop. Our next installment (October 1991 issue) takes a close look at the Yamaha SY22 and TG33 vector synthesizers.

Howard Massey *is author of* Yamaha's Applications Guidebook for the SY22, The Complete DX7, *and* A Synthesist's Guide to Acoustic Instruments.

The Speaker of the House

Electing the right transducer achieves great live sound.

By Steve Oppenheimer and Michael Molenda

Like gravity, speakers often are taken for granted. We plug them into amplifiers and sound comes out. However, the demands of live sound reinforcement force us to listen to speakers with critical ears. Simply put, there is good sound and bad sound, and bad sound doesn't wow audiences. How speakers interact with the rest of your sound system is critical to clear projection of your music.

SPEAKER ENCLOSURES

An unmounted speaker, or *transducer*, produces a minimal amount of air motion. The front and rear surfaces of the speaker cone fight each other, impeding formation of a viable soundwave. For optimum performance, the speaker must be loaded into a resonating space—an enclosure—to produce higher output levels and predictable sound distribution (see **Fig. 1**). The size, shape, and style of the enclosure determine speaker efficiency.

There are a few basic types of speaker systems. Sealed, *acoustic suspension* (sometimes called "infinite baffle") speaker designs mount the transducer on an airtight cabinet. Air pressure, caused by the speaker's rearward motion, helps push the cone forward. Vented, or *bass-reflex*, systems use vents and/or ports cut into the enclosure to increase bass response. You achieve maximum volume and dispersion when the speaker is loaded into a horn-shaped enclosure (Fig. 1). This design enables the speaker to "throw" sound, as when one yells through cupped hands. However, louder is not always better. A poorly designed horn enclosure delivers uneven response and produces ragged low-to-mid frequencies even the best equalizer can't tame.

The average performer needs a *short-throw* system. Few clubs are large enough to demand Herculean projection from the speaker system. The wide dispersion of short-throw designs operates in much the same manner as home stereo speakers. This "near-field" (close-listening) approach to the sound-system environment works well in most situations.

However, if an act plays the "aircraft hangar" circuit (any large hall qualifies), a *long-throw* system helps tame reverberation and maintain vocal intelligibility. True long-throw systems comprise speaker enclosures with deep (typically fiberglass) horn chambers. Some models are three feet

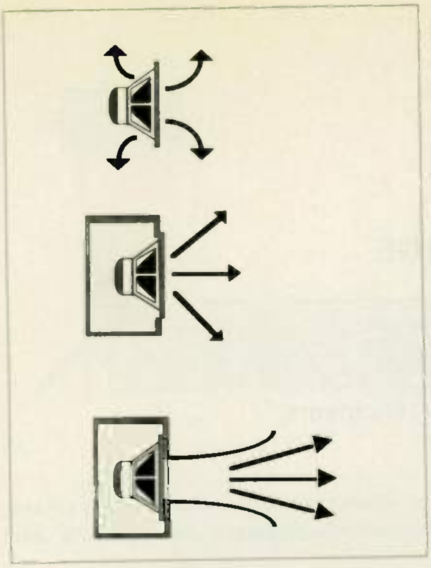

FIG. 1: Enclosures help determine speaker characteristics.

deep, with speakers loaded at the back for maximum projection. Horn-loaded systems are particularly effective in reverberant environments, with more than five seconds of decay in midrange frequencies.

Most large-scale touring systems are not horn-loaded. Modern live-sound concepts employ intelligent speaker engineering and specific array patterns to accomplish long-throw penetration with enclosures less bulky and sonically smoother than horn-loaded systems.

AUDIO IDEALS

Most speaker systems consist of two or more different-sized speakers designed to reproduce complementary frequency ranges. Many musicians define the traditional speaker system as cone (bass) and horn (treble) transducers packed into a single enclosure (**Fig. 2**). It's hard to go wrong with a pair of compact, nearly indestructible, "full-range" 2-way enclosures. This design is favored by musicians working small- to medium-sized venues. A typical 2-way club system has a bandwidth of approximately 60 Hz to 8 kHz (low bass tones to high sibilant frequencies).

In a 3-way system, tweeters usually handle frequencies above 6 kHz. (In a 2-way system, they may go as low as 1.5 kHz.) These high-frequency drivers may be cone-type speakers (usually between two and five inches in diameter) or compression drivers (with a diameter ranging from one to four inches). Frequencies between 500 Hz and 6 kHz are directed to a 5- to 12-inch cone driver or a 2- to 4-inch compression driver. (A few systems use midrange compression drivers as large as nine inches in diameter.) Low frequencies (below 500 Hz) are reproduced by woofers that range from eight to eighteen inches in diameter, although 15- and 18-inch woofers are the most common sizes.

Improving sound system fidelity requires expanding its working frequency range, or *bandwidth*. To move the system toward the "ideal" bandwidth of 20 Hz to 20 kHz (the approximate range of excellent human hearing), you need to add a *supertweeter* system for very high frequencies and a *subwoofer* system for very low frequencies. This costs money, makes the sound system more complex, and requires endless fine-tuning. It also requires more complex crossovers, more power amplifier channels, and more audio cables.

Supertweeters usually handle frequencies above 10 kHz, adding crispness to the sound and letting the tweeters operate within their most efficient range. Either compression drivers or piezoelectric drivers can be utilized as supertweeters.

Subwoofers (**Fig. 3**) handle extreme low frequencies, add sonic excitement without raising volume levels, and reduce the demands on the woofer.

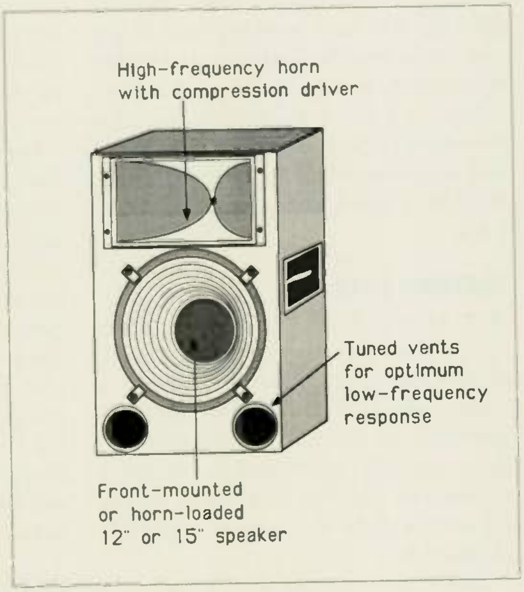

FIG. 2: A typical 2-way loudspeaker system.

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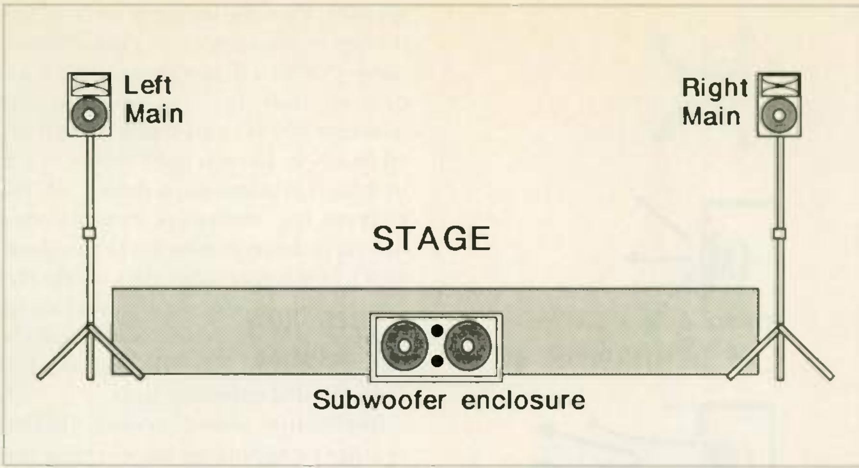


FIG. 3: Subwoofers reproduce extremely low frequencies, improving the sound system's efficiency and bass response. A selective mix of sound sources that contain mostly low-frequency program material is routed through a dedicated crossover and fed to the subs.

Added to a small club system, a beefy 18-inch speaker enclosure really rattles the walls. For most applications (excepting large venues), the subwoofer system can be mono, as low frequencies are essentially omnidirectional.

Subwoofers typically receive a selective mix (from a separate mono output on the mixer) of instruments that produce very low frequencies. This gives the engineer convenient, independent level control over the lows. It requires a narrow-bandwidth, frequency-dividing network (a crossover dedicated to the subwoofer feed), set at approximately 25 Hz to 80 Hz, depending on the type of speakers.

Quick Tip: Use a separate, dedicated graphic or parametric equalizer for the subwoofer feed (**Fig. 4**). Insert the EQ after the auxiliary send and before the special subwoofer crossover network. If possible, put a compressor/limiter inline and set a high ratio and low threshold. This system helps fine-tune bass response, allowing powerful, chest-thumping swells without low-end feedback.

CROSSING OVER

Most commercial 2- and 3-way loudspeaker systems feature an internal, passive crossover that separates (in a 3-way system) low, mid, and high frequencies. The frequency-dividing network, a simple circuit card, is contained within the speaker enclosure and is automatically inline when a cable is plugged in.

The *crossover point*—the frequency where one component (a bass cone) stops receiving input signal and the next component (treble horn) receives it—is critical to speaker performance. Unfortunately, most 2-way systems set the crossover point at the middle range of the human voice. Since bass speakers and horn drivers have radically different tonal qualities, this often produces unnatural vocal sound. Passive crossovers also demand a generous chunk of amplifier power, which decreases system headroom.

When an act's volume level increases and the sound system is operating near its performance limits, efficiency is gained by employing an electronic crossover. This circuit splits a line-level signal at selectable frequencies and routes the resulting signals to two or more amplifiers. Look for electronic crossovers with balanced inputs and outputs that minimize hums and buzzes. Better units utilize XLR connectors, rather than less-reliable, ¼-inch phone jacks.

Quick Tip: The balance settings for low-, mid-, and high-frequencies on the crossover are critical. If improperly adjusted, the crossover can ruin your loudspeakers by routing too much electrical energy to fragile components. Adjust levels by listening to a favorite song (or two) through the sound system. Start adjusting the high-frequency send first, then work downward. If the crossover is wired incorrectly, high frequencies won't hurt the bass speakers; however, a runaway bass signal fries horn drivers without pity.

SYSTEM CONTROLLERS

Electronically controlled systems

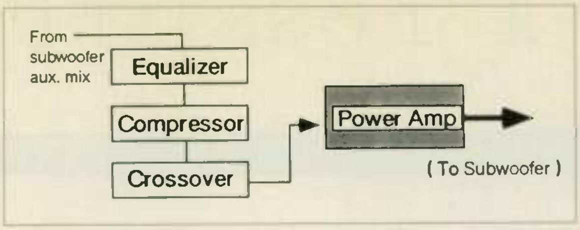


FIG. 4: Dedicated equalizers and compressors help tune subwoofer systems to rock the house.

employ techniques—such as active crossovers, equalization, limiters, and speaker protection circuitry—that are designed to optimize performance. Companies currently marketing these systems include Apogee Sound, Bose, Community Light & Sound, Eastern Acoustic Works, Meyer Sound Labs, Professional Audio Systems, Renkus-Heinz, and Yamaha. Proponents of system controllers point out the advantages of using electronics to tweak performance, while manufacturers of non-controller systems tout the benefits of a traditional, less-is-better approach.

POWER AMPS

Power amplifiers, simply defined, provide the electrical energy that speakers convert to acoustical energy (sound). (For an in-depth discussion of power amps, including a sidebar on speaker-wiring, see "Basic Studio Series, Part 2: Power Amplifiers," in the December 1989 **EM**.) Simple sound systems often utilize an amplifier contained within a mixer. This arrangement makes system setup a breeze, as its sole requirement is plugging speaker cables into the mixer's output jacks. Unfortunately, many inexpensive powered mixers produce audible background noise, which is caused by the close proximity of sensitive mixing and high-level power electronics.

Stepping up to cleaner sound involves using a separate power amp, or amps, driven by the soundboard. If your speaker system uses internal passive crossovers, you need only run a shielded signal cable from the mixer's output to the amplifier's input. Speaker cables are connected to the amplifier, where each stereo channel can drive a speaker enclosure or two.

However, never connect more loudspeakers to an amp than recommended by the manufacturer. Driving additional speakers puts a lower impedance value on the power amplifier's output section, and most amps fail, or engage a shut-off function, when presented with a speaker load measuring too low in impedance. For best results, never go below a 4-ohm load unless the amplifier is specifically designed for it.

Pumping more low frequencies through the sound system with bass guitar, low synthesizer pads, and drum sounds increases the amount of power the system draws.

Adding a subwoofer system often means adding power amplifiers, and this may require a dedicated power-distribution system to prevent blowing house fuses. This means setting up your own electrical panel, with breakers and bare-wire "tails" that an electrician ties directly into a building's main power source. Obviously, it is best to consult a knowledgeable sound technician if you are considering a complex system that employs a variety of power amplifiers. (For more on AC power problems in live-performance venues, see "Performance Power" in the May 1990 issue.)

Quick Tip: When selecting a power amplifier, choose the proper power rating. Don't drive speakers rated at 100 watts with a 500-watt amplifier. A good rule of thumb is to check the power rating on the speaker, then use at least the same wattage to power it. For best system headroom, double your amplifier power (a 200-watt amp channel can power a speaker rated at 100 watts). And remember, input gains need not be run wide open.

CURTAIN CALL

When it comes to assembling a sound system, everything counts. Speakers, amps, crossovers, limiters, and cables comprise a "system" that delivers sound to an audience. While the quality of that sound depends on many things, choosing the proper speaker system is the first critical step toward providing audiences every nuance of your music.

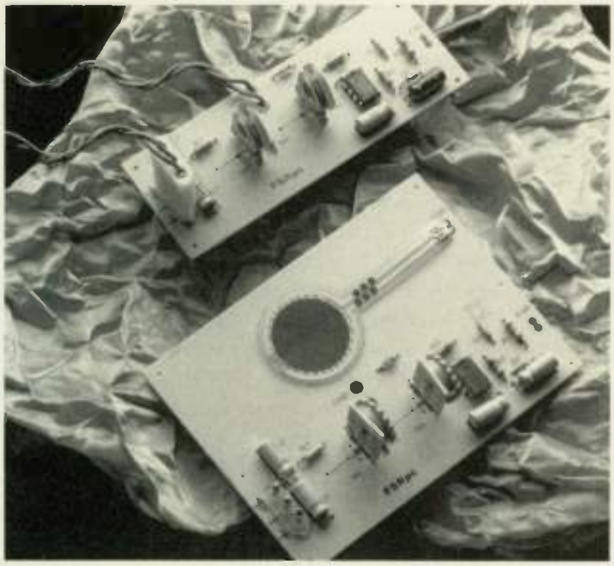
(Thanks to George Petersen and David Scheirman for their insights and assistance.)

EM *managing editor* **Steve O.** *and assistant editor* **Michael Molenda** *just want to rock out and have some fun.*

Fun Under Pressure

By Charles R. Fischer

The EM Presto puts expressive control at the tip of a finger.



DAN ESCOBAR

MIDI and microprocessors have given musicians more than just sequencers and multitimbral modules. Velocity and pressure sensitivity are two of the great breakthroughs in electronic music. Before MIDI, these features were available only on a few high-end synths, but today it's just the opposite. All but the cheapest keyboards have velocity sensitivity, and most have at least channel aftertouch.

But there's plenty of room for improvement. For the last couple of years, I've been using devices that measure finger position and pressure to build more expressive MIDI controllers.

The *Force Sensitive Resistor* (FSR) is one of the more interesting items I've found. It's small, reasonably cheap, and can be interfaced with both analog and digital circuitry. Other devices can be used, but the FSR often performs as well, or better than, more expensive sensors. They have been used as drum sensors by manufacturers such as KAT, Akai, and Simmons.

ENTER THE FSR

Take a look at **Fig. 1**. It's from a business card from Interlink Electronics, and it includes an actual working example of a Force Sensing Resistor. The FSR consists of a grid of conductive ink and a "pad" of resistive ink printed on the card itself. When the resistive pad is placed over the conductive grid, the overall area of contact between the two elements changes with applied force.

The FSR behaves as a variable resistance in series with a momentary switch. With no pressure applied, it's virtually an open circuit; applying force to the sensor lowers the resistance radically. How much the resistance varies depends on the design of the individual FSR. A typical unit may vary over a range of 500 to 1, or even more.

Pairing the sensor with a fixed resistor forms a simple voltage divider, and this produces an output voltage proportional to force. A pair of op amps buffers and amplifies the raw output and provides calibrations for the voltage range and offset.

Most control-pedal inputs on today's instruments and accessories accept a DC control voltage in the range of 0 to 5 volts. This makes it easy to use the FSR with a variety of instruments. **Fig. 2** shows the complete circuit.

Zener diode D1, together with R1, provides a stable reference voltage for biasing the FSR. The FSR and R2 form a voltage divider; with no pressure applied to the FSR, the output of the divider is negligible. Once pressure is applied, the resistance of the FSR falls rapidly, causing the voltage across R2 to rise proportionally.

The reference voltage and the ratio of the FSR to R2 determine the maximum output voltage. Op amp IC1A buffers and amplifies the voltage divider output.

Trimpot R4 adjusts the sensitivity of the FSR by altering the gain of op amp IC1A. IC1B provides an adjustable bias voltage. Normally, R6 is set so that the output voltage is 0 volts when the FSR is untouched.

PUTTING IT TOGETHER

Since everything needs a cute name, I decided to call my circuit the Presto. The opening photo shows the completed project in two styles.

The easiest way to build it is to purchase the PC board or parts kit from PAiA Electronics of Edmond, Oklahoma (see Parts List). If you prefer, you can etch your own board or wire the circuit with flea clips and perfboard.

The one hard-to-get part is the FSR itself. The manufacturer does not maintain retail distribution, and they have a $200 minimum order. PAiA has arranged to carry a stock of 7/8-inch, round FSRs. As it stands, they are the only source of these parts for hobbyists. (If you haven't gotten PAiA's catalog, you should. They have been around forever, and their list of accomplishments includes the first programmable drum machine and a user newsletter that later became a magazine known as *Electronic Musician*.)

The parts kit comes with single-turn trim pots for R4 and R6. If you prefer, you can use multiturn pots for extra precision and stability.

