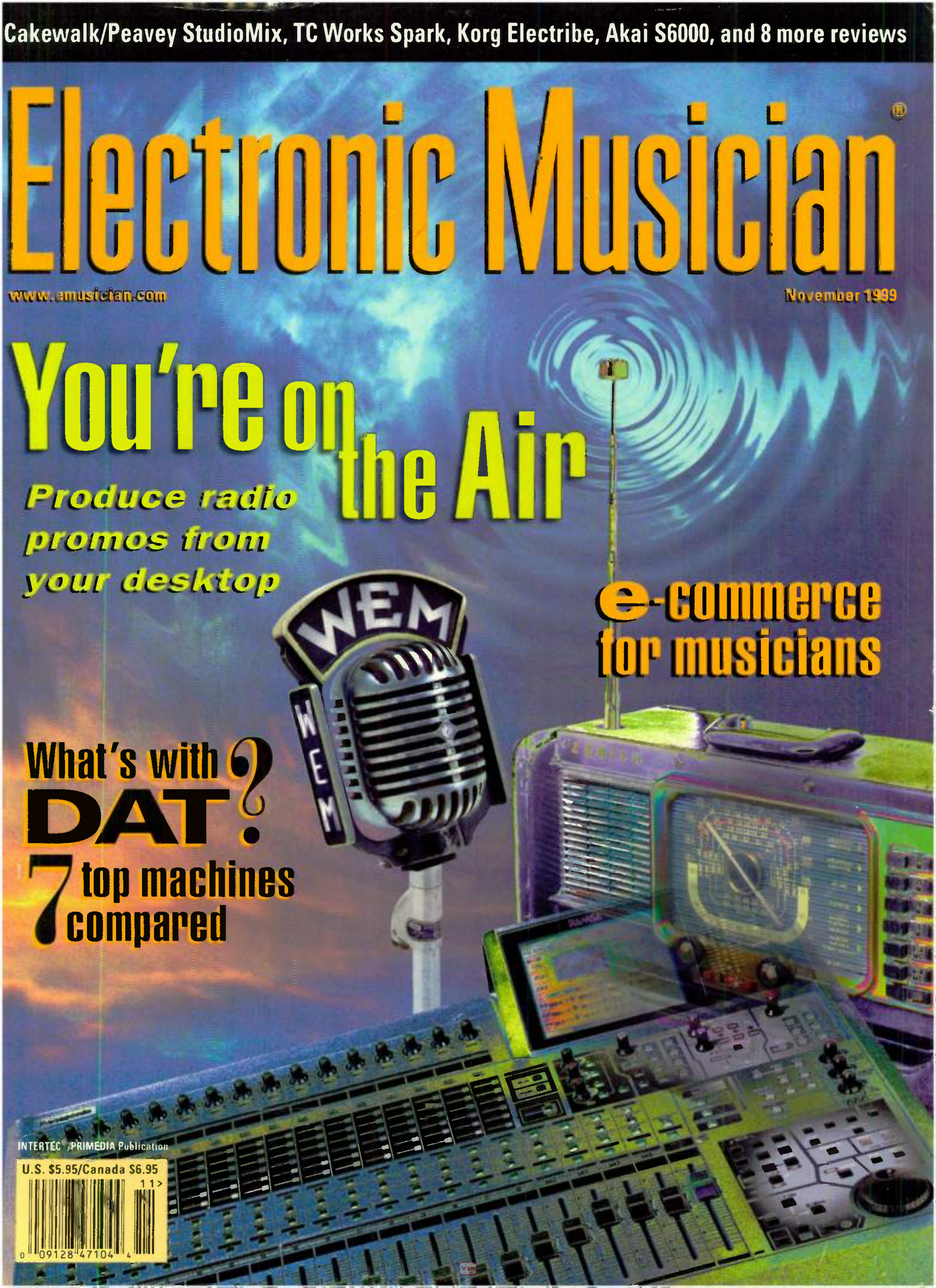
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Electronic Musician
NOVEMBER 1999 VOL.15, NO. 11

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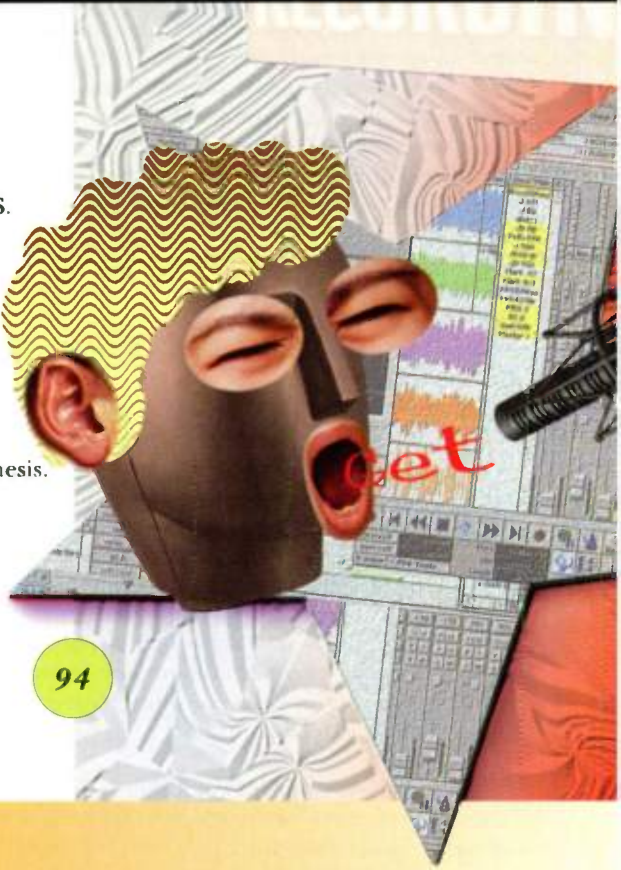