The Presto uses a regulated power supply of ±12 to ±15 volts. (Avoid batteries unless you're just breadboarding a temporary version.) If you don't have a bipolar power supply, you may find a suitable one at your local parts dealer. PAiA sells a bipolar power supply kit,

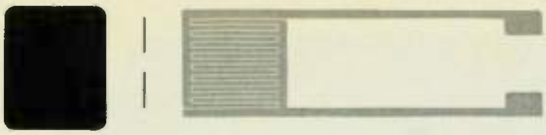


FIG. 1: The Interlink FSR consists of two elements. The square pad on the left is a resistive ink, and the grid on the right is made up of conductive ink. When the pad is placed atop the grid, it forms an element whose resistance varies in proportion to pressure.

though it's rather large if the Presto is all you want to power. The November 1987 **EM** has an inexpensive power supply project. If you elect to build a supply, be sure that you observe suitable precautions with the AC from the wall.

SENSOR MOUNTING

The PC board offers several options for mounting the FSR. The board has a large ground-plane area with mounting holes for the sensor. You can mount the FSR over this area for a convenient, all-in-one assembly. If you mount the sensor on the etch side, you can mount the board upside-down so the sensor is accessible through an opening.

You can also bend the FSR out away from the board, or cut the board in half, and mount the sensor in the holes nearest the other components. Use caution when installing the FSR; excessive heat can burn or melt the plastic substrate, and this can damage or ruin the unit.

As the photo shows, you can also mount the FSR remotely. Use thin-diameter coaxial cable, such as RG-174/U, between the sensor and the circuit board. Attach the shield of the cable to the junction of R1, D1, and C3. The inner connector should connect to the end of R2 that also connects to pin 3 of IC1A. The FSR is not polarized in any way, so the connection of shield and hot lead at that end doesn't matter. Be careful in attaching the cable to the FSR's leads, and use heat-shrink tubing—or something similar—to provide strain relief.

The FSR works best when mounted on a rigid surface. Warning: Never use *cyanoacrylate* adhesives (Krazy Glue) on the sensor; it will ruin the plastic substrate. Instead, use a silicone-based RTV for mounting.

It's a good idea to glue a thin piece of foam rubber or neoprene over the sensor. The rubber enhances the response of the FSR by spreading out the force applied. In addition, it helps protect the FSR against accidental damage.

TESTING AND CALIBRATION

Before you apply power, take time to inspect the finished board carefully. Is every part in the correct place? Are the ICs and capacitors oriented correctly? Are there any solder shorts? Take your time doing this; 90 percent of problems in home-built projects turn out to be simple construction errors. When you are satisfied that everything looks good, set both trimmers about midway and turn on the juice.

You will need a voltmeter or oscilloscope to test and calibrate the circuitry. First, verify that both supply voltages are reading correctly. If one or both are not where they should be, turn the supply off and initiate troubleshooting procedures.

If the supplies are OK, check the CV output. You should see a DC bias voltage of no more than ±4 volts. Now, touch the sensor with your fingertip. If everything is working properly, the output voltage will rise and fall as you vary the pressure.

To calibrate the circuit, turn R4 (Span) up for full gain, then set R6 (Center) for zero volts at the output. Touch the FSR with your forefinger and set R4 for appropriate output at maximum force. Then take

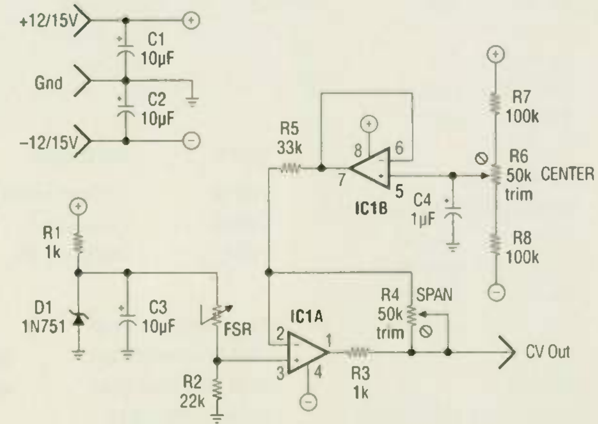


FIG. 2: The Presto interface circuit consists of a voltage divider and a variable-gain buffer with adjustable offset voltage.

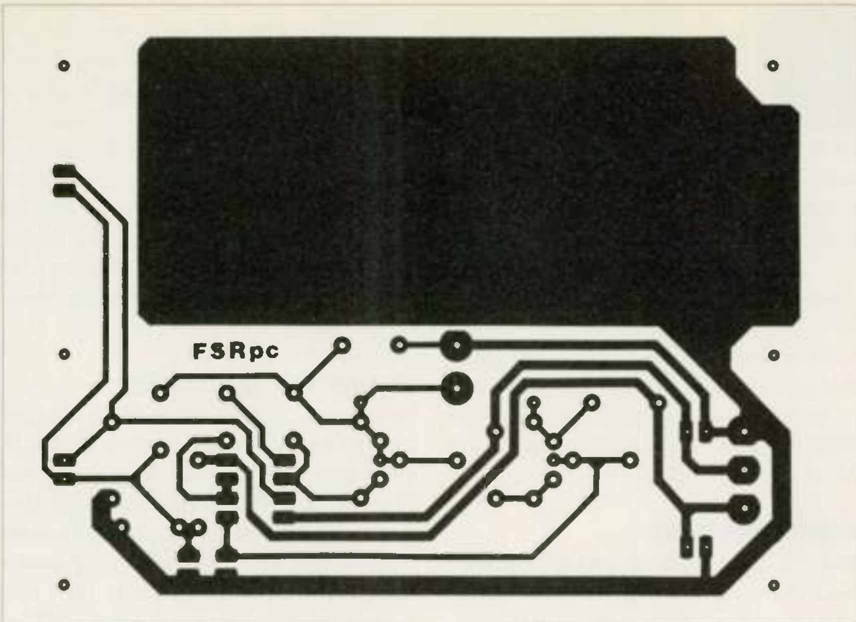


FIG. 3: The Presto is laid out on a single-sided PC board. The large ground-plane area provides convenient mounting for the pressure sensor.

your finger off and readjust R4. There is a slight interaction between the two trims, so you may have to go back and forth a few times in order to get things close.

If you want to increase the output level, replace R5 with a 22k or 20k resistor. You also can change the response of the FSR by substituting a smaller-value (not less than about 1k) resistor for R2.

APPLICATIONS

The Presto has a number of uses, including:

Momentary Footpedal. Many performers use footpedals for continuous control, but they have disadvantages in some situations. For vibrato control, you must bring the pedal all the way back when the modulation is unwanted (unless it's spring-loaded). Worse, some pedals have a tendency to slip

PARTS LIST

RESISTORS
($1/4$-watt, 1%, metal film)

R1, R3	1k
R2	22k
R5	33k
R7, R8	100k

TRIM POTS
(multi-turn optional)

R4, R6	50k

CAPACITORS
(25 working volts or greater)

C1, C2, C3	10 µF Al electrolytic
C4	1 µF Al electrolytic

DIODES

D1	1N751 Zener
D2, D3	1N914 signal (optional)

INTEGRATED CIRCUITS

IC1	dual op amp; LF353, TL072, TL082, or 5532

IC SOCKETS (optional)

(IC1)	8-pin

OTHER COMPONENTS

FSR	Interlink Force-Sensing Resistor (see below)

Part #	Description	Price
FSR78	$7/8$-inch round FSR	$ 12.50
FSRPC	PC board	$ 9.95
FSRK	Complete Kit	$ 29.95

Parts available from:
PAiA Electronics, Inc.
3200 Teakwood Lane
Edmond, OK 73013

Include $2.50 for postage and handling.

toward the maximum position on their own. One alternative is to use the Presto as a momentary footpedal, producing a voltage only when the player places a foot on it.

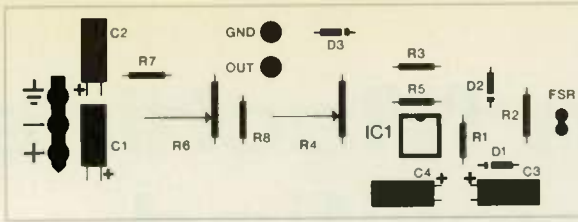


FIG. 4: Parts layout for the EM Presto.

Mount an FSR on a sturdy case of the proper size and shape. Cover the sensor with rubber to protect it and even out its response. You can put the entire circuit inside the case, or mount the FSR in one box, with the rest of the circuit elsewhere. Choose a box that provides a solid surface and feels comfortable after a long gig.

Pressure Pads. Old-time synthesists may remember the PPC pads found on late-model ARP Odysseys. This unusual device replaced the clumsy pitch bend knob on earlier units. The PPC consisted of three pressure-sensitive rubber pads, one for bending sharp, one for bending flat, and one to provide LFO modulation. The PPC was capable of very expressive results. Unfortunately for ARP, most synthesists preferred the Moog wheels.

You can mount one or more FSRs to provide a touch-sensitive pad anywhere you like. As with the pedal, be sure to cover the FSR with rubber or neoprene to enhance the feel. You can use a single pad for modulation or go crazy and stick any number of them on your axe to do whatever.

FSR-TO-MIDI

Many readers will want to use their FSRs with MIDI. The easiest way to convert the raw control voltage into a MIDI controller is to connect the Presto to an instrument's footpedal or CV input. The Anatek Pocket Pedal (reviewed in the May 1989 EM) is another possibility.

The best solution I've found is the "EM MIDI Volume Fader" (MVF) project from the February 1991 EM. By replacing the original slide pots with external control voltages, the unit becomes a powerful MIDI controller. For example, you could use one or two MVF inputs with the Presto(s) and still have six left over for use with faders, wheels, pedals, or joysticks. The MVF can generate Pitch Bend and Channel Aftertouch, as well as most Continuous Controller messages.

Fig. 5 shows the necessary modifications. First, decide which MVF input, or inputs, you want to use with the Presto. Disconnect both the pot's wiper and the appropriate bypass cap (C 14-21 on the MVF circuit board). Add diodes D2 and D3 to the the Presto circuit board and connect the CV out to the MVF's input. The diodes protect the MVF in case the Presto's output goes too high or low. Power up both boards and verify that everything is operating. You'll probably want to retweak the trimpots for the best feel.

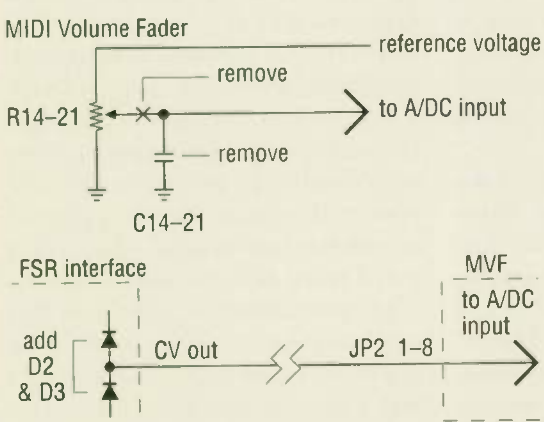


FIG. 5: To use the Presto with the MIDI Volume Fader (see February 1991 EM), install diodes D1 and D2 on the Presto and remove the bypass capacitor from the MVF input.

CONCLUSION

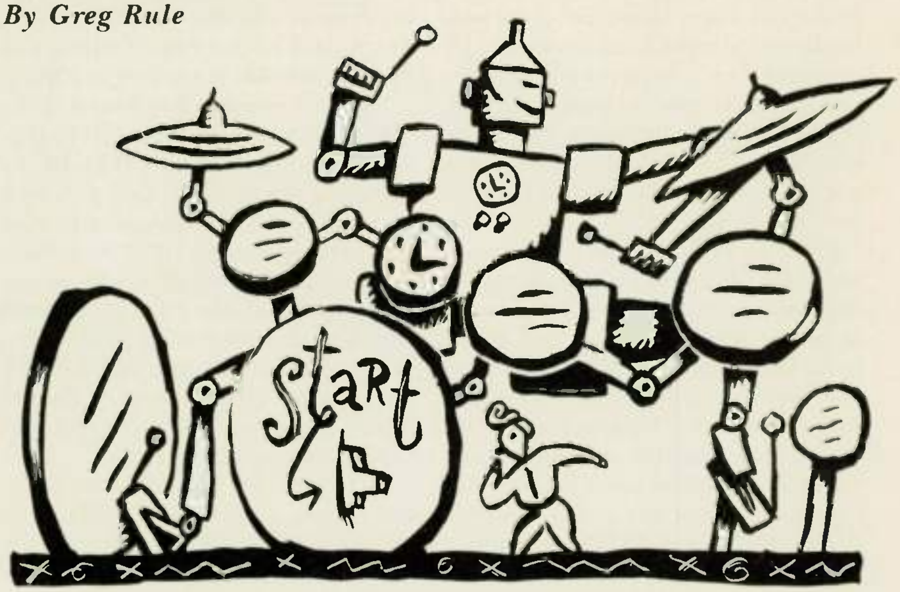
It's hard for me to get excited about electronic gadgets, but the Presto is a definite exception. Maybe it's because it measures gestures associated with feel, making it a device for emotional, improvising musicians. You can squeeze it, press it, beat it, whatever. You'll find the EM Presto is just the thing for converting those gestures into something that your instruments can understand.

As a senior technician in the electronics industry, Charles R. Fischer *knows all about the fun of being "under pressure."*

Drum Machines Demystified

By Greg Rule

A little knowledge helps conquer the beat box.



AD McCAULEY

Wanted: the perfect drummer. Must have lightning-fast licks, solid time, and limitless endurance. Mastery of odd time signatures and polyrhythms required. Must also be reliable, willing to work around the clock for no pay, ready to travel on short notice, and play precisely what you are told.

The above "Musicians Wanted" ad ensures a lonely telephone, because the number of qualified drummers won't fill the radius of a splash cymbal. The cold, hard truth is that drum machines are capable of rhythmic perfection; humans are not. Yet, despite their expert percussive chops, drum machines are merely jumbo paperweights until humans tell them what to do.

Before you make that step, you need to know exactly what a drum machine is. *The Electronic Musician's Dictionary*, by Craig Anderton, defines a drum machine as: "A device that combines sequencing and (typically) sampled drum sounds to allow musicians to program repetitive drum patterns that sound at least somewhat like a human drummer." As complicated as many drum machines seem on the surface, they all break down into two basic components: a sound source and a sequencer.

THE SOUND SOURCE

The sound source is the "voice" of the drum machine. Most early drum machines (mid-1970s to early-1980s) used a form of analog synthesis to construct drum sounds. The resemblance to actual acoustic drum sounds was minimal, but the "synthetic pots and pans" character changed the style of pop music. Phil Collins' classic song, "In the Air Tonight," demonstrates the percussive qualities of early analog sound sources.

Things "got real" when Roger Linn introduced the LM-1 Linn Drum Computer in 1980. This was the first mass-market drum machine to employ digital recordings (or *samples*) of real instruments as sound sources. The revolutionary product became an industry mainstay, despite a $5,000 price tag.

Fortunately, sample-based drum machine prices have decreased dramatically since the days of the LM-1. A high-quality machine such as the Alesis SR-16 is less than $400. What's more, modern drum machines often provide a choice of dozens of diverse instrument samples. Besides percussion, some machines include samples of bass guitars, orchestra hits, and special effects. (Yamaha's RX7 featured James Brown-inspired vocal samples, such as "Get Funky" and "Huh.")

THE SEQUENCER

The second essential component of a drum machine is its sequencer. This allows rhythm patterns to be recorded, played, and edited. A sequencer is a type of computerized music-data recorder that looks at incoming notes and records their precise location and velocity (how hard they were played) as numeric information. This data is stored in the drum machine's memory.

To avoid confusion, remember that sequencers are not exclusive to drum machines. Synthesizers (such as Korg's M1) and samplers (E-mu's Emax II) also have built-in sequencers. There are also software sequencers (see the "EM Guide to Sequencing Software" in

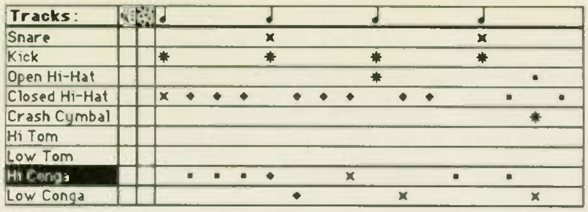


FIG. 1: A typical four-beat drum machine pattern is built by playing individual drum sounds one at a time.

the August 1991 issue) and dedicated hardware sequencers (see the buyer's guide in the April 1991 issue).

ENTERING NOTES

A drum machine's front panel buttons provide the most common method of note entry. Each button is assigned to play, or *trigger*, a different sample. For instance, tapping one button might trigger a bass drum sample, while another button is assigned to a snare sound. Some drum machine buttons are *touch sensitive*, meaning the triggered sounds respond appropriately to how hard they are tapped.

Most units allow creation of customized "drum kits" by assigning a user's favorite samples to specific buttons. A sample's tuning, panning, and audio output assignment (stereo mix or individual output) also are open to user manipulation.

Recording a drum beat into the machine requires selecting a time signature, a tempo, and a (bar or measure) length. When you press the record button, an audible metronome count is activated. This is your conductor. Then you simply tap out a groove, one or two drum sounds at a time. The drum machine allows stacking of sounds until a suitable battery of percussion is playing. It then loops (repeats) the pattern until the stop button is depressed.

A *step record* function is used for drum parts too fast or mechanically tricky to play in real time. Step recording allows notes and rests to be recorded one step at a time without the metronome (or clock) running. A note value (eighth note, sixteenth note, etc.) is entered into the drum machine and each rhythmic figure is tapped in, beat by beat, until the pattern is complete. Note and rest values may be changed throughout the process. Fig. 1 shows a typical 4-beat drum pattern.

EDITING

Once a rhythmic pattern is entered, the drum machine allows editing of any mistakes. You can erase bad notes and overdub new ones, and you even can correct timing errors with *quantization*. For example, if you record a groove and slip some snare hits too far ahead of the beat, you can make the drum machine align each note to a set value (the nearest quarter note, perhaps). Automatically, a sloppy rhythmic track is transformed into the miracle of perfect time. The only down side is that overuse of quantizing can make the drums sound robotic, or lifeless. Several quantization values are available, from quarter notes to 32nd-note triplets (and some drum machines allow even higher values).

Other popular features include *swing* and *note shifting*. When swing mode is activated, the drum machine can "loosen" an eighth-note rock groove into a triplet-based shuffle beat. The percentage of swing imposed on the straight beat is programmable. Note shifting allows beats to be moved "in the pocket" or slightly behind the beat. Users can specify the precise distance notes are slid behind, or in front of, the selected quantization.

PUTTING IT TOGETHER

Once your grooves are entered and edited, you can chain them together to compose a song. To accomplish this, most drum machines utilize *pattern* and/or *song* sequencing formats. Patterns are short, repetitive motifs with user-determined lengths. A couple of short patterns often comprise the entire verse, chorus, and bridge sections of a contemporary pop tune. Using these repetitive (looping) patterns instead of one continuous rhythmic motif saves enormous amounts of a drum machine's memory.

You can construct songs by assembling patterns in linear order (verse/verse/chorus/verse, etc.). Once the initial song tempo is entered, select your first drum pattern. You can set this pattern to repeat several times before it advances to the next pattern (such as the chorus). To change tempo, insert a command into the list at the appropriate point. When the song list is complete, you can save it to its own memory location for future recall.

MEET THE FAMILY

There are other electronic drum gadgets worthy of mention. Pre-programmed *preset* units, such as Roland's CR-80, are available for musicians who don't want to write their own patterns. These machines permanently store hundreds of patterns in memory. You need only pick a groove, anything from reggae to rap, and press the Play button.

If you want a drum machine to record or play an actual performance, MIDI *drum controllers* trigger external sound sources when a drum stick hits a special pad. Products such as Kat's drumKAT and Roland's Octapad II are connected to a drum machine's MIDI input and transmit note and velocity information from their MIDI output(s). A drum machine sound (snare drum, for example) is assigned a MIDI note number and triggered every time the selected pad on the controller is struck. In essence, the drum machine's note-entry buttons are replaced by surfaces that can be played in real time by an actual percussionist. All other aspects of the drum machine continue to work the same if you use a drum pad. In other words, you can still edit or quantize a pattern played on a MIDI drum controller.

MIDI *drum modules* are sound sources without a sequencer and note-entry buttons. Most units, such as Kawai's XD-5 and E-mu's Procussion, are housed in a rack-mount chassis and must be used in conjunction with an outboard MIDI sequencer or drum controller. Like drum machines, modules are triggered by assigning MIDI note numbers to the desired drum sounds. If you use a MIDI drum controller, any drum machine becomes a MIDI drum module.

SITTING IN

Hopefully, this information eases any discomfort about playing with "the perfect drummer." You can avoid further intimidation by having friends demonstrate their favorite drum machine or checking the selection at a music store. Don't be shy, and ask lots of "stupid" questions if necessary. Also, make it a point to study accomplished drummers. A working knowledge of "live" drumming styles enhances the character and feel of your own drum-machine programming.

Greg Rule *is the former assistant editor of* Drums & Drumming *magazine.*

Questions and Answers

By Alan Gary Campbell

Every technician needs tools, but the musical equipment service person must have a good set of ears as well.

Q. My Korg T3 synth has distortion in the line and headphone outputs. It's a subtle, analog-like distortion that's more noticeable with clangorous sounds that use prominent effects, such as electric piano with chorus. The local service center investigated this and performed what the technician called a "DAC adjustment," but the problem remains. I've compared my T3 with another one that doesn't have this distortion, so I know there's something wrong with mine, but what?

A. This type of subtle, analog-like distortion can be caused by the C2878 output-muting transistors used in the T-series and Wavestation, but this only affects some units. Replacing the transistors alleviates this; the common 2SC945 type is an adequate substitute. Bob O'Neal, of O'Neal Custom, Franklin, Tennessee, called my attention to this solution and was among the first to troubleshoot the problem.

On the T3, the affected transistors are Q2, Q3, Q4, and Q5. The distortion generally affects all the line outputs and the headphone output, sounds like a slightly overdriven analog VCF, and is unaffected by reducing oscillator level.

A domestic cross-reference for the 2SC945 transistor is the NTE 85. Mr. O'Neal has substituted this part successfully, but the 2SC type seems preferable. Note that the transistor has an emitter-collector-base (ECB) pinout, *not* an emitter-base-collector (EBC) pinout.

The service tech probably performed a "DAC Offset" adjustment. This compensates for input offset of the op amp used as a current-to-voltage converter at the output of the digital-to-analog converter to minimize linearity errors. This parameter is not generally thought to be a source of audible distortion unless the offset is especially high, as caused by a defective component or serious misadjustment. The distortion resulting from that situation has a pronounced "digital" sound, with many non-integer harmonics.

The DAC offset adjustment typically is performed with the aid of a high-accuracy (0.03% DC) DVM. However, some technicians feel they get a slight sonic improvement by tweaking the offset "by ear." To try this, select a sound that contains naturally occurring non-integer harmonics—such as electric piano—and disable any effects. While monitoring the audio output, adjust the DAC offset to yield the most musical effect, with minimum distortion. "Digital" distortion that cannot be rectified by DAC offset adjustment may indicate a mainboard defect.

Note: To preclude voiding the warranty, these service procedures should be performed by a Korg authorized service center.

Q. I'm interested in do-it-yourself service and possibly a career as a service technician. What tools and test equipment do I need for basic service work?

A. The array of tools and equipment you *could* need for service is mind-boggling, but a basic selection is enough for most jobs. The sidebar gives my recommended list, with selected manufacturer's catalog numbers in parenthesis. The culling of the indispensable from the merely desirable is arbitrary at best, and any list of this type is necessarily personal.

For the aspiring technician, quality tools can be prohibitively expensive. It is wise to purchase the best tools available, even if it means you can afford

THE COMPLEAT TECHNICIAN:

The Alan Gary Campbell List of Recommended Tools and Equipment

HAND TOOLS
- 6-inch extra-thin needle-nose pliers (Bonney 777-6C)
- 4 ½-inch standard needle-nose
- 6 ½-inch standard needle-nose
- 6-inch standard diagonal cutters
- 4 ½-inch, 45-degree angled-tip cutters (Xcelite 73CG)
- 5-inch wire-stripping pliers (Diamond ST55RP)
- 6-inch slip-joint pliers
- 10-inch slip-joint pliers
- 7-inch Vise-Grip pliers
- 6-inch crescent wrench
- Selection of slotted- and Phillips-type screwdrivers
- Selection of English and metric nutdrivers
- Selection of English and metric hex-socket (Allen) screwdrivers
- Selection of Swiss needle files
- 4 ½-inch fine-tip self-closing tweezers (Hunter Tools CD-51)
- 4-inch nonconductive tweezers
- Selection of trimpot adjustment tools (Radio Shack 64-2230)
- Conductive wrist strap (Radio Shack 276-2397)
- IC extractor
- IC pin aligner (Radio Shack 276-1594)
- headphones
- Small monitor amplifier (Radio Shack 277-1008)
- Clip-type heat sink
- X-acto knife, #1 handle with #11 blade
- 5-inch scissors
- 6-inch flexible ruler
- Pocket magnifier
- Temperature-controlled soldering station (see text)

MECHANICAL
- Hammer
- Center punch
- Drill and bits
- Hole punches
- Vise
- Heat gun

TEST EQUIPMENT

Hand-held digital multimeter (Radio Shack 22-171)
Logic Probe (Radio Shack 22-303)
20 MHz, dual-trace oscilloscope

only a few at a time. Poor-quality tools yield poor-quality work. It is tempting to economize with bargain-bin screwdrivers, but my advice is *don't*.

Test equipment is even more expensive, but you can do a lot of useful work with a pocket digital voltmeter (Radio Shack 22-171), a logic probe (Radio Shack 22-303), and some savvy.

An oscilloscope may be the most-coveted item for the aspiring tech, and rightly so. A decent 'scope is invaluable for service and troubleshooting. It's a wonderful tool for learning about electronic circuits, as well.

In recent years, the quality of low-cost oscilloscopes has improved markedly. Today, you can buy a decent 20-megahertz dual-trace 'scope, such as the B & K Precision model 2120, for about $400. While less than ideal for troubleshooting high-speed digital circuits, 20-meg dual-trace is sort of a small-bench standard for oscilloscopes and is more than adequate for most purposes.

Soldering equipment may be the most critical category of service gear. It's certainly the most neglected. You can do good work with a good-quality soldering iron (e.g., Ungar or Weller) *if* you maintain the iron and tip, allow sufficient thermal recovery time between joints, and use the iron/tip combination only with work of the size and thermal capacity intended.

Consider making the investment in a closed-loop, temperature-controlled soldering station. The improvement in speed, consistency, and quality of work—and the inherent reliability—make it a better investment than a plain iron. I recommend the Ungar UTC200 and Weller WTCPS stations. (Note: The December 1990 "Service Clinic" has a discussion of different types of solder.)

Q. Where can I obtain a chassis punch the right size for mounting MIDI jacks?

A. Mouser Electronics carries Greenlee round punches in the sizes required for the 180°, 5-pin DIN jacks that MIDI uses. You need a 5/8-inch round punch (catalog number 565-3803) for panel-mount DIN jacks and/or an 11/16-inch round punch (catalog number 565-3804) for PC-mount jacks. Both sizes require a 1/4-inch pilot hole. Mouser is an excellent source of tools and components. Write or phone for a catalog: Mouser Electronics, 2401 Highway 287 North, Mansfield, TX 76063; tel. (800) 992-9943.

You also can use a tapered reamer or cone-cut drill bit to make DIN jack holes, but the results are not as clean or as uniform as you get using a punch.

ENSONIQ UPDATE

In the July 1991 "Service Clinic," I stated that revision 3.5 ROMS for the Ensoniq ESQ-1 were available at no charge from authorized service centers. That statement, true at the time of writing, is no longer operant. As of June 1, the cost of the ROM is $39.95. This change also affects update ROMs for the ESQ-M and SQ-80. For more information about Ensoniq updates and service policy, call Ensoniq's service department, tel. (215) 647-3930.

EM *contributing editor* **Alan Gary Campbell** *is owner of* Musitech™, *a consulting firm specializing in electronic music product design, service, and modification.*

Reviews

Roland JD800 Synthesizer

By Peter Freeman

Roland combines contemporary sound quality with programmer-friendly design.

The past few years have seen dramatic changes in the world of synthesizers. Electronic musicians have benefitted from the flexibility and sound quality of powerful, affordable, multitimbral synthesizer and all-in-one synthesizer/sequencer/effects-processor combinations. However, there also has been a decrease of easy-to-use programming functions in many instruments. Increasingly, musicians have been forced to negotiate daunting layers of pages and menus to get to the nuts and bolts of their instruments.

The Roland JD-800 synthesizer's 57 dedicated parameter sliders are active at all times, providing extensive, easily accessed editing capabilities.

It is both surprising and welcome when someone bucks the trends and looks to an earlier, but still-valuable approach. With the JD-800, Roland has introduced an instrument that takes a much-needed "backwards leap" to the concept of easy and extensive user programmability.

OVERALL LAYOUT

Roland's JD-800 is a 24-voice synthesizer that combines digitally generated waveforms with traditional synthesis architecture, built-in effects, and a 5-octave keyboard with velocity and channel aftertouch. Roland included its usual combination left-right pitch bend/up-down modulation lever and ¼-inch jacks for the sustain footswitch and sweep pedal. Unlike some synths, the JD-800 does not scan for and adjust to the polarity of your sustain footswitch, so you have to use a Roland/Yamaha-type switch; that extra E-mu/Ensoniq-type pedal (the polarity of which is the opposite of Roland's) has to stay in the parts box.

The synth has two stereo pairs of ¼-inch outputs, Direct Out L/R and Mix Out L/R. The Direct Outs only function in Multi mode, where it is possible to route the signal for each Part to either set of outputs. A headphone output is included, but I wish it were located on the front of the instrument, rather than on the back panel.

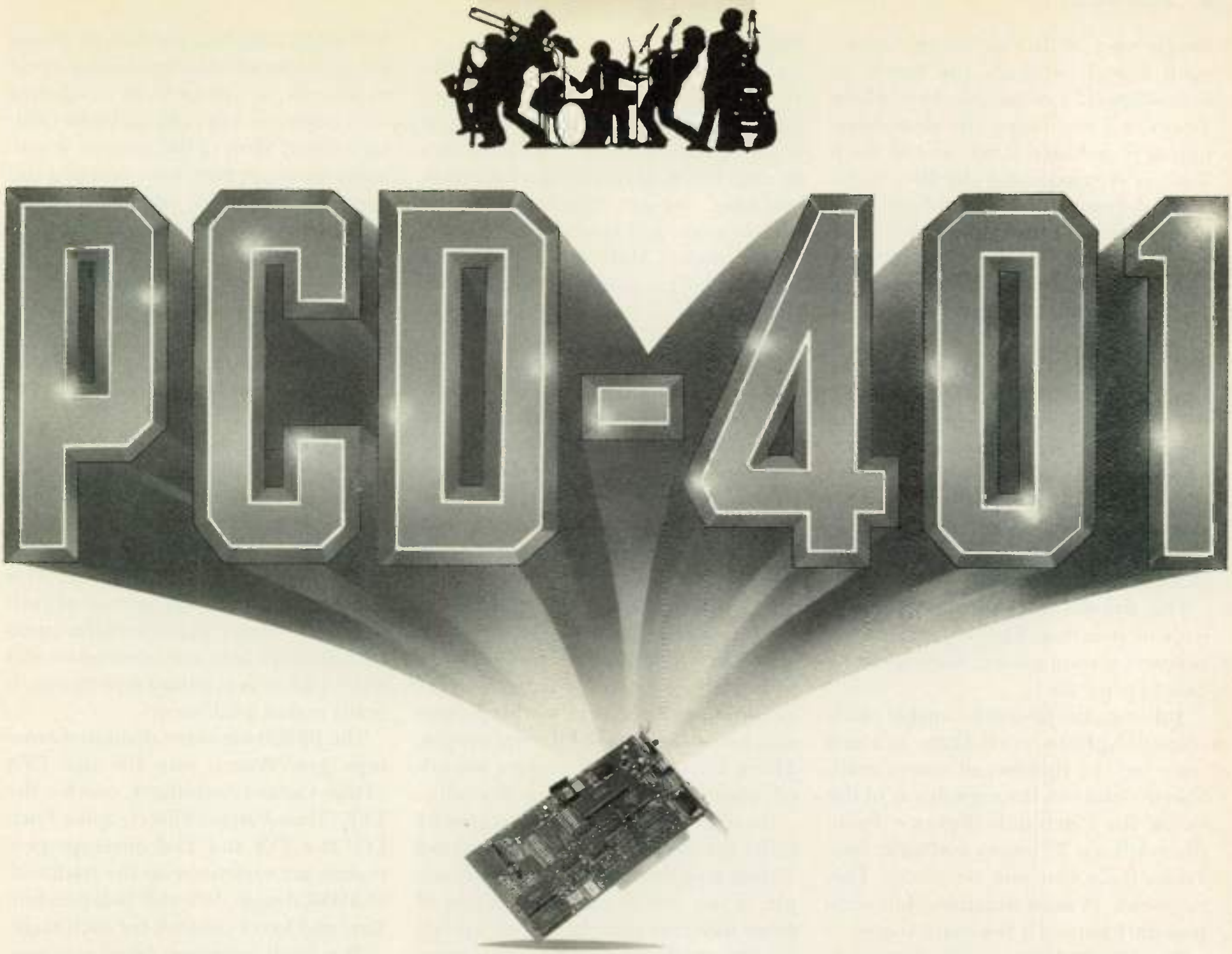
The biggest difference between the JD-800 and other recent synths is obvious: The front panel sports 57 dedicated parameter sliders that are active at all times. This reduces (but doesn't quite eliminate) paging through layers of parameters and makes it easy to accomplish most programming moves.

In addition to the dedicated parameter sliders, the JD-800 can be programmed by means of two adjacent, orange, backlit LCD displays; four cursor keys; increment/decrement keys; and a data-entry slider. The cursor and data-entry keys are not always active; if you select a parameter by moving its slider, the data entry keys have no effect, and you must adjust the parameter with the slider or the Palette. This is surprising, as it seems logical to use these keys for single-increment adjustments.

The Palette is a useful and unique feature. This group of four adjacent, identical sliders assumes control of the currently active parameter, with each slider in the Palette corresponding to one of the four Tones in the current Patch. This "quick edit" feature is handy in sound-editing. It would be great to see stuff like this on other instruments.

TONES AND PATCHES

The JD-800 has two playing modes,


PCD-401

Single and Multi. In Single mode, each sound program (or Patch, in RolandSpeak) can consist of up to four Tones, a Tone being the most basic unit of sound on the instrument. Each Tone is created using the JD's front-panel sliders, and any or all Tones in a Patch can be edited independently, together, or in any combination. A Layer/Active button determines which Tones are "live" on the front panel (Active) and which are heard in the Patch (Layer). The state of the Layer/Active button is stored with the Patch, so if you leave a Patch in the middle of editing, you can pick up where you left off. A Patch also includes the settings of the onboard multi-effects unit and a set of Common parameters, including pitch-bend range and aftertouch sensitivity.

This sounds a bit confusing, but it isn't in practice. The system quickly becomes second nature, making the JD easy to program.

Although the JD-800 is capable of 24-voice polyphony, each Tone uses one voice, so the number of voices available depends on the complexity of the Patch. If a Patch only uses one Tone, the synth has 24 voices available; four Tones leave you with six voices. This may work for most situations, but occasionally I wanted a few extra voices.

The JD-800 holds 64 Patches internally and 64 more on an optional RAM card. One advantage of the JD-800's architecture is that each Tone has an independent Key Range control. Thus, a Patch can have four separate sounds spread across the keyboard (three split points). This is a nice feature, especially for performance.

Product Summary

PRODUCT:
Roland JD-800
Synthesizer

PRICE:
$2,895

MANUFACTURER:
Roland Corporation
7200 Dominion Circle
Los Angeles, CA 90040-3647
tel. (213) 685-5141

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ● ●
EASE OF USE	● ● ● ● ◖
DOCUMENTATION	● ● ●
VALUE	● ● ● ●

MULTI MODE

In Multi (multitimbral) mode, up to five Parts can be played at once, each on its own MIDI channel. In Multi mode, a Part includes one Patch plus settings for MIDI receive channel, level, panning, output assignment, and effects mode and level.

In addition, Multi mode includes a Special Setup, in which each key on the keyboard is assigned a sound of its own. These sounds are similar in structure to the Tones used in constructing Patches. Each of these "Special Tones" can be modified individually, and although it's tedious, there are advantages to this mode of operation. Each sound can have its own level, pan position, envelope mode (sustain or no sustain), and Effects routing and send level.