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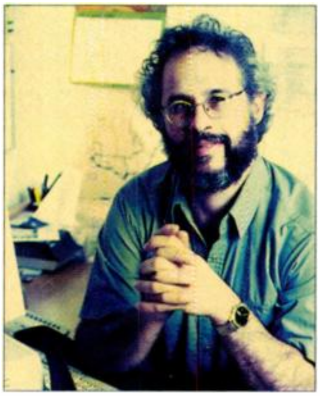
REVIEWS

Cover: Illustration by Ron Brown

The Price We Pay



ANTHONY PIDGEON

As you have probably noticed, all prices published in our articles are manufacturers' suggested retail prices (the MSRP, or "list" price) for the U.S. market. A manufacturer's representative recently contacted us to complain about our policy. He pointed out that his company discounts off the list price, so that the street price of his product is well below the list price. In contrast, his competition's list price is lower, but they do not discount off their list price, so the street price and list price are the same. As a result, the street prices of the two products are nearly the same, but their MSRPs differ significantly. The rep felt that our policy of quoting only list prices was unfair to his product. He was concerned that our readers would make purchasing decisions based on illusory list prices.

The rep had a good point, as far as it went. If you want to find out the best price for a particular product, you won't get a completely clear picture from the MSRP. So why doesn't **EM** quote street prices? Because there is no such thing as a single "street" price. The same item could sell for one price at a music store in Los Angeles, cost a bit more direct from the manufacturer, sell for a third price via mail order, and go for yet another price at a music store in Cleveland.

Street prices vary for several reasons. Some companies sell direct to the public at list price, others sell through dealers at discount prices, and a number of manufacturers do both simultaneously. Small dealers don't always get the same wholesale discounts offered to huge megastore chains. Furthermore, even if two dealers buy a given product at the same price, that doesn't mean they will sell it at the same price.

Given all these variables, publishing street prices as the bottom line in a national (and increasingly international) magazine is meaningless. The only way we can give **EM** readers something that even resembles a common frame of reference is to rely on the MSRP. That's why every price you see in the magazine—in features, columns, reviews, and "What's New"—is a list price.

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A few closing notes: This summer we published a pilot issue of a small but promising new live-performance magazine, which we tentatively titled *Performing Musician*. Now we are revving up this mini-magazine in earnest: in January we'll begin producing it quarterly.

We have renamed our new live magazine "*Onstage*," which pretty much tells you where we're taking this project. Much to my delight, **EM** associate art director Tami Needham has agreed to direct the new project. You've seen her designs monthly in **EM**, and she'll be ably assisted by her colleagues in our art department, so you know *Onstage* is going to look great.

When *Home Recording* and *Guitar* executive editor Mike Levine agreed to join our staff, we knew the stars were aligned in our favor. We have been trying to get him on our team for quite a while. So Mike will join **EM** as a full-time associate editor—and the editor of our *Onstage* supplement—in time for our January issue.

Electronic Musician®

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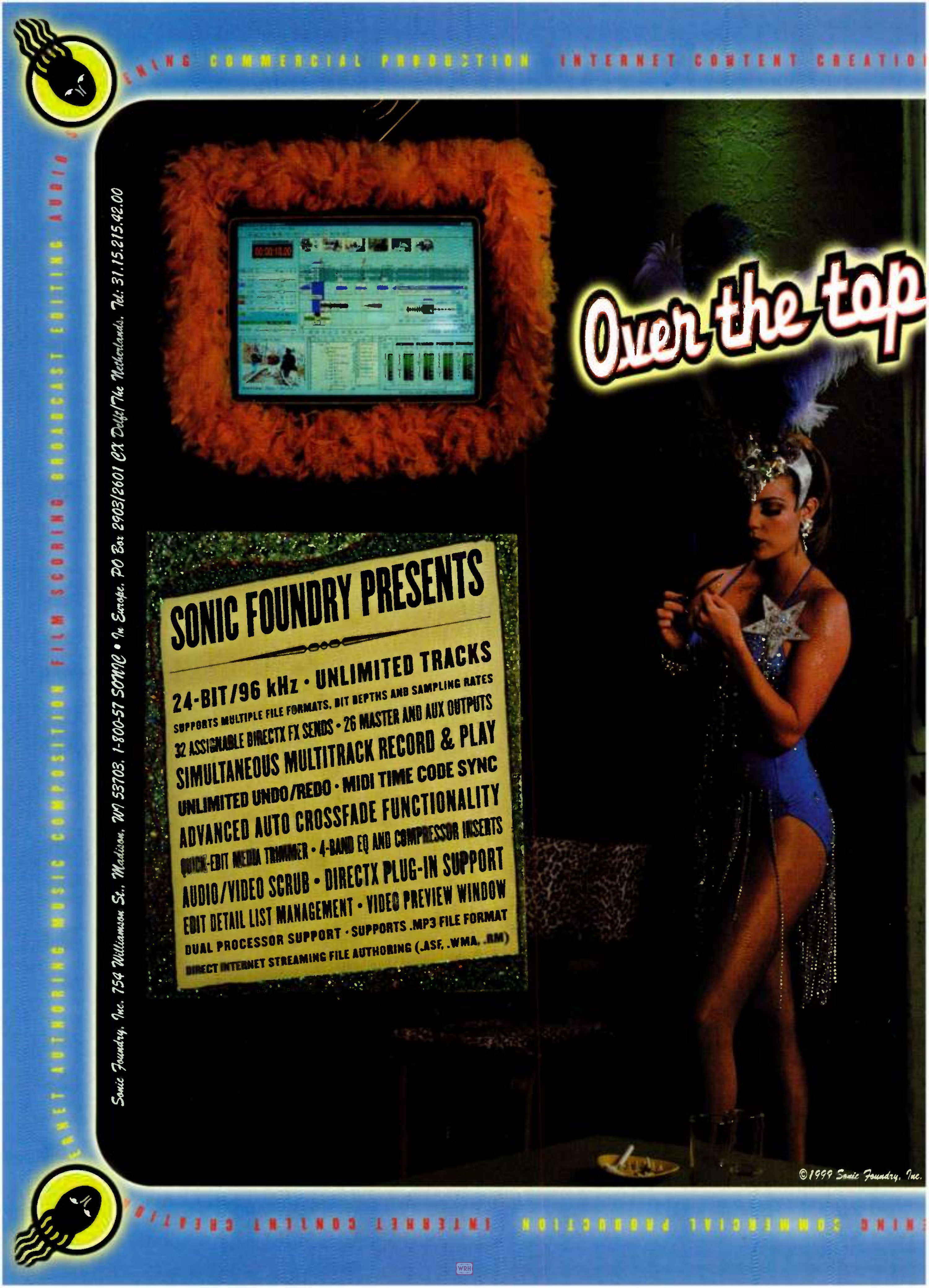

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guide. Another explained that his job is not too different from being a lead singer in a rock band. Not too much ego there! It is obvious that the DJ is the performer of the music, and the dance audience either identifies with him or her or goes to the next club down the street.

The DJs were also asked what they thought it was about electronic music that affects people so deeply. The answers seemed fairly sincere and had to do with soul, emotion, and energy: all the things anyone would want from any piece of music, whether it be classical, jazz, bluegrass, or Balinese barong. It reminded me of an interview of Janis Joplin (widely known for the soul, emotion, and energy she put into a song), when she was asked a similar question. She gave her interviewer a disbelieving sidelong glance, threw her head back, and laughed hysterically.

Fred Treece
via e-mail

MEET YOUR MARKER

Regarding "The Twice-Baked Mix" by Erik Hawkins ("Recording Musician," July 1999), I enjoyed the overall tone of the article but have one suggestion for creating the MIDI tempo map. I know that the technique described, which took almost an entire page of a three-page article, is a common one used by MIDI folks. But there is a feature within Pro Tools for creating tempo maps that is dead-on accurate to the audio, and probably takes a fraction of the time of the old-school method.

By using Bar & Beat markers, one simply puts a marker at the appropriate points in the song, by placing the cursor on the exact audio event and typing Command-I, or Identify Beat. Even for a song cut to a click, the audio rarely is totally in sync with the click. The Bar & Beat method works no matter how the track was cut. You need more markers if the tempo moves. After the map is created in Pro Tools, it can be exported to any sequencer via Export MIDI.

There are a few gotchas. Bar 1, beat 1, must be at the SMPTE start of the session, so the downbeat of the song usually ends up at bar 3 or 5, depending on how much preroll you want. Also, sometimes the sequence program won't recognize the PT generated tempo map unless there is *some* MIDI data in the export. Often, I'll import a bar of nonsense into PT, just to get the map out. When that map is imported into the sequencer program, the MIDI will be dead on with the audio, or as accurate as your map. The main point is that rather than hit-or-miss, this method puts tempo changes *exactly* where they belong, on the corresponding audio events.

I hope you can get some use out of this technique, because it's a lot faster and more accurate than any tap-tempo exercise.

Lou Giordano
via e-mail

SEARCHING FOR ISOLATION

I was wondering if you can help me locate a product. I saw this product in your magazine, and I thought it was a great idea. It was a small cube with a speaker built into it. It also had a door that allowed enough room for a microphone to fit inside the cube.

The idea was that if you needed an isolation area for an amplified instrument, such as a keyboard or guitar, you could plug into this, mic it if you needed to, and play away. If you guys can in any way help me find out what this product is called or who the manufacturer is or *anything*, it would be greatly appreciated.

Andrew Mitchell
via e-mail

Andrew—You're thinking of Folded Space Technologies' Micro Room, which we reviewed in the November 1996 issue. You can contact the company at (770) 427-8288; fax (404) 321-5094; e-mail fspace@mindspring.com; Web www.mindspring.com/~fspace.—Mary C.

REVIEWING REVIEWS

I'm a fan and regular reader of your magazine. I buy it each month here in Montreal, Canada.