In a Special Setup, multiple Tones can be assigned to a Mute Group, which allows only one Tone in the group to sound at a time. This is perfect for obtaining realistic results with percussion sounds such as hi-hats and congas. There are eight Mute Groups provided, which makes for a lot of flexibility.

It can be extremely useful to have 61 different, independent, simultaneous sounds on a single keyboard. For example, if you only need access to one of many different sounds, as with a drum kit, this mode is excellent. This capability also is handy for special-effects sounds.

Only one Multi setup is possible at a time on the instrument, containing only one Special Setup (a second Multi and Special Setup can be stored on an optional data card), which is surprising. It seems Roland views the JD-800 as primarily a live-performance machine and figured no one would need more than one Multi setup. This is unfortunate, considering the current climate of high-powered, mega-memory, multitimbral synths.

VOICE ARCHITECTURE

At the core of the JD-800 are 108 ROM-based digital waveforms. Additional waveforms are available through the use of the instrument's waveform card slot, but it would be nice to have the option of loading your own waveforms via MIDI Sample Dump or SCSI. This would make it much more open-ended.

The JD's onboard waveforms cover a wide range of sounds. Roland gives you 16-bit sampled sounds (sampled at 44.1 kHz), including percussion, piano, stringed-instruments, and various vocal waveforms, as well as basic waveforms such as square, sine, and sawtooth (with variations). Many of the acoustic instrument waves are better for creating the impression of these instruments than for exact simulation, but sample-playback is not the focus of the JD-800; the point of the instrument is to create new sounds. Some of the waveforms suffer from conspicuous loops, but these are rarely noticeable, and on the whole, the waveforms prove useful for programming.

A multimode (highpass, lowpass, and bandpass), 24 dB/octave filter with resonance is one of the synth's more powerful aspects because it allows the basic character of the waveforms to be drastically modified and/or disguised. This lets you get lots of mileage out of each waveform. I hope more manufacturers take Roland's lead and reintroduce this type of filter as standard equipment; it really makes a difference.

The JD-800 has three dedicated envelope generators, one for the TVA (Time-Variant Amplifier), one for the TVF (Time-Variant Filter), and a Pitch EG. The TVF and TVA envelope generators are variations on the traditional ADSR design, but with independent Rate and Level controls for each stage.

The Pitch envelope generator provides a 3-stage (Attack/Decay/Release) envelope. Velocity can be routed to envelope depth or attack time (with positive or negative values). A Time Key Follow feature tracks the keyboard position and varies the decay and release times accordingly. For example, positive values produce shorter notes at higher pitches.

There are two independent LFOs, each of which can be routed to filter cutoff, oscillator frequency, and amplitude. The LFOs are equipped with the usual Rate and Delay parameters but also have some nice surprises. There are five possible LFO waveforms: triangle, sine, square, sawtooth, sample-and-hold, and "smooth" random (which sounds like a justified sample-and-hold). A Fade parameter allows the LFO's depth to change over time. The Fade parameter, like many other parameters on the JD-800, has both positive and negative modulation. You can, for instance, begin with LFO 1 at maximum depth, and gradually decrease LFO 1's depth while the other LFO (set to a different Rate and modulating a different parameter) fades in.

EFFECTS

The JD-800 has a built-in, programmable, multi-effects processor whose settings are stored with each Patch. The Effects section is divided into two sections, Sequences A and B. Sequence A contains a nice phase shifter, a distortion unit, an enhancer, and a 6-band, fixed-frequency (250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz, and 8 kHz), graphic EQ algorithm called "Spectrum." Sequence B offers chorus, delay, and reverb.

The effects can be routed in different orders, though Sequence A always feeds its signal to Sequence B. Few instruments with onboard effects allow this degree of flexibility. The effects don't provide many parameters, but they sound very good and go a long way toward adding a polished, produced quality to the JD-800's Patches.

In Multi mode, you just have Sequence B and don't have the enhancer/phaser; you can choose reverb only, chorus plus reverb, delay plus reverb, and dry signal. If your Single sound depends on phasing, you'll lose that in Multi, which is a drawback.

Roland has provided seven types of distortion: Cry Drive, Overdrive, Mellow Drive, Light, Fat, and Fuzz. The only controls are Drive and Level; it's too bad a dedicated tone (EQ) control was omitted. The distortion algorithms are metallic, thin, and edgy, a sound preferred by many metalheads and modern rockers. My taste runs to fat, analog distortion, so the JD-800 won't make me put away my ProCo Rat.

The Reverb section sounds surprisingly good, with a fair selection of algorithms: four Halls, two Rooms, Gated, Reverse, and Flying Reverb. The latter is a kind of reverse, gated effect that pans from left to right (Flying 1), or right to left (Flying 2). You can get up to 120 ms of pre-delay, depending on the algorithm you choose, and up to 20 seconds of reverb time.

On the down side, there is no provision for real-time control over the effects, which is somewhat limiting. The Effects section can be accessed only through the cursor keys, not through the sliders.

MIDI

The JD-800's MIDI implementation is

adequate. The instrument responds to six incoming MIDI channels when in Multi mode, which is good for using the JD with a sequencer. Five of these are used for Parts 1 through 5, and the sixth is reserved for the Special Part.

Each Patch can transmit on two different channels. Since this is done by dividing the keyboard into upper and lower halves, each half can be mapped to send a different Program Change command that is stored with the Patch, an extremely useful live-performance feature. The JD-800 can save and load its data (all types) through MIDI System Exclusive, which means it can be fully supported by universal editor/librarian program templates.

Every slider on the JD-800 can transmit its movements over MIDI. Real-time parameter change is a great facility for an instrument like this, but the way Roland has implemented this feature seems clumsy: For each increment of any slider on the instrument, an 11-byte SysEx message is transmitted, which is long-winded and quickly devours both MIDI bandwidth and sequencer memory. You can switch off the sliders' SysEx transmission, but then they send nothing. This makes it tough if you use the sliders as an expressive device in a MIDI mixing situation.

SOUNDS GOOD TO ME

The JD-800 sounds great. It also is a lot of fun to program, which makes for an enjoyable instrument overall. Although the factory presets seemed designed to prove the instrument's potential as a commercial synthesizer (with such Patch names as Millennium, Fantasia 90's, and Crystal Rhodes), the instrument proved capable of interesting timbres, both simple and complex.

The ability to work with multiple Tones simultaneously provides lots of power and flexibility. Its sliders make the JD-800 instantly malleable, both when creating sounds and during performance. Hopefully, musicians will take advantage of its instant accessibility, and other manufacturers will move toward this type of musician-friendly design.

The JD-800 is not cheap, so it may not turn up in every keyboardist's setup right away. The most significant aspect of the instrument is that it represents a "return to old values," both in terms of programmability and timbral quality, yet it can produce the full range of "contemporary" sounds. This approach makes the JD-800 unique and well worth the investment. ❂

Rocktron Intellifex Processor

By Peter Freeman

A new player brings quality and versatility to the digital effects party.

Best known for its Hush noise-reduction units, Rocktron has charged into the digital effects wars with its smartly designed Intellifex. The processor is capable of producing

DAN ESCOBAR

Rockton's Intellifex Digital Effects Processor offers five simultaneous effects and excellent audio quality.

five digital effects simultaneously, including 4- and 8-voice stereo chorusing, digital delay, reverb, noise reduction, ducking, and either flanging or pitch-shifting.

Rocktron uses three separate, 16-bit converters with 64-times oversampling to achieve a dynamic range in excess of 100 dB (with noise reduction). Most digital effects units time-share a single converter for all inputs and outputs; separate converters provide less distortion and greater dynamic range. Like many effects units, the Intellifex sums a stereo input to mono, generating stereo in the effects output through processing. The heart of the Intellifex's processing is a 24-bit Motorola 56001 processor.

I found the Intellifex's design quite intuitive. Editing functions are accessed through a group of three knobs (yes, knobs!) for function selection, parameter selection, and value adjustment, with a fourth knob for program select. This system quickly becomes second nature. What a relief to have real knobs for parameter control instead of those infernal up/down keys found on almost everything these days.

EFFECTIVE SOUND

The Intellifex's sound proved crystal clear, making it a pleasure to use in a variety of musical contexts. The unit's strongest suit seems to be delays and chorusing effects; these sounded positively sparkling.

The delay section is precise-sounding, of extremely high quality, and flexible. The three programs (Stereo, Ping-Pong, and 2-Tap) produce a wide range of sounds, from complex imaging effects to long repeats with lots of regeneration. Total delay time is 1.5 seconds, divided into 750 ms per side in Stereo Delay and Ping-Pong, and 1,500 ms per tap in the 2-Tap program.

The Chorus algorithm offers 4- and 8-voice chorusing with independent LFO, delay time (up to 418 ms), and regeneration for each voice. Chorus effects shimmered impressively, adding a sparkle to almost any source, particularly keyboards and guitar.

In the pitch-shifting department, the unit produces harmony intervals from one octave above, to two octaves below the input, with adjustment in 1-cent increments. (One-hundred cents equals one half-step.) The pitch-shifting is good for most applications, but don't expect it to perform like a higher-priced unit (e.g., the Eventide H3000) in this department; the quality of the shifted signal tends to be a bit on the watery side (i.e., slightly unstable) when shifting by intervals greater than about a fifth up or down.

The Intellifex's reverb algorithms are disappointing. I tested the reverbs with classic synths (including Prophet VS and Oberheim OB-X), guitars, and vocals. To my ears, the eight algorithms (Plate A and B, Room A and B, Hall A and B, Dual, and Stadium) sound somewhat metallic and unnatural, lacking in richness and depth. A respectable selection of reverb parameters is provided, including high- and low-frequency damping, independent left and right predelay, and gate time.

On the other hand, the Gated Reverb algorithm sounds good on a wide range of input sources, especially percussive sounds, making it a good complement to your "main" reverb. The algorithm includes an internal

Product Summary

PRODUCT:
Intellifex Digital Effects Processor
PRICE:
$1,149
MANUFACTURER:
Rocktron, Inc.
2870 Technology Dr.
Rochester Hills, MI 48309
tel. (313) 853-3055

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ● ●
EASE OF USE	● ● ● ●
EFFECTS QUALITY	● ● ● ◖
VALUE	● ● ●

SOUND-CHECKING THE ROCKTRON INTELLIFEX

Rocktron pitches the Intellifex as a fully professional processor, with specifications to match. Input and output levels reach +20 dBu, which is very comfortable for pro recording but a little low for broadcast. (Believe it or not, those guys expect an audio device to output +24 dBu.)

The claimed specs for THD+Noise (0.009%) and dynamic range (94 dB with the Hush disabled and 104 dB with it in) are some of the most impressive I've seen. Using a Panasonic EP-7722A audio test system, my associate William Orner and I confirmed that the Intellifex delivers on its claims. This is one quiet, clean unit.

Note that the specs apply to the digitized signal, but without effects processing. Any device introduces some noise and distortion in processing, and the Intellifex is no exception. Most manufacturers do not cite measurements through processing because the results vary with the effect and the setting of parameters.

The Hush noise reduction gives the Intellifex its beyond-16-bit dynamic range. The cost is an exaggeration of dynamics at very low levels. This is unlikely to be a problem in real life, considering that expansion only kicks in at 70 dB or more below peak.

The frequency response is listed (and confirmed) as 10 Hz to 18 kHz (+0.5, -3 dB), a touch lower than the "20 to 20k" figure usually cited as full range. It may be that Rocktron chose this response to minimize high-frequency noise. In any case, it's not likely to affect real use, and the extra octave at the bottom could prove beneficial.

—*Gary Hall*

threshold control that is useful with a repeating signal (such as a drum machine). The decay of each successive input signal is subtly cleared, so when an input signal opens the gate, you won't hear the remaining decay portion of the previous signal. This helps retain clarity, especially in mixing.

The Ducker and Hush sections are quite useful. A ducker suppresses a signal's level, depending on the level of another signal. In the Intellifex, the effects output is ducked by the input signal, reducing the delay or reverb level while a phrase is being played. When the phrase ends, the output returns to full level. This provides more overall clarity but with the full decay audible between phrases.

The Hush section is effective and a welcome feature. This digital version of Hush noise-reduction is configured to reduce input noise and works by "downward expansion," reducing output gain in proportion to signal level. The Expander Threshold is set 5 to 20 dB above quiescent noise. At normal levels, noise is masked by signal. Once signal level drops below threshold, the processor begins reducing output gain at a ratio of a little more than 2:1. In the absence of input, any noise is effectively suppressed.

CONFIGURATIONS

Effects can be combined and routed in any of six ways, called "Configurations." The six Configurations are: Hush/Chorus/Delay/Reverb, Hush/Reverb, Hush/Delay/Ducker, Hush/8-Voice Chorus/Delay, Hush/Pitch Shift/Delay, and Hush/Pitch Shift/Delay/Reverb. The Ducker section also is available in some Configurations (such as the first and last listed) that don't name it in their titles.

These Configurations provide a fair degree of flexibility, and you can bypass an unwanted effect within a Configuration (e.g., if you don't want Chorus in the Hush/Chorus/Delay/Reverb Configuration, you can bypass it). Still, it would be nice to be able to define your own Configurations instead of being limited to presets.

MIDI AND MORE

The Intellifex boasts an impressive MIDI implementation. Of 160 programs, the first 80 (the user programs) allow simultaneous MIDI control of up to eight parameters, in real-time, with programmable scaling (upper and lower limits) for each parameter. In addition, presets can be mapped to MIDI program numbers. As you can imagine, this makes for interesting possibilities, especially in live performance and mixing.

In practice, the unit responds smoothly to MIDI parameter changes, producing musical, lively effects when controlled by a MIDI sequencer. This is one of the Intellifex's most impressive features, greatly extending the unit's usefulness and versatility. A rear panel 7-pin DIN jack connects to and supplies power for Rocktron's MIDI Mate controller pedal, providing remote program and parameter changes.

Rocktron is to be commended for achieving its goal of exceptional sonic purity. On the test bench, the unit lived up to its claimed specs (see sidebar "Sound-Checking the Rocktron Intellifex"), and it sounds great.

At a list price of $1,149, the Intellifex is not cheap, but in terms of pure sound quality, design, and flexibility, it clearly outshines many other digital effects processors. If you are looking for an extremely clean, versatile processor and can pay for top quality, this may be the one.

Peter Freeman *is a freelance bassist/synthesist and composer living in New York City. He has worked with John Cale, Chris Spedding, Hipsway, Jon Hassell, L. Shankar, Sussan Deihim, and Richard Horowitz.*

Bars&Pipes Professional 1.0c

By Todd Souvignier

The Amiga finally gets a high-end, multimedia-oriented sequencing package.

The release of NewTek's Video Toaster (reviewed in the July 1991 issue) caused a resurgence of interest in the Commodore Amiga family of computers. Many who had never before considered the Amiga, or had written it off as a cult computer or games machine, now take

Advertiser Index

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c. "Fundamentals of MIDI Mixing," p. 58	709	710	711	712
d. "Vector Programming, Part 1: Korg Wavestation," p. 68	713	714	715	716
e. "The Speaker of the House," p. 76	717	718	719	720
f. "DIY: Fun Under Pressure," p. 80	721	722	723	724

BARS&PIPES ADD-ONS

Bars&Pipes was intended as an open-ended, expandable system. That expandability comes in the form of various optional disks ($59.95 each). Many of the add-ons have been around for awhile and are not included in the purchase price of *Bars&Pipes Professional*. They're still worth mentioning, however, as they were not available when the original *Bars&Pipes* hit the streets, and they add to the gestalt of either version of *Bars&Pipes*.

Internal Sounds Kit. This comes with over 80 8-bit samples for the Amiga's internal sound chip. The AmigoPhone tool lets you assign any of these IFF sounds to a *B&P* track. The Spare Keys tool, also included in the Internal Sounds Kit, puts a little 2-octave keyboard onscreen that can be substituted for MIDI input on any track. This lets you audition sounds or enter parts by mouse-clicking on the mini-keyboard. You also can enter notes on the bottom two rows of keys on the Amiga keypad and see them reflected on the Spare Keys keyboard. This is a nice package for those who haven't made the investment in MIDI gear but still want to get into sequencing.

Rules for Tools. C programmers who want to develop their own modules will be interested in this disk. The manual repeatedly asks people to send their home-brewed tools and utilities to The Blue Ribbon Soundworks.

Music Boxes A and B. These two disks contain extra tools, too numerous to list here, with strong emphasis on algorithmic toys. Also coming soon: The Creativity Kit and the Pro Studio kit ($69.95 each), two new tool collections currently being beta-tested. The Pro Studio kit features a hip new "Feel Factor" tool (sometimes called "humanizing," this term refers to an attempt to add "live" feel to quantized sequences) and a gadget that translates MIDI Volume messages into note velocities.

Multi-Media Kit. The goodies on this disk come in handy when using *B&P* to add soundtracks to your Electronic Arts *Deluxe Video III* animation, Elan *Performer* presentation, or other multimedia work where music and image need to be synchronized. The *ARexx* accessory (also included in *Bars&Pipes Professional*) lets you slave *B&P* to other applications via *ARexx*, a utility that transports data between programs. The Multi-Media kit also features recorder and player modules that let you use your songs without loading *Bars&Pipes* and a tool that converts MIDI notes to *ARexx* messages.

the Amiga seriously. Although it still lacks the installed user base and vast software library of the Mac and IBM PC, the Amiga has a second lease on life.

The timing couldn't be better for new Amiga software releases, and The Blue Ribbon SoundWorks, Ltd. (formerly Blue Ribbon Bakery) is right there with the release of *Bars&Pipes Professional*. Evolved from their previous Amiga sequencer, *Bars&Pipes* (reviewed in the April 1990 EM), *Bars&Pipes Pro* boasts a host of new features and enhancements and eliminates much of the instability and several of the amateurish aspects that blemished the earlier release. (Incidentally, until December 15, 1991, Amiga music-software owners can upgrade to *Bars&Pipes Professional* for $199 plus the cover page from any Amiga music-software manual.) Since many of *Pro*'s basic functions are essentially the same as in the original *Bars&Pipes*, I'll focus on new features and changes.

THE WORKING PRO

The *Bars&Pipes* user interface has been cleaned up in *Pro*, and the main screen has been reorganized (see Fig. 1). Transport and edit controls now are professional-looking, rectangular, "Chiclet"-type buttons, as opposed to the jagged polygons of the original *Bars&Pipes*. The trade-off is increased

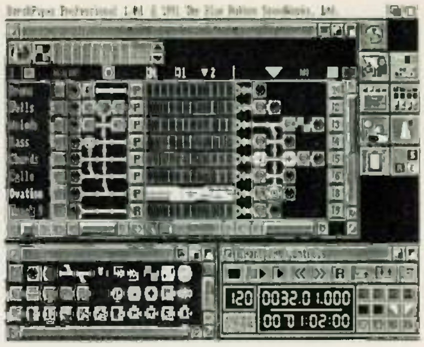


FIG. 1: Bringing objects from the Toolbox into the Pipeline on the program's main screen is a simple click-and-drag operation.

screen clutter. The transport controls are now in a box that never really knows where it should be. No matter where you put it, it's covering up something, and as soon as you close it, you invariably need to go back in. Still, this is a much better-looking program than the original *Bars&Pipes*.

As with its predecessor, the primary feature of *B&P Pro* is the flexible, graphics-oriented sequencing environment, which uses onscreen icons (graphic representations) to help you visualize the flow of data. The track sheet ("Pipeline") dominates the screen, as *Bars&Pipes Pro* allows a virtually unlimited number of tracks. Unlike the original *Bars&Pipes*, *Pro* makes a workbench icon for each songfile, which certainly makes file management easier for "mouse babies." Unfortunately, the icons are screen hogs (fully twice the size of the songfile icons made by other Amiga sequencers), which creates screen clutter.

TOOLING UP

Most large-scale operations are accomplished using "tools." These fall into three general categories, grouped according to function. Tools that route

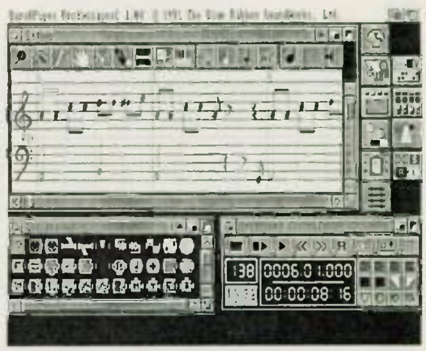


FIG. 2: An open Track Edit window, showing the standard notation option.

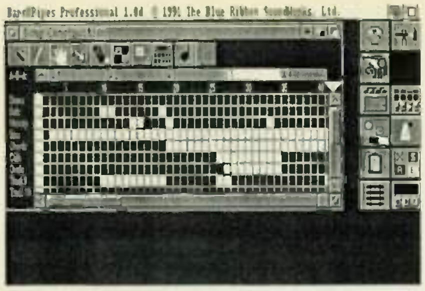


FIG. 3: The Song Construction window facilitates global changes within reiterating sections and provides a good visual reference.

data within *Bars&Pipes Professional* include the Input and Output, Branch and Elbow, and Feedback In and Out tools. Some tools correspond to conventional sequencer features, e.g., the Quantize, Note Filter, Transpose, and Modulate tools. Finally, there are algorithmic tools, including Reverser, which plays a sequence backwards; Phrase Shaper, which adjusts track dynamics; Inverter, which acts like a note-number compressor; and the Harmony Generator.

Many tools can be used either on the input or output sides of the Pipeline. When placed on the output side, tools affect a track's output, while leaving the recorded MIDI data untouched.

A flexible Loop tool is the only new tool that comes with *Bars&Pipes Professional*. It allows independent looping of entire tracks or sections of tracks. Loops can begin at a predetermined start time or can be triggered by incoming note events. Looped sections can be transposed or modulated. More exotic tools are available in the Bars&Pipes Add-On Series of accessory disks (see sidebar "*Bars&Pipes* Add-Ons").

EDITING FEATURES

Graphic note-editing now comes in three flavors: piano-roll, standard notation (see **Fig. 2**), and a staff-hybrid in which notes are seen as bars on a conventional treble and bass clef. Trained musicians may be glad to see standard notation editing; to my knowledge, this is the first time the feature has been available in an Amiga sequencer.

It's a particular joy to be able to make a change on the piano roll and see it reflected in the staff. This free-and-easy interplay is just what I want out of graphic editing. However, you can push it too hard by clicking fast through a number of operations. It just doesn't seem to calculate and refresh fast enough.

Event-list editing is a somewhat under-implemented option in *Bars&Pipes Professional*. Changing event parameters is accomplished by clicking on the desired value and dragging the slider at the top of the window. One feature notably missing here is the option of choosing which types of MIDI events to display, even though that option is available in the graphic-editing mode.

What was called "A-B-A" in the earlier release of *Bars&Pipes* appears in an enhanced form as the Song Construction window in *B&P Pro* (see **Fig. 3**). The window now displays every active track and provides a nice graphic reference showing, measure-by-measure, which tracks have MIDI activity in them. This is a handy feature when doing music with repetitive structures. Once "verse-chorus-bridge" sections have been defined, a change made in one iteration of a section can be propagated to all other occurrences of that section. If you want to switch verse and

Product Summary

PRODUCT:
Bars&Pipes Professional 1.0c

REQUIREMENTS:
Amiga 500, 1000, 2000, 2500, or 3000 with 1 MB of RAM (2 MB recommended); MIDI interface; one or more MIDI instruments; hard disk recommended

PRICE:
$379

MANUFACTURER:
The Blue Ribbon SoundWorks, Ltd.
1293 Briardale NE
Atlanta, GA 30306
tel. (404) 377-1514

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ● ●
EASE OF USE	● ● ●
DOCUMENTATION	● ●
VALUE	● ● ● ◖

bridge, add an extra chorus at the end, or make other large-scale structural changes to a composition, the Song Construction window facilitates it.

The Tempo Map window also has been enhanced. What was largely a text-driven operation in the earlier release is now accomplished graphically. Tempos can be changed instantaneously or rise and fall along linear or exponential curves. It's as simple as specifying the curve you want, picking up the pencil tool, and defining the end points of the line. Another nifty new function is the Conform option, which adjusts the tempo of a song section so it fits within a specified time.

Also new and noteworthy is the Mix Maestro window (see **Fig. 4**). It gives you real-time control over the muting, pan, and MIDI volume of each active track. Just open the mixer window and start your sequence rolling. As you drag the volume and pan controls, continuous controller data in the track are overridden by the new values. Once you've run a mix on a track, you'll see the sliders automatically move up and down in real time on subsequent playbacks. This window makes it easy to build a mix, one track at a time. You can always go back and change a track's mix, save a mix and try another one out, or bypass the module entirely. Remember, your synthesizer or sampler must respond to MIDI Continuous Controller messages for this module to work.

MULTIMEDIA

Bars&Pipes Pro is particularly well-suited for desktop multimedia applications. The program multitasks smoothly, provided you have adequate RAM, and includes an *ARexx* accessory (also found on the Multi-Media option disk; see sidebar) that permits control of song-playback from other Amiga applications. This facility for getting sequences to sync with animation or slide-show programs is unparalleled.

Time-Line Scoring is a completely new feature that is of special interest to people working on film or video scores. It allows you to load multiple *Bars&Pipes* songs into memory and freely define their relative timings so they play sequentially, simul-taneously, or in overlapping order. Synchronization to external MIDI Clocks and Song Position Pointer is solid in *Bars&Pipes Professional*, and the pro-

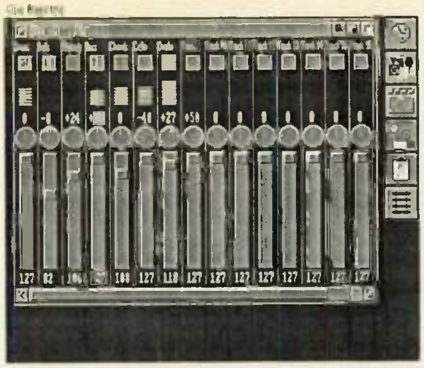


FIG. 4: The Mix Maestro screen allows you to mute, pan, and fade each track in a sequence and provides automated mixing capability.

gram includes an accessory supporting Dr.T's Phantom MIDI and SMPTE interface.

NOTATION

Although the ability to transcribe and print out scores is a highlighted feature, *Bars&Pipes Professional* is not a full-featured scoring program. Think of it instead as a sequencer that can print out a rough hardcopy. While transcription in this program is relatively quick and adequately accurate for applications other than music publishing, it is sorely lacking in terms of user control of score layout. Even though a song may have a repetitive structure, the print utility doesn't realize it. Scores are printed out linearly; you don't have the option to include repeats and codas. Fortunately, you have the option to transcribe and print only certain tracks and sections of tracks. The quality of its bitmapped output can vary greatly, depending on the printer used, but it's safe to say that this program is not intended for engraving applications.

PRACTICAL CONSIDERATIONS

The 1 MB of RAM recommended by Blue Ribbon is a bare-minimum requirement; 2 MB and a hard disk is more like it. By unloading most of the tools and accessories and using "Memory Saver" options (such as going to a gray-scale screen and eliminating the Workbench), you can clear enough RAM for basic recording and editing on a 1 MB machine but not enough for printing.

A program with this many unusual features needs accurate, understandable documentation. The manual does a good job of explaining the program in clear, coherent terms and is loaded with screen dumps that help illustrate most of the functions. Unfortunately, the manual has several problems, most notably in the indexing and proofreading.

Everyone's needs are different. If you're only interested in cranking out quick MIDI demos, with a minimum of mousing around, this may be too much program for you. If you like Blue Ribbon's design philosophy, consider the current version of the regular *Bars&Pipes* (which lists for $199). If you're into a very directed, straightforward, "I-know-what-I-want" style of sequencing, *Bars&Pipes Professional's* unique features may seem like superfluous bells and whistles. If you're in search of a full-featured scoring and notation program, you'll need to look elsewhere.

But this is the only Amiga sequencer that supplies an object-oriented programming (OOP) environment, so if you've waited patiently for OOP, your vigil is over. Also, if you're interested in using algorithmic tools in a sequencing environment, are looking for a sequencer with an open architecture, or want your sequences running behind computer animation, this unique program may be the right tool for you.

(Special thanks to Commodore Business Machines for loaning an Amiga 2000HD.)

Todd Souvignier *is operations manager of the Mix Bookshelf and plays bass with San Francisco hard-rockers Bangkok Cocktail.*

Roland Studio M Music System

By Charles R. Fischer

Good things can indeed come in a small package.

It's amazing how musicians work for years to buy their dream setup, only to discover that it's a nightmare to transport, set up, and tear down. Suddenly, you begin to dream of a lightweight, portable unit that would replace half of your working rig. You could walk into gigs or sessions with your machine, connect a MIDI controller and amplifier, power up, and play.

JULIAN OKWU

Roland Studio M Music System.

Sometimes, dreams come true. The Roland Studio M (a.k.a. the MV-30) includes a high-performance sequencer, 8-part multitimbral tone module (including percussion), automated MIDI mixing, and a pair of signal processors in a convenient unit the size of a small VCR. For $2,695, you can replace a rack full of sequencers, tone modules, and audio equipment, saving a lot of hassles and time.

Does this sound like the answer to your prayers?

WHAT YOU SEE

The advantages of the Studio M's design were obvious from the time I took it out of the box. The large, detailed graphic LCD comes to life with an impressive power-up routine. I often am frustrated by instruments with inadequate displays, and I would dread using a Studio M-type product with the typical 16 × 2 display. Roland earns my gratitude here.

The front panel features eight data sliders for the Compu-mixer section, a master volume slider, a large assortment of function and parameter buttons, and an alpha dial for data entry. On the back are two slots for sound cards (compatible with the SN-U110, MV-30, and D-70 cards); MIDI In, Out, and Thru ports; three stereo pairs of audio outputs (one set of Mix outs and two sets of Direct [dry signal] outs); a metronome output with level control; tape sync in and out; a footswitch jack for punch-in and sequencer start/stop; and a jack for the external power supply.

To get into a specific function, simply press the appropriate switch, and the display shows you the options on a menu. Next, hit one of the five function buttons under the display, and you're there. Some users might be deterred by the number of buttons, but I think most will prefer dedicated parameter buttons to endless sub-pages or multiple keystroke sequences that must be memorized.

Sequences and programs are stored on 3.5-inch floppy disks, and the operating system also loads from disk. I prefer ROM-based op systems for live-performance devices because they boot (or reboot, if there's a power failure) faster and more reliably. (On the other hand, floppy disk-based op systems are more easily upgraded.) Users also will need to keep backup copies in case a disk is damaged or erased.

THE SEQUENCER

The Studio M sequencer is similar to the Roland MC-50 Microcomposer (discussed in the April 1991 "EM Guide to Hardware Sequencers"), with a few added features. You can port a number of file types to the Studio M, as it loads MRC-500 files, Super MRC files, Director-S files, and W-30 files and loads and saves Standard MIDI Files. The timing resolution is 96 ppqn, the same as most hardware sequencers today. The memory holds approximately 50,000 notes but holds fewer notes if you record Continuous Controller data. Roland promises a memory option to increase note memory to around 120,000 notes; check with a Roland service center about availability. A Chain Load feature is provided for live performance.

Of the sequencer's eighteen tracks, eight are dedicated to the internal voices, and eight control outboard MIDI equipment. One track controls tempos, and the last track carries Compumixer data.

Product Summary

PRODUCT:
Studio M (MV-30) Music Production System
PRICE:
$2,695
MANUFACTURER:
Roland Corporation
7200 Dominion Circle
Los Angeles, CA
90040-3696
tel. (213) 685-5141

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ● ●
EASE OF USE	● ● ● ● ◖
AUDIO QUALITY	● ● ● ●
VALUE	● ● ● ●

A variety of methods are available for recording sequences, including real-time and step-time recording. (Real-time entry requires an external MIDI controller.) You also have a choice between (or can combine) Pattern and Track sequencing. Pattern sequencing involves putting together a complete song out of a number of shorter sequences, while Track sequencing uses a single sequence for the entire song.

Both methods use a set of editing tools that bring the sequencer's power into the range of some software sequencers. In addition to the features we've come to expect, there are a few things the Studio M can do more easily than all but a few computer sequencers.

One example is the Change Velocity command, which allows you to compress or expand the values of velocity data on a track by an adjustable percentage. At the same time, you can add a variable offset that is added to (or subtracted from) the values. With this and other editing functions, you can specify which sections and notes are to be changed.

The Studio M also features Roland's "Microscope" editing function, used to view and change isolated events in a sequence. Between the Microscope and the facilities for editing larger sections of data, the sequencer is capable of handling the editing needs of most professionals.

However, some features could be improved. For example, the quantization function is destructive, i.e., it can't be undone if you don't like the results. The sequencer allows you to move the quantized data to another track, so you have the option of keeping the original performance, but what happens if you've already used them up?

A more subtle irritant is that the editing tools remain assigned to the last track on which you used them, even when you are working on another track. For instance, if you erase Track 1, record a new part on it, and start recording parts on other tracks, the next time you want to erase something, the erase function still will be set to erase Track 1, even though you've been working on another track. Some users might like this scheme, but I find it inconvenient, as it can be all too easy to erase the wrong part.

REAL-TIME PHRASE SEQUENCING

The Real-time Phrase Sequencer (RPS)

is especially useful for live performance, letting you call up patterns by playing the appropriate note from an external controller. A performer could use this feature to switch between song sections, to extend a solo, or repeat a hook as many times as the audience can stand it. Another possibility is to record a number of drum fills, making it easy to add fills on top of a basic riff. A factory disk contains a demo of some of the possible uses of this feature.

Though you'll have to decide how to arrange the patterns for optimum results, Roland has made this fairly easy. You can select the MIDI channel for RPS functions and map out which note number is used for each pattern. A Stop By feature allows you to stop the sequence using a selectable Note-on command, and there are several operating modes for each pattern. The modes determine whether the pattern triggers (plays once) when the specified Note-on is received or plays through as long as you hold down the key. Triggered response is especially useful with drum controllers.

Some may dismiss the RPS as a gimmick, but it has the potential to bring excitement and spontaneity back into sequenced performances. It could become next year's cliche, but I hope a few artists take the time to explore the serious uses of this feature.

TONE MODULE

All the features in the world won't help if the onboard sounds don't measure up. Fortunately, the internal sound module is one of the Studio M's strongest features. With up to 8-part multitimbral operation and 30-voice polyphony, it offers enough sonic resources for many applications.

The quality of the internal sounds isn't surprising, as the voice structure is similar to that of Roland's D-70 synthesizer (reviewed in the September 1990 **EM**). The sounds are clear and crisp, although a few suffer from stretching a single sample to cover a range better handled by several multisamples.

The voice structure includes Tones (which consist of either one or two RS-PCM multisamples), processed by a TVF (Time-Variant Filter) and a TVA (Time-Variant Amplifier). The TVF resembles the voltage-controlled filters found on analog synthesizers, with multimode operation and variable resonance. As a result, the Studio M can produce good analog synthesizer imitations.

Each voice includes two envelope generators (dedicated to the TVF and TVA) and an LFO. MIDI performance controllers (Pitch Bend, Velocity, Mod Wheel, and Poly and Channel Aftertouch) can be used to modulate voice parameters such as filter cutoff, envelope time, and volume.

There are 220 sampled sounds aboard the Studio M. While the samples are stored in internal ROM, the programs must be loaded from disk along with the operating system and song files. Additional samples are available through the two card slots on the rear panel.

The sounds include synths, orchestral instruments, and acoustic and electric keyboards, guitars, and bass-es. There are two variations or more of many instruments. Typically, this in-cludes a regular and a doubled (cho-rused) variation; just remember that the doubled sounds eat up available voices twice as fast.

The drum sounds are among my favorite parts of the Studio M. These offer many of the same parameters as the other voices. Each drum sound in a kit can be tweaked individually. For example, you can specify which drums will get reverb or chorusing, or whether they are routed to the Mix or the Direct outputs. Because the drums use the TVF and TVA, you have considerable control over the sounds. I hope Roland will put the drum section in a dedicated module someday; it would make a great product by itself.

COMPU-MIXER

The Compu-mixer is one of the special features that make the Studio M more powerful than the sum of its parts. It controls the level, panning, and output assignment of the internal sounds and sends MIDI Volume and Pan messages to external sound modules. Best of all, you can make changes to these parameters and record them into the sequencer, giving you a usable automated mix.

Using the Compu-mix is a blast, thanks to the slick graphics provided by the display. The eight data sliders set the volume and panning of the various parts, and the movements are portrayed graphically on the display. It's more entertaining than Nintendo.

The resolution of the sliders is reasonably good, though it is possible to add zipper noise if you get carried away. The sequencer provides a command to reduce the amount of mixer data stored (saving sequencer memory) and a command for converting mixer data into MIDI Volume and Pan messages and vice-versa.