In my opinion, what your average reader needs is accurate and timely information. Yesterday I bought your September 1999 issue. In that issue, there are reviews of at least two products that have been available since last spring! I already read reviews on these a few months ago in another magazine.

I think that people are relying quite a bit on your evaluation of a new product before they decide whether to buy it. Reviews should be done a lot faster than they are now.

Carl Pettigrew
via e-mail

Carl—Thanks for your comments. One reason we do not get reviews out as fast as you (and we) would like is that our authors take a lot of time to thoroughly test the products on real projects.

In addition, we run a large number of reviews of a broad range of products, and we try to serve up a "balanced diet" in each issue, combining coverage of transducers, signal processors, Mac and Windows software, sound cards, synths, sample CDs, and so on. Let's say a half dozen large Windows product reviews are ready to go at the same time; we won't run them all in one issue, because musicians who don't use Windows PCs for production will not be well served. In that type of situation, a few reviews will have to wait their turn. Of course, a given review can also be delayed because of the human factor: authors sometimes have to turn their attention to other commitments.

That said, we would like to get at least the key reviews out faster, and we are looking at ways of accomplishing that.—Steve O.

AWWW, GO ON

I would like to thank everyone at **EM** for their superb writing, reviews, and features. The research that was available to me in the May ("Seven Studios of Gold"), June ("The Complete Desktop Studio"), and July 1999 ("Sequencing Games") issues helped me construct my awesome studio with G3 400/9 GB, racks and synths, supporting peripherals, *Cubase* VST/24 4.0, and so on. I have been in production for 14 years at my appropriately renamed GEARlust Studios, and these articles have helped me make a great leap forward! Thanks again. Curious readers can e-mail me for a gear list (synthman@peakpeak.com).

Marcel
via e-mail

WE WELCOME YOUR FEEDBACK.
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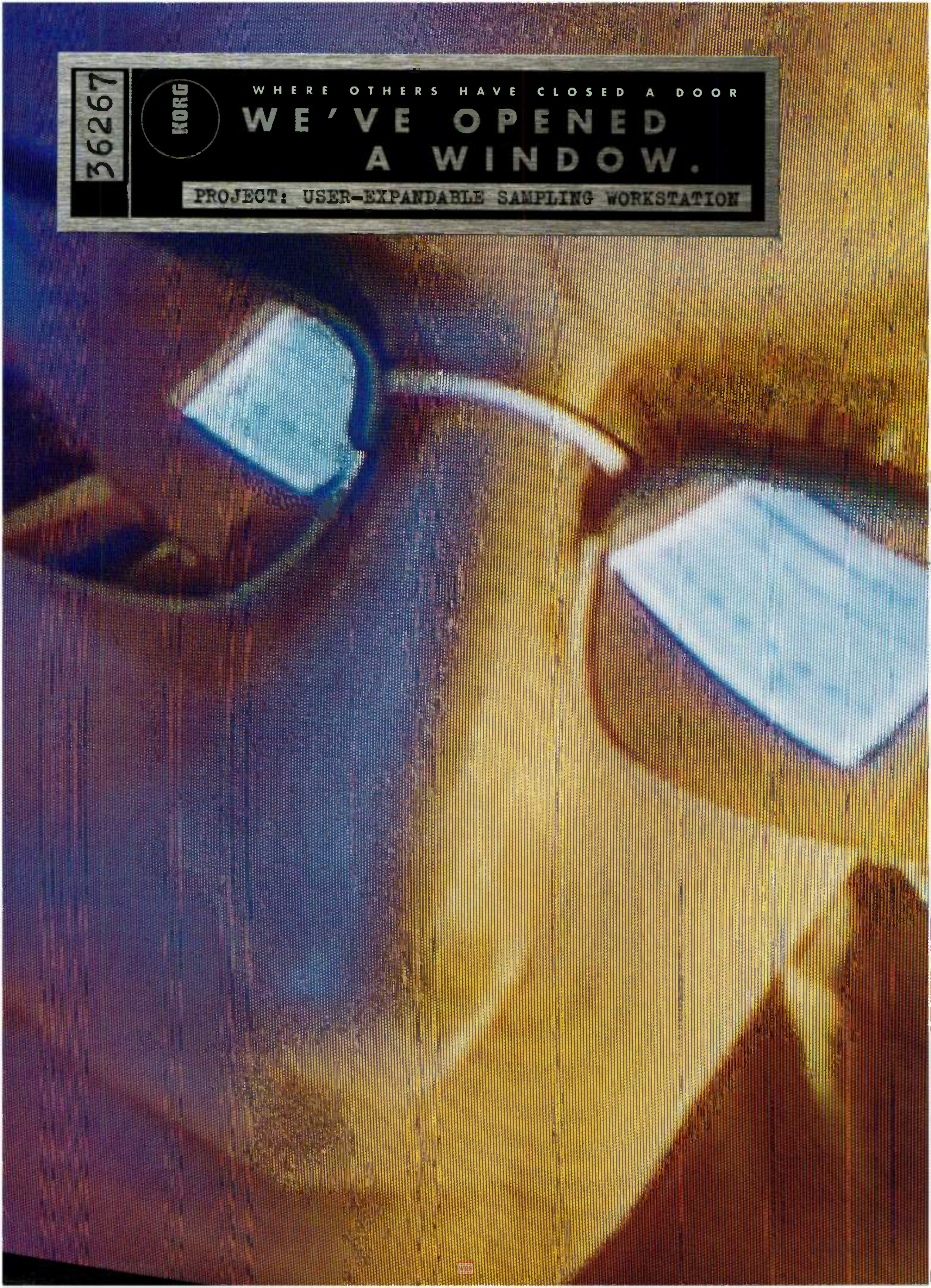
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WHAT'S NEW