The Compu-mixer is useful as long as you're aware that it is not a general-purpose automated mixer. It will not mix external audio signals, only internal sounds and MIDI-controlled sound generators, and it deals with level, pan, and output assignment only. (If you intend to use it with external modules, remember that all instruments must respond to MIDI Volume and Pan messages.)

EFFECTS, SYNC, AND DOC

The Studio M features two onboard effects processors. One covers reverb (including gate) and delay, and the other produces chorusing (two types), flanging, and slapback.

Reverb algorithms include three rooms and two halls, and you can adjust reverb time and level. Delay parameters include delay time, level, depth, and feedback. There is a cross-delay algorithm that alternates between the hard left and hard right pan positions. Flanging, slapback, and chorusing algorithms also have a full complement of parameters. Unfortunately, the processors cannot respond to MIDI controllers in real time.

Both processors have better-than-average sound quality. You also can use your favorite outboard processor, sending the dry signal from the Studio M's direct outputs.

The Studio M features what Roland calls "Tape Sync II," better known as "Smart FSK." Tape Sync II retains the simplicity of FSK sync but allows the sequencer to synchronize starting at any point on tape. In this, it is functionally similar to SMPTE time code.

The Studio M is engineered to minimize the learning curve. Any product with these capabilities requires some time and effort to master, but the job is much easier with a good human interface. You'll still need the two user's manuals, though. One is a "quick start" guide for jumping in right away, while the second is a reference manual that covers each feature in detail. (Roland also plans to publish a supplemental guidebook, *Up and Running With the MV-30.*) Both are well-written, though I found a few instances of vague wording and some amusing "Japanese English." The reference manual badly needs an index, as well.

WHO NEEDS IT?

I find the Studio M impressive and a worthwhile investment for any musician using a sequencer in live performance. (It's great for the two-singers-plus-sequencer acts on the lounge circuit.) In addition, it's a terrific toolkit for composing and producing demos. With a MIDI controller, a small multitrack, and a monitor system, it would make a fine composer's studio (and so compact). For Roland, the MV-30 Studio M is a big step in the right direction. ❂

Soundcraft Spirit Studio

By Neal Brighton and Michael Molenda

A full-featured mixer nudges homebodies closer to the charts.

Producing master-quality recordings in a home studio sometimes is a question of money, rather than love. Our CD society demands expansive sonic landscapes, and capturing these often requires expensive professional equipment. The ambitious musician is caught in a creative Catch-22: Continue making semi-pro demos, or take the risk of deconstructing a comfortable home environment into a debt-ridden "pro" studio in a house.

Luckily, some manufacturers now offer professional quality and features in equipment priced for the home-studio market. Soundcraft's Spirit Studio 16×8×2 mixer (also available in a 24×8×2 version) is such a product. At $3,995, the Spirit prevents fiscal nightmares from sabotaging the muse, while its professional configuration promises a massive upgrade in recording quality.

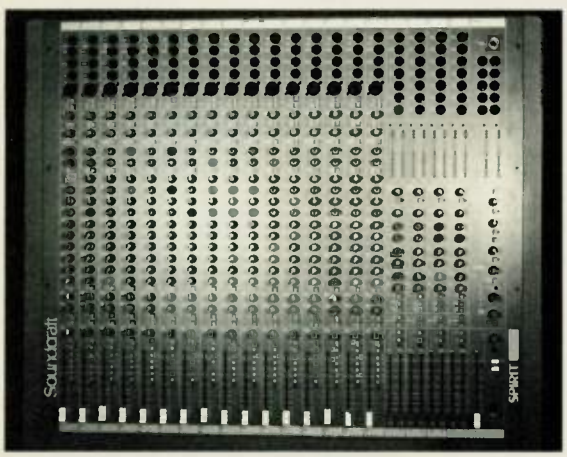
SETTING UP

Thoughtful layout spares the Spirit's wiring from being the usual exercise in terror. All sends and returns are easily connected via the top panel, rendering a patch bay superfluous. A fully loaded system is a visible jungle of wire, but the luxury of accessible patch cables more than compensates for the mess. We completely integrated the Spirit into our studio within a half-hour.

All channel line inputs are *balanced*, ¼-inch phone plugs, which caused initial concern about plugging in unbalanced lines from keyboard modules and signal processors. Applying a dash of rock 'n' roll attitude, we connected everything anyway, and all lines worked fine. Soundcraft provides users the option of utilizing balanced lines to diminish hum and other line noises, but we used unbalanced lines on short runs to ensure minimal noise. The Spirit's insert points and aux sends are unbalanced.

To facilitate interfacing with professional and semi-professional equipment, the Spirit offers some user selectable options. The tape sends and 2-track returns are factory set to -10 dBV (semi-pro), but they can be changed to +4 dBu (pro) output by removing resistors. This is a relatively simple matter, and the manual illustrates the operation clearly. The bargraph LED meters are factory-set to monitor peaks, but average signal level readings can be tracked by resoldering PCBs (definitely a job for a competent audio technician).

Where you place the board becomes an important consideration, as the Spirit's non-modular design requires the entire mixer to be opened for servicing. (Modular designs, though more expensive, allow removal of the faulty module.) Service access is gained by removing screws on the top *and* bottom of the Spirit, so the board must remain innocent of built-in consoles and other confinements. This is disappointing for home recordists who want to build customized "mixing desks" around their boards. In addition, the control panel is not angled within its casing, which makes viewing of some knobs and markings difficult from a sitting position. Normally, you could rectify this ergonomic *faux pas* by inclining the bottom surface of a custom console. However, it's a small (and often welcome) inconvenience to lift oneself out of a chair every once in a while to tweak a "hidden" knob.



JULIAN OKWU

The Spirit Studio offers a wonderland of features to enrich the home studio environment.

To avoid a chronic stiff neck, be sure to position your multitrack recorder where input levels can be comfortably viewed from the mixing position. The absence of a meter bridge on the Spirit necessitates frequent checking of your tape deck meters.

We discovered the hard way (ouch) that the Spirit's power supply runs extremely hot. Since recording sessions often run countless hours without a break, we were concerned the power supply might self-destruct under constant use. Soundcraft assured us the unit has been reliable and the design "tried and true." We still recommend placing the power supply where ventilation is optimum.

RECORDING

Traffic control on the Spirit is simple. Incoming signals can be sent directly to tape via the Direct button (see **Fig. 1**) or assigned to a subgroup by the usual pan pot routing: The left position services groups 1-3-5-7, right services 2-4-6-8, and center establishes a stereo send. Groups 1 to 8 are normalled to tape send jacks 1 to 8 and 9 to16 in parallel. This means a subgrouped signal on group 1 is simultaneously routed to tape sends 1 and 9; group 2 to tape sends 2 and 10; and so on. Selecting which track is recorded is done by enabling the appropriate (track) record function on the tape deck.

The Spirit's direct tape sends run at -10 dBV, while the subgroup outputs run at +4 dBu levels to facilitate a possible dual purpose as a live sound

Product Summary

PRODUCT:
Soundcraft Spirit Studio
16×8×2 mixer
PRICE:
$3,995 (24-channel model $5,650)
MANUFACTURER:
JBL/Soundcraft
8500 Balboa Ave.
Northridge, CA 91329
tel. (818) 893-8411

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ● ● ◖
EASE OF USE	● ● ● ●
SOUND QUALITY	● ● ● ●
VALUE	● ● ● ● ◖

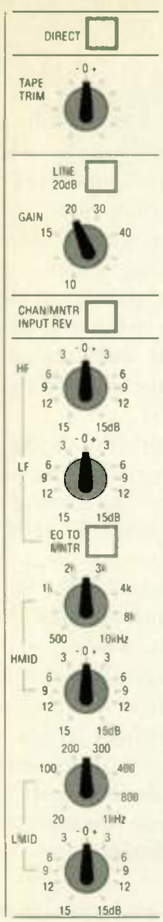


FIG. 1: The Spirit's 4-band EQ can be split between channel and monitor signals with the EQ to MNTR button.

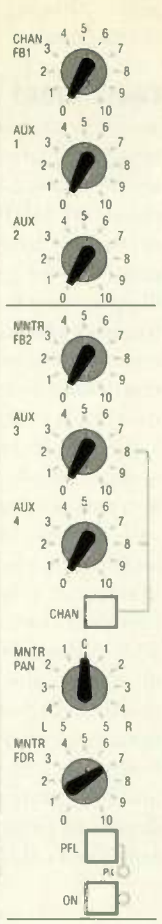


FIG. 2: Monitor signals return on the input channel via the Spirit's in-line design.

mixer. Connections are standard (1/4-inch for line inputs and 3-pin XLR for microphone inputs), and +48V phantom power is available on each individual input channel. We found all mic, line, and tape return amp gains flexible enough to accept a wide range of signal strengths.

The Spirit is an in-line design, which means the tape returns are monitored on the input channels. Each input channel has one dedicated pre-fader "foldback" send and two post-fader aux sends(see **Fig. 2**). Three identically configured aux sends assigned to the monitor section can be switched to the channel module by depressing the Chan button. The availability of six total aux sends eases the chore of setting up headphone mixes. For example, a MIDI percussionist can get a separate, ear-splitting level on a click track and bass, while a vocalist is simultaneously provided a moderate volume level of an "overall" mix, complete with reverb. An integral talkback mic, with its own volume pot, insures easy (and comfortable) communication throughout the headphone link.

The Spirit's headphone socket is switched, which means the main "control room" level is shut off whenever a plug is inserted. For the musician attempting to perform and engineer simultaneously, this feature is quite handy when open microphones are involved. No longer will your best vocal takes be ruined by excessive audio bleed because you forgot to kill the main speaker levels.

Considering the Spirit's professional profile, we weren't surprised it sent everything we recorded cleanly to tape. Soundcraft's obvious commitment to engineering professional capabilities into a budget mixer resulted in electronics that are absolutely pristine.

Because of this commitment, we *were* surprised to discover the Spirit offers PFL (Pre Fader Listen) instead of a more functional, albeit more costly, solo-in-place feature. Solo-in-place allows stereo signals (drum subgroups, stereo keyboard patches, etc.) to be referenced by depressing solo buttons on two appropriately panned input channels. Unfortunately, PFL is mono, so achieving a stereo "solo" reference on the Spirit requires muting all tracks except the ones you wish to monitor. That's a lot of buttons to push every time you want to check a stereo signal. This arrangement also revealed an idiosyncrasy of the mute buttons: They actually are channel on/off switches. Most engineers are accustomed to depressing a mute button to silence a track. On the Spirit, this action turns the channel "on" (audible). Turning the channel "off" (inaudible), requires popping the button *up*. Again, a minor irritation, but it took getting used to.

To be fair, most mixers in the Spirit's price range do not offer solo-in-place or true channel-muting. However, with all the thought Soundcraft put into versatility, it would have been nice to enhance the mixer's "studio" configuration by including these features.

MIXING

In our opinion, a mixer really shows its mettle during the mixing process. In this arena, the Spirit proves to be a good board with great features. Noise

SPIRIT STUDIO 16x8x2 SPEC TEST

	SOUNDCRAFT CLAIM	EM TEST
Bus Noise, Mix (Masters Down)	-98 dBu	-92.5 dBu
Bus Noise, Mix (Masters Up)	-86 dBu	-87.2 dBu
Bus Noise, Group (Masters Down)	-95 dBu	-95 dBu
Bus Noise, Group (Masters Up)	n/a	-86.5 dBu
Mix Noise	-80 dBu	-80 dBu
Microphone Gain, Terminated 150R	-129 dBu	-126.8 dBu
Distortion, Measured 1 kHz at +20 dBu, 20 Hz to 20 kHz	0.006%	0.006%
Crosstalk, Measured 1 kHz sine wave, Routing isolation (Mix L/R and Group)	100 dB	100 dB
Frequency Response, Measured 20 Hz to 20 kHz, Relative to 1 kHz	±0.5 dB	±0.5 dB

(Thanks to both Michael Gore of BASE and Gary Hall for system testing.)

and crosstalk are nil; signals that enter clean, exit clean (see sidebar). The in-line design allows tape *and* source returns for each of the sixteen input channels. Along with the four stereo effects returns, a total of 40 signal paths are available. That's an incredible amount of processing power for MIDI musicians who combine virtual sequencer tracks with analog tape tracks. Signal-routing headaches and cheap line mixers are now memories; your entire production can be mixed on the Spirit's modules.

One of the nicest features of the Spirit is the way it takes advantage of its multiple returns. The musician is not left with a bunch of lines and no way to process sounds. A Chan/Mntr Input Rev (**Fig. 1**) switch routes either the input source or the tape return (monitor) down the full channel module. This provides incredible flexibility in processing signals. For instance, you can assign virtual sequencer tracks to the monitor pot and reserve the full channel controls to tweak sounds on tape. If the sequenced sounds need a bit of help, you can even split the channel equalization to assign high-frequency (10 kHz) and low-frequency (100 Hz) control to the monitor section, leaving high-mid- and low-mid-frequency adjustment on the channel section. You also can assign the monitor section aux sends to the channel section, giving the module six aux sends for effects processing.

Opinions on the sound of a mixer's EQ provide textbook examples of subjective perception: One engineer's delight is another's damnation. On the objective front, Soundcraft has provided a budget board with powerful equalization options. The fact that a basic 4-band design can be split between two separate audio paths and maintain generous, swept mid-band control is a tribute to Soundcraft's ingenuity. Given the inevitable budget compromises, the sound of the Spirit's EQ is above average for a board in this price range.

This said, we still perceive the Spirit's EQ quality as somewhat brittle for the demands of modern mixing. The EQ always delivered good sounds (although the human voice was problematic), but never quite matched the shimmer of the CDs we referenced. Also, the split-EQ function requires the high-mid pot to leap from 500 Hz to 16 kHz, which made it difficult to find certain frequencies among the crowd. Critical sound-sculpting demands the flexibility of the full 4-band EQ. Finally, there is no EQ in/out button. While this undoubtedly is a budget compromise, we would be willing to pay extra for the feature. Pushing a single button beats marking settings and returning all pots to flat (no EQ) positions every time you want to check your tone enhancement against the "virgin" sound.

When all the fader moves are over, it's important to compare final mixes to the marketplace. We constantly listen to appropriate (major label) CDs to test whether our audio landscapes match the world's hot producers and engineers. Unfortunately, the Spirit's lone 2-track return makes it difficult to critically compare a CD or cassette mix to your master. Some repatching is required to monitor a CD or cassette player through available line returns on the board.

CONCLUSIONS

Soundcraft has done a marvelous job bestowing pro power upon the home studio market. The Spirit embraces the needs of MIDI musicians and home recordists by offering a multitude of returns and processing options. Yet, despite pro features, the Spirit does not overwhelm the home-studio ambience. Patching is easy and accessible, controls are clearly labeled, and signal routing is well-defined.

Despite its budget profile, the Spirit is surprisingly close to delivering commercial studio quality (and some so-called "pro" studios could benefit by jettisoning their current mixers and installing a Spirit, *post-haste*). Some compromises are unfortunate, but $3,995 doesn't buy the world. For the most part, the Spirit gives the critical home recordist the chance to produce master-quality recordings "from the bedroom." Few mixers promise this level of quality and flexibility. If you dream of being a home hitmaker, the Spirit is a valuable partner.

Sound & Vision studio owners **Neal Brighton** *and* **Michael Molenda** *produced "Step Back" and "Rev's Song" for Imago Records' newest act, The Sextants.*

Symbolic Sound Kyma System

By John Duesenberry

This composer's workstation could be the "wave" to catch.

The academic computer music scene has provided some of our most important tools, from FM synthesis to the *Max* visual programming language. (See "Max Programming Workshop" in the August 1991 **EM**.) The research from these elite labs only occasionally coalesces into products that are practical for earthbound musicians, so it's worth paying attention whenever a research group makes that step.

Composer Carla Scaletti and engi-

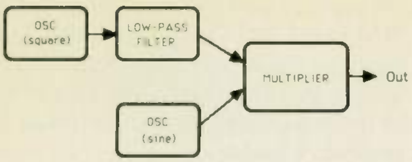


FIG. 1: The process of sound creation often begins with a block diagram of the desired patch.

neer Kurt Hebel have done just that. After pursuing their vision of a "composer's workstation" with the University of Illinois' CERL Sound Group, they've formed Symbolic Sound Corporation to manufacture and sell the Kyma System (pronounced "KEE-ma").

Kyma's designers are ambitious, bundling software synthesis, real-time processing, object-oriented programming, and hard-disk recording into one box (and it's MIDI-compatible to boot).

OVERVIEW

In Symbolic Sound's view, a composer's workstation is an environment for exploration and experimentation. The Kyma System emphasizes synthesis, algorithmic composition, and score processing.

The system integrates these activities under a common interface. The user works with objects, called *Sounds*, that look (and feel) the same throughout the system. This eases the transition from one level of thought to another. At one moment, you might attend to the details of a synthesis "patch"; later, you can work on an algorithm to generate an entire composition. Normally this switch would involve serious disruptions in hardware, software, and user psychology, but in the Kyma System it just takes a few mouse clicks.

The Kyma System has two components: the Capybara signal processor and a Macintosh program called *Kyma*. Capybara is built around several Motorola 56001 DSP chips (the same chip used in Digidesign's Sound Accelerator) and is entirely software-configured. It might serve as a reverberator for a hard-disk recording at one moment and as a MIDI-controlled synthesizer the next. Still later, it might be used to filter and ring-modulate an incoming signal. Conceivably, it could do all of these at once.

The Kyma software controls Capybara. Kyma superficially resembles synthesis programs such as Digidesign's *Turbosynth*, in that the user specifies synthesis and processing algorithms by manipulating onscreen objects. But Kyma does more; it's really a self-contained programming environment, including a score processor that works in the manner of *Music-N* (a family of languages prevalent in university computer music departments) and a powerful, general-purpose programming environment, *Smalltalk-80*.

The Kyma program does no sample computation. When commanded to "Play," Kyma generates and downloads a 56001 machine-code program to Capybara, which generates the actual samples.

MAKING SOUNDS

Let's build a simple Sound. **Fig. 1** shows the block diagram of a patch that amplitude modulates a sine wave with a filtered square wave, using two oscillators, one filter, and one multiplier.