By Rick Weldon

MCC NUMBER CHART PRO

In most Nashville studios and many other facilities around the United States, chord progressions are written in distinctive chord-number charts, providing a common language for session musicians. Music Computer Consulting's *Number Chart Pro* v. 2 (Mac/Win; $44) is a custom font package that enables you to create pro-quality number charts within any word processing application.

You can make the charts in any system font and at any point size. You get symbols that denote first, second, and third endings; 8th- and 16th-note triplets; and most common chord structures, including 9th; diminished 7th; 13th; 11th; and 9 (♭5). You also get symbols for split bars, "pushes," walk-ups, repeat bars, fermatas, and codas. You can insert symbols used in traditional notation above the numbers in order to show rhythm patterns. Plain text can also be used anywhere in the chart.

The font can be used with virtually any Mac or Windows word processor. Music Computer Consulting; tel. (615) 851-6633; e-mail numchart@aol.com; Web members.aol.com/numchart.

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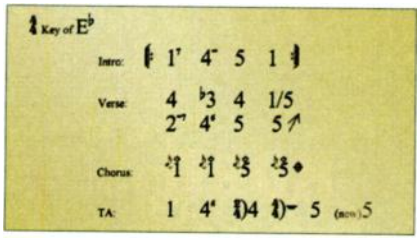

SUZUKI QCHORD

In a bid to make anyone a musician in minutes, Suzuki has released the QChord ($199), an update of the company's '70s-era Omnichord. The futuristic-looking MIDI instrument has three sets of 12 buttons that cover major triads, minor triads, and 7th chords in each key. You can play progressions in a four-octave range, using these chords as well as major and minor 7th, augmented, and diminished chords. By using the QChord's pressure-sensitive "strumplate" to activate the chords, you can give the progressions a guitarlike texture. You can also tap on the strumplate for hammer dulcimer-like performances.

You get 100 General MIDI sounds and ten preset, auto-accompaniment tracks—automatic bass lines, chord progressions, and rhythms—with control over tempo and relative volume. Intro/End and Fill buttons provide variety. You can disable the accompaniment tracks or play with any or all of them. A pitch-bend wheel is provided, and onboard effects include reverb, vibrato, and chorus.

The QChord also accepts optional Suzuki QCard Song Cartridges ($19.95 each). Ten cartridges are available; each features 8 to 12 fully orchestrated accompaniments. Songs contained on the QCard cartridges are assembled by theme; QCards include *Lennon and McCartney, Country Classics, Holiday Song Favorites, Religious Standards,* and five others. A rhythm-only QCard is also available.

The QChord has a 5-inch speaker and bass porting. The line output is on a ¼-inch unbalanced connector. The unit can be powered by an included AC adapter or by eight C batteries. Suzuki Corp.; tel. (800) 854-1594 or (858) 566-9710; fax (858) 560-9517; Web www.suzukimusic.com.

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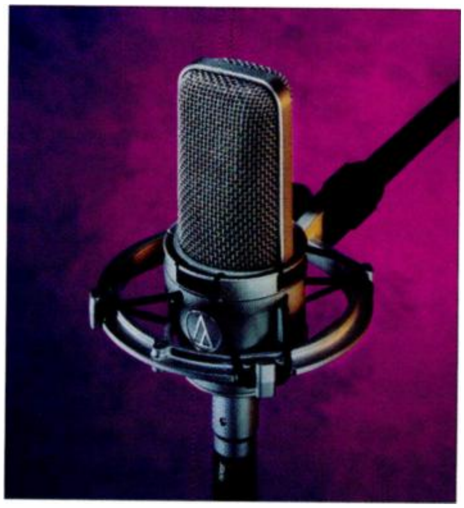
AUDIO-TECHNICA AT4047/SV

Breaking away from the transformerless design concept behind Audio-Technica's other 40 Series microphones, the 4047/SV ($695) cardioid condenser mic incorporates a capacitor and a transformer in its circuitry and is designed to replicate the response of vintage FET studio mics. The 4047/SV's two diaphragms are gold-plated and aged with a process designed to ensure that the mic's response characteristics remain constant over years of use. A floating-construction element is used to provide the best possible isolation from noise and vibration.



A 12 dB per octave low-cut filter rolls off the signal below 80 Hz. There is also a 10 dB pad. The mic ships with the AT8449/SV shock-mount, is finished in a vintage-style silver matte, and weighs 14.5 ounces. It requires a 48V DC phantom power supply.

According to Audio-Technica, the 4047/SV has a frequency response of 20 Hz to 18 kHz, a dynamic range of 140 dB (at 1 kHz), self-noise of 9 dB, and a signal-to-noise ratio of 85 dB (at 1 kHz). The mic boasts a maximum SPL of 149 dB (1 kHz at 1% THD) or 159 dB when using the 10 dB pad. Audio-Technica U.S., Inc.; tel. (330) 686-2600; fax (330) 686-0719; e-mail pro@atus.com; Web www.audio-technica.com.

Circle #403 on Reader Service Card

Logic Audio 4.0
Unrivaled Power.

Utilizing the most advanced software technology available, Logic Audio seamlessly integrates the musical media of digital audio, MIDI and notation into a leading edge composition and production system. With its speed, precision, reliability and unsurpassed flexibility, Logic Audio has gained the reputation as being the foremost audio production tool in the international studio world.

Welcome to version 4.0: you'll appreciate the power of up to 34 (thirty four!) high quality native plug-ins, 24 bit audio recording, an enhanced user interface, compatibility with all popular plug-in and driver formats, as well as a tremendous amount of other improvements.

The Logic Series has set itself apart by providing versatile, configurable, cross-platform solutions for beginners and professionals alike. More commercially available CDs are made with Logic Audio than with any other digital audio/MIDI sequencer.

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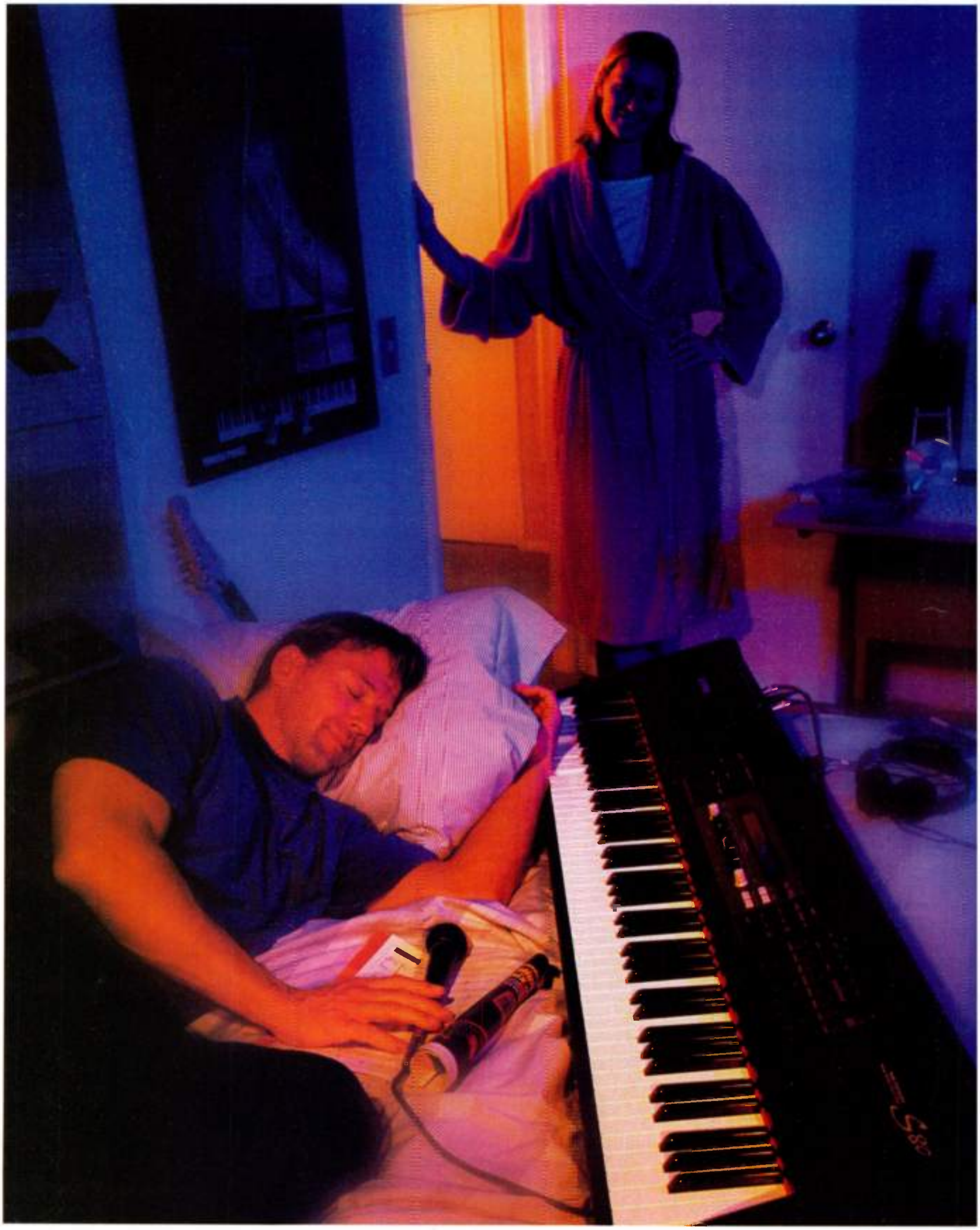
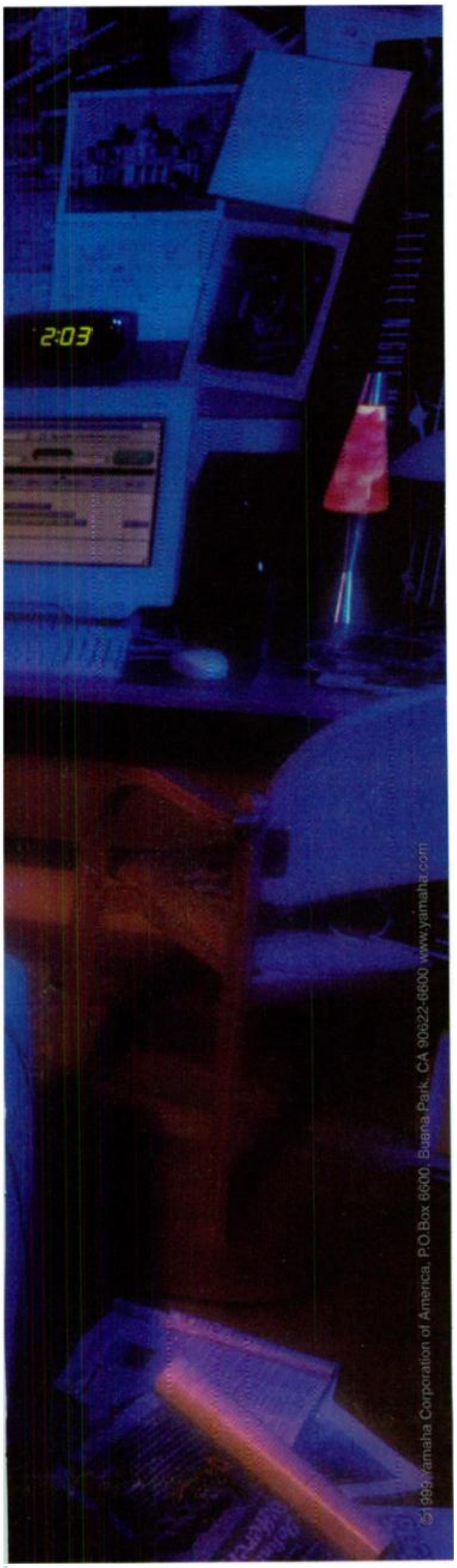
The Keys in a Successful Relationship.