Fig. 2 shows the *Prototype Strip*, which is Kyma's menu for selecting Sound objects. The system comes with a strip containing some 80 Sounds, grouped into categories listed in the vertically scrolling window on the left. The rest of the strip displays Sound icons from the selected category in a horizontal, scrolling window.

The objects in the Prototype Strip are *classes* of Sounds. For example, we've selected the *Oscillator* class from the category *Generators*. To work with an actual Oscillator, you must create an *instance* of that class. (A hardware analogy may help clarify the difference: A schematic describes a certain class of circuit. When you build a circuit from the schematic, you make an instance of the class.)

To create an Oscillator instance, drag its icon into a Sound File window. The window now contains a working Oscillator, with parameters set to default values. To hear the Oscillator, click on it and select the Play command from the Action menu.

Fig. 3 shows a Sound Editor window containing the realization of our AM patch. At the top is a Sound called

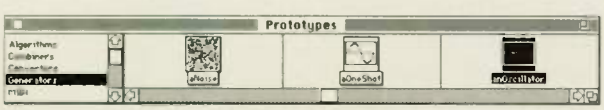


FIG. 2: Kyma's Prototype Strip provides access to a selection of Sound classes, organized by category.

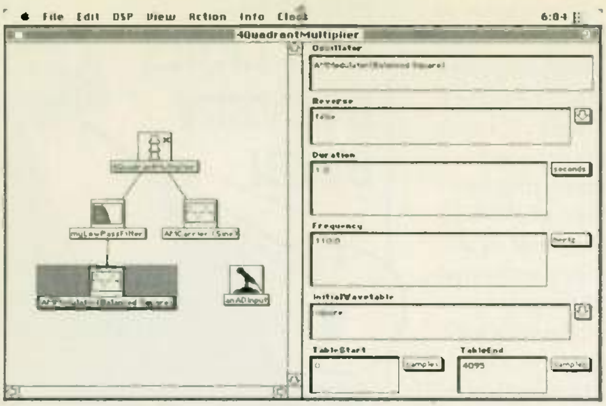


FIG. 3: The SoundEditor window shows the completed Sound and provides access to all parameters of each SubSound.

4QuadrantMultiplier, an instance of the class *Product*. The inputs of this object are *SubSounds* whose outputs are multiplied together. The multiplier's left input, the *LowPassFilter*, itself has an Oscillator SubSound as an input.

The Sound Editor accesses the top-level Sound and all of its SubSounds. The left pane displays the structure of the Sound.

You can listen to any sound component or use the right pane of the window to edit any of its parameters. In Fig. 3, an Oscillator has been selected, and the right side of the window shows the parameter settings that apply. Any parameter field can open into a full-screen text-edit window. Some fields feature popup menus or graphic sliders for convenient data entry.

You could change this patch, for example, by replacing the Oscillator with the *ADInput* object shown. The new Sound will function the same as the original, except that it will process the output of Capybara's analog-to-digital converters instead of the Oscillator.

Kyma uses this tree-like structure throughout. At the top level is the final Sound, which probably has SubSounds that in turn may have their own SubSounds. The only limit to this nesting process is available memory (and your sanity).

This structure makes Kyma's unified "look and feel" possible. Everything in Kyma is a Sound. There are Sounds that synthesize signals, Sounds that combine other Sounds, and so on. Even a "score" is a Sound, one that controls its SubSounds.

Once you've finished editing a Sound, you can drop it into a *Collection* or save it to a *Sound File*. You can create your own Prototype Strips containing Sound Files that you develop. You can also use a Sound as the basis for defining a new class of Sounds.

HARDWARE AND PERFORMANCE

Capybara is a black box about the size of a Mac II. Its rear panel contains ¼-inch stereo audio outputs and inputs, MIDI In/Out/Thru jacks, and a connector to the Mac. This communicates, via a parallel cable, with a NuBus interface card.

Capybara uses 16-bit stereo A/D and D/A converters with oversampling and supports five sample rates, from 11.025 kHz to 44.1 kHz. The cutoff frequency of the anti-aliasing filter is controlled by the user. Capybara generates very clean audio: A simple pass-through test at 44.1 kHz sounded completely transparent.

A minimal system contains two 56001s (one on the motherboard and one on an expansion card) with slots for seven more cards. Expansion cards also provide RAM for wavetables, delay lines, and synthesis parameter data. Programs downloaded from Kyma run in parallel, distributed across the available 56001s.

As you install more cards, processing power increases by 10.25 MIPs per card. This is a lot of horsepower (the Mac II averages 1.5 MIPs), but there are limits. The more cards installed, the more Capybara can do in real time. The unit has a short time to generate each sample (22.68 microseconds for a mono sample at a sample rate of 44.1 kHz). Symbolic Sound indicates that Capybara can run from eleven to 64 oscillators, in real time, at a sample rate of 22.05 kHz, depending on the number of cards installed.

If you devise a Sound that requires more computation than Capybara can handle in real time, the system tries to deal with the situation gracefully. After a warning, it will play in a degraded mode with clicks, pops, and general

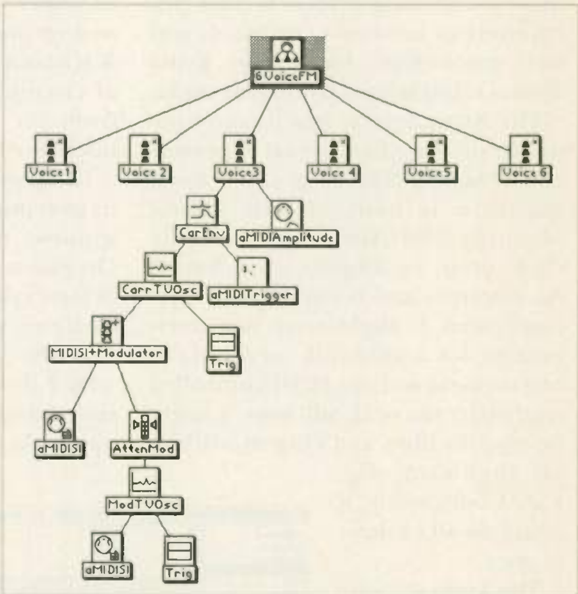


FIG. 4: Complex Sounds, such as this 6-voice, 2-operator FM patch, test the limits of Capybara's DSP processors.

Product Summary

PRODUCT:
Kyma System version 1.05

SYSTEM REQUIREMENTS:
NuBus-compatible Macintosh with 8 MB RAM, min. 40 MB hard disk w/ average access time 19 ms. or less. One NuBus slot required for interface card

PRICE:
Starter System I (one expansion card) $4,445.17
Professional System VIII (eight expansion cards) $8,550.19
Expansion cards $695

MANUFACTURER:
Symbolic Sound Corp.
P.O. Box 2530 / Station A
Champaign, IL
61825-2530
tel. (217) 328-6445

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ● ● ◖
EASE OF USE	● ● ● ●
PERFORMANCE	● ● ● ● ◖
VALUE	● ● ● ●

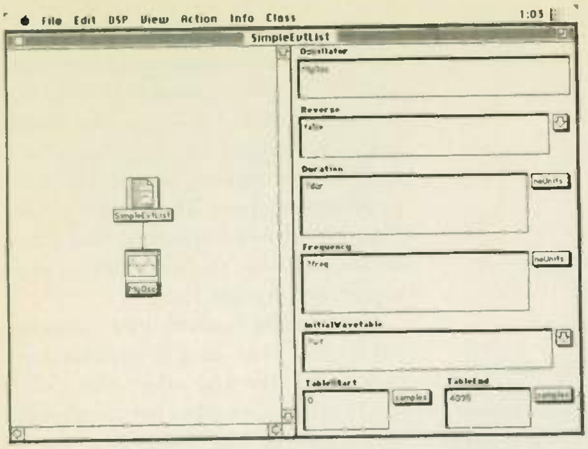


FIG. 5: A Score object controls playback of the connected SubSound.

"breaking up" of the sound. To work around this, you must either buy more cards, lower your sampling rate, or simplify your Sound. You also can save samples to disk, outside of real time. This produces a glitch-free sample file, but it won't work for live input.

Most music uses more than simple oscillators, so I devised a simple real-time performance test. Fig. 4 shows a "2-operator" MIDI-controlled FM Sound. It requires two oscillators, one envelope, one adder, one attenuator, one multiplier, two constants, and four MIDI processors. The results:

# Cards	Sample Rate	# Voices
2	22.05 kHz	2
4	22.05 kHz	4
8	22.05 kHz	6
8	44.10 kHz	1

The "# Voices" column shows the number of FM voice objects I could use without overloading Capybara. This chart gives a rough indication of the relationship between sample rate, hardware configuration, and processing power. Kyma can do many things, but it can't necessarily do them all at once.

SOFTWARE

Kyma's main features include:

Synthesis. The available synthesis techniques include additive, subtractive, FM, AM, granular, nonlinear waveshaping, FOF (formant-based vocal synthesis), and synthesis via resonators with feedback (good for plucked-string effects).

Signal generators include several oscillator types, with variable waveshapes and initial phase. Usually, the system preloads wavetables from AIFF or text files. (There are a variety of waveform files provided.) You can define a custom wavetable in terms of its partials or as a series of line segments.

Kyma offers several envelope-generation methods, including an envelope with an arbitrary number of linear segments. You also can use a one-shot table lookup as an envelope generator, with the wavetable of your choice. You can build even more complex envelopes by concatenation. It's even possible to do this algorithmically. (An upcoming release will feature graphic wavetable and envelope editing.)

Signal Combination. The *Sum* object adds together the outputs of its SubSounds. You can use *Attenuators* with Sum to make an [illegible]-input mixer. The *Difference* object subtracts one SubSound from another.

There are time-based combiners as well. The *Concatenation* object schedules its SubSounds one after the other. Unique time-based combiners delay the start of each SubSound so that all SubSounds end simultaneously.

Signal Modification. Kyma provides a panner, noise gate, gain booster, harmonizer, 7-tap delay line, and allpass and comb filters with feedback. There are time-variant versions of many of these, with control inputs that modulate their parameters. Other objects provide more unusual processing, such as repetition.

Surprisingly, there is no reverberator Sound, although a tutorial file gives an example of a reverb constructed from allpass and comb filters. Otherwise, Kyma's selection of processing features is appropriate and useful.

External processing. Audio input can be processed like any other source, and external signals also can control other Sounds. For example, the AmplitudeFollower can extract envelopes from an incoming signal. Other useful devices include a threshold detector and a sample/hold. There is also a zero-crossing detector that provides rather unstable pitch-following.

You don't see many devices that combine synthesis and processing of

(a)
"LISTING 1."

```
"Score supplies oscillator start time & duration (in beats), frequency (in pitch units), and waveform"
MM := 120. "This sets the tempo"

MyOsc start: 0.0 beats      dur: 5.0 beats    freq:4 d sharp pitch  wf:'sine'.
MyOsc start: 1.5 beats      dur: 4.5 beats    freq:4 e pitch        wf:'ramp'.
MyOsc start: 2.0 beats      dur: 3.0 beats    freq:2 c pitch        wf:'expon'.
```

(b)
"LISTING 2."

```
| r |                                              "declare a variable to hold random values"
r <- Random new.                                   "initialize random gen.
MyOsc   wf:'square'.                               "set the waveform"
1 to: 100 do: [:i |                                " MAIN LOOP - generate 100 events
    MyOsc
    start: (i-1) * 0.05 seconds                    "start event at time 0,0.05,0.10...4.95 seconds"
    dur: ((i-1) * 0.04 * r next ) seconds          "duration is .04,.08,.... 4.0 sec, times random value"
    freq: ((r next * ( i-1) * 20) +100) hertz].    "freq. is 100,120, ...1980 hz transposed randomly"
```

FIG. 6: In Kyma, a Score can determine every parameter of the SubSounds it controls. This score generates a series of notes with random pitches and durations.

external signals. This is one of Kyma's more exciting dimensions.

Score Language. The Smalltalk-80 programming language is an integral part of Kyma. The parameter fields of any Sound—though they may appear to contain simple values—in fact contain Smalltalk expressions that are evaluated at play time. If a parameter field contains variable names rather than constants, the parameter can be controlled algorithmically. A Kyma score is actually a Smalltalk program that supplies values for such variables.

A Kyma score is embodied in a *ScoreLanguage* Sound, which has one or more SubSounds. It schedules these SubSounds to run at event times specified in its *Score* field. The Score field also supplies parameter values for the SubSounds.

Fig. 5 shows a single SubSound, *MyOsc*, controlled by a ScoreLanguage object, *SimpleEvtList*. The variable names *?dur*, *?freq*, and *?wf* appear in the duration, frequency, and waveform fields for MyOsc. **Fig. 6a** shows the Score of SimpleEvtList.

This simple score specifies three events for the SubSound MyOsc. The score supplies values for the oscillator variables and event start times by listing parameter/value pairs after the SubSound name. In Music-N parlance, the SubSounds are equivalent to the "orchestra," with the score providing data to the "instruments."

The Music-N approach is perfectly viable in Kyma. The *FileInterpreter* object can be programmed to parse scores in older Music-N languages, but that's only the beginning. Kyma scores are themselves Sounds and can interact in ways that Music-N scores cannot.

For example, several scores could be SubSounds of a controlling "master" score. The output of several scores could be modified by signal processors and in turn controlled by MIDI or live input. A score could control real-time processing of live input. The possibilities are mind-boggling.

Smalltalk code can be embedded into the score itself, as in **Fig. 6b**. If you've done a little programming, the code shouldn't be too hard to understand.

This score generates 100 events at a rate of 20 events per second. Each event's frequency and duration is proportional to the loop variable i, so the total effect is one of rising pitch and increasing duration; but the frequencies and durations are somewhat random.

Kyma's score language lets you define your score with any degree of specificity. You can decide every detail, as in Fig. 6a, or compute every parameter, as in Fig. 6b. You can mix these extremes in the same score, or take an approach somewhere between.

MIDI. Kyma sends MIDI note and controller events, as well as arbitrary streams of Sysex or other messages. On the input side, Kyma performs channel-routing, extracts information of interest, and translates it into a usable format. The *MidiSI* Sound, for example, receives note events on a range of channels and translates note numbers to oscillator frequency. Kyma also reads Standard MIDI Files, handling file events such as direct MIDI input.

Kyma Sounds can be triggered from MIDI Time Code by setting up a "playlist" in the form of a score that specifies event times in HH:MM:SS.FF format. Kyma currently does not handle MIDI clocks, nor is it compatible with *MIDI Manager*.

Sample recording, playback, and editing. The primary purpose of recording in Kyma is to provide digital output, and to handle processor overload situations. Kyma's Record command writes samples from Capybara to an AIFF file, providing compatibility with products such as Digidesign's *Sound Designer*. The recording format can be 8-, 16-, or 24-bit, stereo or mono, at any supported sample rate.

The *SamplesFromDisk* object plays back AIFF files. This sample stream can be processed like any other source and the result recorded to yet another file. This is more like "sound-on-sound" than multitracking, as you can't separate out the original streams later. Only one input and one output file can be open at a time, so you can't mix several files at once.

Kyma supports limited sample file editing. You can cut, copy, paste, or clear file segments. To save your work, you must transfer the edited samples to a *SamplesFileSplicer* object and record its output.

PRICE, DOCUMENTATION, AND SUPPORT

Kyma runs on any NuBus-compatible Mac II with 8 MB RAM and a 40 MB hard disk. A larger drive is needed for recording. The program takes up about 3 MB of disk space. It runs under *MultiFinder* but leaves little room for other applications.

System prices range from about $4,500 (one expander card) to about $8,550 (eight cards). Individual cards cost $695 each. A 3-card system ($5,600.01) is the minimum recommended "for serious composers."

The Kyma System is easy to set up: Just plug the expander cards into Capybara and a NuBus card into the Mac, then auto-install the software. I was up and running within my first hour on the system. Symbolic Sound has worked hard to create a friendly environment, and the graphical interface is quite intuitive, especially if you've worked with modular systems. Online help is available for all Sound classes and their parameters.

The Kyma System comes with an excellent manual that includes many examples. The reference section is skimpy; much of it duplicates information you can get from Kyma's online help, and some of it presupposes a higher level of engineering knowledge than is appropriate. Even the most technical user would probably rather not enter complex numbers into the parameter fields of a filter. I would have liked to see this section supplemented by detailed applications notes for each Sound class.

Kyma is well-supported. Symbolic Sound publishes a user newsletter, has a bulletin board, and is promoting seminars and user groups. The company was responsive to my questions and problems and open to suggestions for improvement. In fact, they implemented several of my suggestions during the review period. They also quickly turned around almost all the bugs I reported.

CONCLUSION

I mentioned that Kyma's designers are ambitious. Have they realized their dreams? On the whole, I think they have. As a synthesizer, Kyma offers huge possibilities, without the arcane syntax and long compile waits of other computer-music systems. It's an impressive live-input processor as well. The integration of Smalltalk-80, the score language, and the Sounds they control is beautifully thought out, and it sounds good, too.

Kyma does have certain shortcomings. Based on my polyphonic-FM test, I'd say that Capybara, considered as a MIDI-controlled instrument, can't match the real-time performance of dedicated synthesizers. Most synths are hardware-optimized for one specific technique; a more powerful generation of general-purpose DSP chips will have to arrive before systems like Kyma can keep up.

Kyma is a little weak in the hard disk recording/editing area. Its editing features are rudimentary, and the sample-file windows are slow. It can play digital audio and MIDI simultaneously, but with little integration between the two, as in Opcode's *Studio Vision*. Despite Capybara's integrated A/D and D/A hardware, the Kyma owner must still plan on buying Sound Tools or similar products to do professional-level editing or multitracking.

Kyma will have great appeal to those challenged by and interested in modular synthesis and not put off by programming. This doesn't mean that Kyma is only for highly technical musicians, or that it excludes beginners. Kyma would make a fine teaching instrument in computer music, digital signal processing, acoustics, or even computer-programming courses.

The bottom line: Kyma is a *composer's* system, designed by a composer, for other composers. Symbolic Sound describes the potential user accurately: "Kyma assumes that you are a creator, not a consumer...that you know something about sound and structure...that you enjoy playing with sound, abstract structure, and logic." If you identify with this description and you have the budget, you would do well to take a hard look at the Kyma System.

John Duesenberry *is a Boston-based composer and software engineer. You can reach him as JOHNDU on the PAN network's* **EM** *Forum. He reminds you that Kyma means "wave" in Greek and urges you to look up the word "Capybara."*

Midisoft Studio 2.02 (IBM)

By Dennis Miller

If getting started is the hardest part of sequencing, Midisoft's program could be just the ticket.