• CD-ROM with XGWorks 3.0 Lite software • Pitch-bend, modulation wheels • Effects: 12 reverb, 23 chorus, 93 insertion • Sounds: 364 AWM2 voices, 128 performances •

• Modular Synthesis Plug-In System • PLG boards add complete synthesizers, polyphony, effects and multitimbral capacity

MIDI control • 4 Assignable data sliders, 5 assignable knobs • Quick Access key: easily, instantly select your favorite voices • 4 Programmable keyboard zones • Channel aftertouch • 128 memory locations • A/D input jack • Effects applicable to external audio • Direct computer interface •



Respond to each other's needs and touch.

88-Note weighted action MIDI master controller keyboard with aftertouch,
4 assignable sliders, 5 assignable knobs, 3 pedal and 1 breath controller inputs

Excite the passion inside each other.

Incredible, evocative sounds based on the award-winning EX-series including
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Grow together.

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Create a long-term relationship with your perfect partner

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not just sounds • Two PLG expansion slots • Ultra-thin, high-capacity SmartMedia™ cards store voice and performance setups

SOUND ADVICE ▲▲▲▲

LOGOS

Logos Research Systems has released a massive collection of traditional hymns—more than 3,000 in all—as Standard MIDI Files. Entitled *Steve Green's MIDI Hymnal* (Win; $129.95), the CD contains the hymns as four-part MIDI files that represent SATB vocal parts. An included application lets you play these MIDI files without the need for additional software.

Included in the package is the *Logos Library System*, an application that lets you view the lyrics to hymns as well as organize and search for related text. You can search through the collection by author, title, and subject. You can also search by words and phrases, and an index lists the Biblical text that was the basis for the hymns.

Also included on the disc are several companion volumes, including *101 Hymn Stories, 101 More Hymn Stories,* and *Amazing Grace,* which detail the history behind songs such as "Jesus Loves Me," "Silent Night," and "Amazing Grace." *Spiritual Lives of Great Composers* is a collection of stories illuminating the lives of Handel, Bach, Schubert, Mendelssohn, Liszt, and others. Finally, the entire text of the King James Version of the Bible is included, along with Strong's Numbers, which let you easily identify and search for underlying Greek and Hebrew words in the English text. To use the *MIDI Hymnal*, you'll need a 386 or better PC; Windows 3.1, 95, 98, or NT; and a sound card. Logos Research Systems; tel. (800) 875-6467 or (360) 679-6575; fax (360) 675-8169; e-mail sales@logos.com; Web www.logos.com or www.logosbiblesoftware.com.

Circle #404 on Reader Service Card

▶ RAREFACTION

Rarefaction's *Digital Dysfunctions* (Mac/Win; $149) is a CD-ROM containing 16-bit, 44.1 kHz mono, stereo, and split-stereo AIFF files created by Chris Grigg. Grigg, a software

programmer and member of the found-sound collage group Negativland, compiled these tracks from his personal collection.

The sounds have been digitally mangled in Digidesign's *Sound Designer II,* BIAS's *Peak,* or Tom Erbe's *SoundHack* to the extent that there are not only inconsistent levels but also radical jumps in volume. Audio examples of misused or broken digital audio gear, video game–style sound effects, and groups of nonpitched sounds are listed with tempos in bpm and have zero crossings at their ends for easy looping. Other groups include files that have been pitch-shifted, time-compressed, or simply slowed down or sped up. One group of files was derived from Grigg's television and another from his voice. Still other files include bleeps, blips, buzzes, sirens, thuds, and ticks.