Look around these days, and you'll see sequencers of all shapes and sizes. Some try to be all things to all people, while others aim for a specific niche. Midisoft Corp. has clearly chosen the entry-level user as the target audience for *Midisoft Studio 2.02*. This 64-track sequencer for IBM PC (also available in version 3.03 for Atari ST) doesn't go for feature-overload, but its clean look and ease-of-use suit the intended market well.

Midisoft Studio can handle lots of musical situations, and its features list includes the basic options that musicians need: quantize, transpose, real-time tempo and velocity adjustment, etc. The program also supports Standard MIDI Files. The user interface is straightforward, generally well thought-out, and should present no problem for the first-time user.

The manual is clear, well-organized, and certainly adequate for the beginner. Midisoft also provides a glossary of elementary terms. The short tutorial, included as an appendix, could be improved, however, especially in the

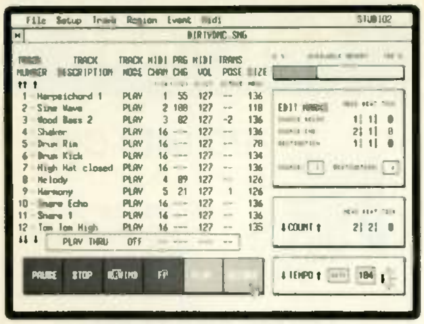


FIG. 1: Midisoft Studio's Main window features an editable track list. The Edit Marks box is used to set playback markers.

area of advanced features.

Though it doesn't run under *Windows*, *Midisoft Studio* is the first sequencer to use Dynamic Link Libraries (DLL) to integrate MIDI with presentation authoring tools such as *Asymetrix ToolBook*. That should serve it well with the multimedia crowd.

Midisoft Studio requires an IBM PC or compatible with 640K of RAM and a graphics adapter. A mouse is optional, but strongly recommended. The software sells for $99, or $149 when bundled with an MPU-401-compatible interface (a CMS401, or a Music Quest PC MIDI Card). Midisoft has chosen not to copy-protect the program.

Product Summary:

PRODUCT:
Midisoft Studio 2.02
sequencing software

REQUIREMENTS:
IBM PC or compatible with 640K of RAM (or Atari ST with 512 KB RAM); CGA, EGA, VGA, or Hercules graphics adapter; mouse (optional); MPU-401-compatible interface

PRICE:
$99

MANUFACTURER:
Midisoft Corp.
PO Box 1000
Bellevue, WA 98009
tel. (800) 776-6434
or (206) 881-7176

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ◖
EASE OF USE	● ● ● ●
DOCUMENTATION	● ● ●
VALUE	● ● ● ◖

Midisoft Studio uses the mouse-and-windows-oriented GEM interface, but unfortunately, it doesn't make use of GEM's graphic resources. The program only supports text input, with no graphic editing. You also have limited possibilities for viewing data, with two principal work areas (the Main and Event Edit windows). If you prefer a piano-roll display, or measure-by-measure views of multiple tracks, you should look elsewhere.

The program boots to the Main window (**Fig. 1**), where you'll find an editable track list, with track name, mode (solo, mute, or play), playback channel, program-change assignment, volume, and track size. You can view twelve tracks at once and scroll to other tracks using up- and down-arrows. A set of tape-deck controls appears at the bottom of the screen, two boxes show Tempo and Count (location), and one final box, Edit Marks, provides the facilities to set location markers during playback.

RECORD AND PLAYBACK

Recording and playback couldn't be easier. Click on the Record button and *Midisoft Studio* automatically enables the first empty track for recording. (You can select a different track by clicking on its status indicator.) You can only record one track at a time, without overdubbing, but you can bounce tracks as often as you want. The program records with 96 ppqn resolution.

You can apply auto-quantize while recording, but from the Main screen, you can't tell if the feature is on or off. Be sure to check the Quantize Region menu before recording. Quantization is to the nearest unit, from 64th-note triplets to whole notes.

Recording can start from any point,

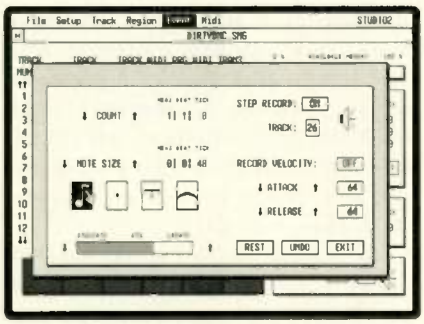


FIG. 2: In the Step Record window, as in most of the program, you can set parameters with the arrow keys, the mouse, or by typing in values.

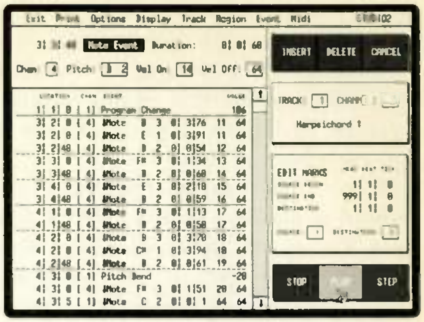


FIG. 3: To edit in the Event Edit window, click on the event, and a window appears that displays the parameters.

but there's no punch-in or punch-out in the traditional sense. I like the adjustable lead-in feature, but I'd prefer to use the lead-in on record only. As it stands, you get the lead-in on playback as well.

STEPPING IT IN

You can enter data by step recording, though you need a MIDI controller to specify pitch (**Fig. 2**). You select note duration, from dotted-whole-note down to a single tick, by using the arrow keys, clicking on the note icon, or typing in the value. (You can change most values in the program using these same options.)

You can record attack and release velocities from your performance, or enter values for each note before recording. I like the ability to step in program changes and the undo-most-recent-event feature, but I wish there were some way to enter pitches directly from the screen.

To play back, just rewind to the measure you want and hit Play. You can change MIDI channel, program change, and tempo while playing. To mute a track, just click on its Status button. You also can "solo" one or more tracks. You can step one event at a time by clicking on the Play control, but I found that the program responded slowly to my clicks. There is no real looping, but you can insert the same region several times for a similar effect. Finally, for the mouseless, there are function-key substitutes for all of the tape-deck commands.

EDIT ON

One of the nicest touches is *Midisoft Studio's* complete consistency of operation. Once you learn how to use one edit function, you'll have no trouble with the rest. This is especially true with large-scale edits performed on a selected area of a single track. Select "Region" from the menu bar, and you'll see a window listing various operations: Insert, Delete, Paste, Erase, Transpose, Quantize, Scale Velocity, and Change Tempo.

Selecting any of these brings up an appropriate dialog box. For example, the Insert Region dialog prompts you for source and destination tracks, start and end times, and number of repetitions. After making your selections, click "OK," and it happens, faster than with any program I've seen. You then hit "Exit" to return to the track list. This is handy if you are editing more than one track, but I'd prefer to have the program switch back to the work area. Unfortunately, there are no multitrack editing capabilities.

To edit individual events, move to *Midisoft Studio's* second main work area, the Event Edit window (**Fig. 3**). You'll see a standard event list displaying location, channel, event type, pitch, duration and velocity information about every event. To edit, click on an event, and a window appears at the top of the screen displaying all that event's parameters. Highlight a parameter, then change the value using the mouse or keyboard. The process is simple, but it would be even easier if you could edit right from the event list. To delete an event, set Auto Delete (in the Options menu) and click directly on the event.

One of the program's nicest touches is its consistency of operation.

TAKE A LOOK

Midisoft Studio lacks piano roll and multiple-track views, making it hard to get a different perspective on your music. It also could stand better integration across work areas. For example, if you mute a track in the Main window, you can't unmute it from the Event Edit screen, though you can jump into Solo Mode for that track.

Limitations aside, this program is completely acceptable for basic composition and song-writing. And at $99, you certainly get your money's worth. If you're looking for a tool to create presentations or uncomplicated music, you ought to take a look.

Dennis Miller *is on the music faculty of Northeastern University in Boston, Massachusetts, where he teaches music theory and technology.*

InVision Protologic

By Charles R. Fischer

E-mu's Proteus module takes on new life with fresh, ROM-based samples and patches.

The E-mu Proteus/1 is legendary for delivering excellent sound quality, programmability, and extensive multitimbral power at an unprecedented price. However, a major drawback is its inability to upload new samples from memory cards or MIDI Sample Dump, since many users wish the Proteus/1 had a good electric piano, a strong fretless bass, and a few analog synth pads.

To address those limitations, InVision Interactive, in cooperation with E-mu Systems, has produced the Protologic upgrade board for the Proteus/1 and /1XR. The Protologic includes four megabytes of ROM samples and 128 new patches that add many "missing" sounds to the Proteus.

HARDWARE

The Protologic package includes the necessary circuitry, a special mounting bracket, and a brief user's guide. Unfortunately, the upgrade kit is not intended to be installed by the user and must be added by an authorized E-mu service center. According to an InVision representative, installation typically takes 30 minutes.

The unit is shipped with a bracket that should be used when the upgraded Proteus is rack-mounted. This bracket provides for heat dissipation, or as the user documentation says, "will prevent the Proteus from overheating and causing damage to itself and the board." This means the upgraded unit may run warmer (although mine didn't) and will take up $1^1/_2$ inches more space.

Documentation is sparse; a number of subjects are covered in a mere eleven pages. This included warranty terms, lists of new instruments and patches, diagrams of available drum maps, and programming hints. There's not a lot of detailed information, but the text is written in a casual style that delivers the essentials.

THE SAMPLES

One of the best things about the Proteus is it offers the programming power for tweaking and combining factory samples to yield a huge variety of new sounds. The new samples added by the Protologic upgrade offer programmers an even larger palette.

Most of the new samples were prepared with the same high quality as the original sounds. Unfortunately, I found a few losers. For example, noise was audible on the higher notes of the Rock Organ sample (instrument 126). There also are several samples with less-than-optimum loops; the Sax Section (instrument 140) is especially noticeable.

Percussive samples are especially powerful when combined with other instruments, notably tablas and kalimba. Finally, samples of a tambourine, shaker, and triangle are looped to create repetitive patterns and polyrhythms (try playing octaves and fifths to start).

THE PATCHES

I found the Protologic patches somewhat uneven when compared to E-mu factory sounds. There are some real

Product Summary

PRODUCT:
Protologic Proteus/1
Expander Board
PRICE:
$495
MANUFACTURER:
InVision Interactive, Inc.
269 Mt. Hermon Rd.
Suite #105
Scotts Valley, CA 95066
tel. (800) 468-5530
or (408) 438-5530

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AUDIO QUALITY	● ● ● ●
SAMPLE SELECTION	● ● ● ● ◖
DOCUMENTATION	● ● ●
VALUE	● ● ● ●

FIG. 3: Kawai's MM-16 MIDI Mixer offers seventeen assignable MIDI faders for real-time MIDI mixing.

envelopes into your sequencer or by using mute automation.

You also can use the speed and precision of automation to send bursts of audio to effects for brittle reverb hits and the like. If a particular mixing event—a series of fast fader moves, for instance—occurs more than once in a mix, MIDI mixing gives you even more. Once you perfect all of the nuances for the first instance, you can copy them to all the other places—a big time-saver.

PERFORMING THE MIX

When I first heard of automated mixing, I assumed that it used the mechanical "flying faders" idea, in which faders move like the keys of a player piano and show you the exact fader position at all times. This is handy, and it rates pretty high on the gee-whiz scale, but it's expensive. To date, Yamaha's digital mixers are the only under-$10,000 systems to use mechanical faders. It's more common to use faders to enter the mix and then view a graphic display to see their positions on playback.

What if your mixer doesn't have faders? Then get your hands on a generic MIDI fader, either hardware or software. Many sequencers offer onscreen, assignable graphic faders; some MIDI mixers, such as Mark of the Unicorn's MIDI Mixer 7s, come with stand-alone fader software. A few master keyboards, such as the Yamaha KX88, offer hardware sliders that can be programmed to send MIDI Volume or other controllers.

The best of the breed are the specialized units that feature a bank of sliders, such as the J.L. Cooper Fadermaster, Lexicon's MRC, and the Kawai MM-16 MIDI Mixer (see **Fig. 3**). These devices combine the benefits of software reconfigurability with the immediacy of a hands-on interface. These boxes are inexpensive, and if you do a lot of MIDI mixing, they're well worth it. (We explain how to make a MIDI Fader in the Feb. 1991 issue.)

Mixing by MIDI takes time and effort. But if you want complete control over your finished musical work, there's nothing like it. Good luck and happy mixing.

Dan Phillips, *a Bay Area composer, singer, and keyboardist, works as a product specialist for Korg Research and Development. He's looking forward to the reaction to next month's long-awaited cover article, "MIDI-Controlled Journalism."*

● **PROTOLOGIC**

killers, such as Like A Movie (a wonderfully warm analog pad), amidst a selection of losers. Lead Guitar (267) is remarkably bad. Fortunately, the patches log far more winners than losers.

Some patches are poised to become this year's aural clichés. The new acoustic and synth strings are pristine and luscious, and I prefer them to the E-mu samples. The drum kits are remarkable for their sound quality and variety. There's something for nearly everyone: punchy TR-808 drums for rap and hip-hop, aggressive snares for rock, and ethnic percussion for new age.

To increase usefulness of the drum sounds, Protologic has five separate percussion maps. These are mapped to follow E-mu, Korg, Roland, Alesis, and Yamaha standards, which is a great aid in using Protologic drums with other instruments.

In my opinion, two of the biggest omissions on the original Proteus were an adequate electric piano and fretless bass. Protologic has solved this with a gorgeous electric piano. While the sound is more digital FM than authentic Rhodes, its crispness and clarity makes it a standout. The crystalline quality is helped by the Proteus' excellent high-frequency response.

I've also spent hours trying unsuccessfully to create a fretless bass that would compare to the standout on the Korg M1. Thanks to an authentic sample, the Protologic fretless is a close second.

Some Protologic patches are poised to become this year's aural clichés.

While the Protologic's synth patches didn't convince me to sell my Oberheim Xpander, they are usable (especially if you don't have access to the real thing).

Probably the weakest set was the guitars. I was surprised to discover the electric guitar patches were sadly lifeless. Admittedly, the guitar is not the easiest instrument to reproduce convincingly, but it *can* be done (as demonstrated by the Roland U-220).

CONCLUSION

While the original Proteus/1 is a landmark instrument, it's even better with the Protologic installed.The Protologic provides electric piano, a variety of drum sets, and many other essential sounds left out of the original instrument.

The Protologic patches and samples were chosen for the average working musician, and for the most part, they deliver what they promise. I would prefer fewer emulative sounds and more atmospheric panoramas. However, InVision promises to issue new sets of patches on a quarterly basis, so a wider variety of sounds is forthcoming.

Despite a few lame sounds, most people are going to like what they hear. If you're performing contemporary music, you'll find plenty of uses for the Protologic. If the Proteus/1 is your primary (or only) tone module, you'll find it invaluable. ⊛

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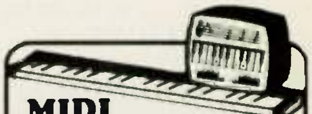

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THE BACK PAGE

I Can't Work This @%$#& Thing!

In the quest for more features, we designers sometimes forget the user's needs.

By Roger Linn

The other day a friend gave me a copy of a magazine with a cover story entitled "I Can't Work This @%$#& Thing!" that discussed the difficulty of operating consumer electronics products. The piece stated that in the 1980s, customers constantly demanded more features. But now, in the 1990s, customers ask for simpler products and often request that unnecessary features be removed. I liked this idea so much I decided to steal the title for this article.

Let me give you an example. I have a CD player that has a button on the remote to open or close the CD drawer. Now, if I have to walk over to the player to remove the CD anyway, what's the point of opening it from across the room? To make it worse, these unnecessary buttons are made very small so they can fit on the remote, making it awkward to press the correct button.

Not surprisingly, electronic music products share this problem. To reduce costs, products often have a small LCD display and a few hundred features packed into six or eight tiny multi-purpose buttons. For example, if you hold Shift and Parameter while simultaneously pressing Data+, then press the cursor button five times and do this on alternate Tuesdays after 3:00 p.m., you will access the sound select screen or some other commonly needed function. In a typical MIDI studio containing many products from different manufacturers, the problem is compounded because each product uses a different set of operating rules. This sort of studio is usually the domain of the technically minded or those who can afford to hire them.

What, then, is *user friendliness*? I think a product is user-friendly if the average customer can walk up to it for the first time and figure out how to perform most of its commonly used functions without reading the manual.

Why are some products more user-friendly than others? One reason, as stated earlier, is that in order to give customers features *and* low cost, manufacturers must use smaller displays and fewer panel controls. Another reason is that customers request new features far more frequently than user-friendliness, and manufacturers are merely giving them what they ask for. (Even customers who ask for easier operation often buy products with more features over those with easy operation.)

Also, companies based in non-English speaking countries have more difficulty because the products must communicate in English. But sometimes the reason is simply that design engineers forget musicians are busy making music and don't have time to study every product in their studios.

How do you design a user-friendly product? Here are a few of Roger's guidelines for user-friendly product design:

• Talk to customers. Find out which features they want and, just as importantly, how they expect them to operate. Then try to design it that way.
• Assume the customer will never read the manual, and design the product so it doesn't need one. (But write a great manual anyway.)
• Make all commonly used functions easily accessible, by a single button-press if possible.
• Develop a set of user interface rules, and stick to them in implementing all features.
• Get customer feedback throughout the design process: revise, get feedback, revise, get feedback, etc.
• Include a context sensitive "help" function.

I'm the first to admit I rarely achieve all of these goals, but if every design engineer had these "rules" framed above his or her workbench (and the manager's desk), I believe their products would be easier to use.

What can the consumer do to make products easier to use? Suggest to the manufacturers you want it. They need and (hopefully) appreciate your feedback. If enough people say user-friendliness is important, the manufacturers will respond.

Is this the decade of truly user-friendly products? With increased customer demand for simplicity and decreased cost of components such as larger LCD displays, we finally may be able to work these @%$#& things. That is, if we all think that it's sufficiently important. I do.

Among other products, **Roger Linn** *created and produced the first programmable rhythm machine with sampled sounds. One of his recent designs is the Akai MPC60 MIDI Production Center.*

The opinions expressed in "The Back Page" are those of the author and do not necessarily represent the opinions of ACT III Publishing, *Electronic Musician* magazine, or its staff. Opposing viewpoints are welcome and may be published, at the editor's discretion, in "Letters."

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