Text files provide instruction on how to make sounds similar to those included on the disc. Other files include the complete text of the U.S. Copyright Act's Fair Use Law, essays by Negativland, discussions of court decisions regarding Fair Use cases, and the complete transcript of a guerrilla interview with U2 guitarist the Edge by the members of Negativland, following Island Records' headline-making lawsuit against the band and SST, their label at the time. Rarefaction; tel. (415) 333-POKE; fax (415) 333-5022; e-mail paul@rarefaction.com; Web www.rarefaction.com.

Circle #405 on Reader Service Card

▼ KID NEPRO

The Kid has been gathering sounds for *Millennium II* ($200), the second set of sounds made for the Akai MPC 2000. The collection can be purchased on CD-ROM or Zip disk and features new samples of classic synths such as the Sequential Circuits Prophet 5 and Prophet VS, Fairlight, Korg Wavestation, Roland Juno, and others. There are drums collected from the company's hip-hop sound libraries and samples from Simmons MIDI drum brains, as well as acoustic drum samples. New recordings of instruments range from Fender Rhodes electric pianos to East Indian percussion.

Another section of *Millennium II* highlights city and traffic noise. These urban ambiences were recorded in New York City using a Tascam DA-P20 portable DAT and a pair of Sennheiser MD 431 mics. The recordings were then digitally edited in Macromedia's *Sound Edit 16.* You'll find sirens, honking, passing vehicles, and street ambiences. If the sound of the city gets to be too much, you can load more peaceful sounds, including forest and jungle ambiences and animal sounds.

You can also buy the CD-ROM as a set with its predecessor, *Millennium,* for $300. Kid Nepro; tel. (718) 642-7802; fax (718) 642-8385; e-mail kidnepro@aol.com; Web www.kidnepro.com.

Circle #406 on Reader Service Card

ADAT or Hard Disk? Yes!

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ALESIS

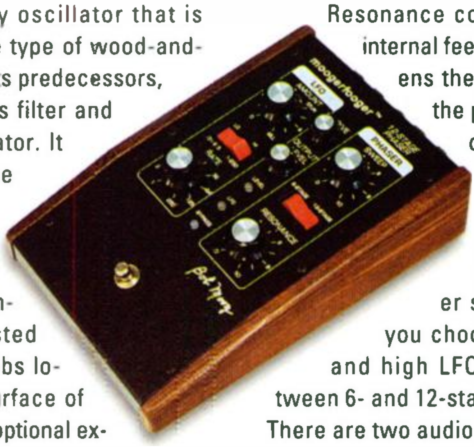
KEY CHANGES

Adaptec has purchased German software developer **CeQuadrat,** makers of CD-R and CD-RW recording software such as *WinONCD* and *justaudio.* Adaptec has also entered into an exclusive partnership with **MP3.com.** Under the terms of the agreement, the two companies will cross-promote products and develop others to facilitate burning CDs of downloaded MP3 files...**AKG** announced that a breakthrough in diaphragm manufacturing techniques has allowed the manufacturer to reduce the retail price on several of its key large-diaphragm microphones. These mics include the SolidTube, now $1,398, and the C 414 B/ULS and C 414/B/TL II, now $1,258 and $1,398. The mics previously retailed for $1,500, $1,332, and $1,751, respectively. All C 414 mics will now ship with an H100 shock-mount...Nonprofit organizations **Electronic Music Foundation** and **Leonardo** have formed the EMF/Leonardo Guide to the World (www.emf.org), a Web-based event calendar, news broadcaster, and e-mail update list that focuses on new music and media...**QSC** is offering a free seven-minute video that discusses the company's PowerWave technology and other key features of its PLX series of power amplifiers. To receive your free copy, contact QSC at (714) 957-7100...**Parts Express,** distributor of electronic parts, stage lighting, test equipment, loudspeaker drivers, and more, has released a free catalog. To receive it, call (800) 338-0531, or order it on the Web at www.partsexpress.com...**Audio Ease** has made its *Make a Test Tone* v. 2.0 software available as freeware from the company's Web site, www.audioease.com. The Mac-compatible software generates sine waves that can be saved as 16- or 24-bit sound files in SDII, AIFF, WAV, or Ensoniq Paris format.

—Rick Weldon

BIG BRIAR MOOGERFOOGER MF-103

The good Dr. Moog and his staff at Big Briar have been at it again: the Moogerfooger MF-103 ($399) has arrived. The MF-103 is a 12-stage phaser and low-frequency oscillator that is housed in the same type of wood-and-steel enclosure as its predecessors, the MF-101 lowpass filter and MF-102 ring modulator. It also boasts the same control over LFO rate, LFO amount, sweep, and resonance. These parameters can be adjusted either by using knobs located on the top surface of the MF-103, or with optional expression pedals fed in through the unit's ¼-inch TRS jacks. You can feed a control voltage on a 2- or 3-conductor cable in order to control the parameters.



The MF-103 accepts instrument- and line-level source signals between –16 and +4 dBm through its unbalanced ¼-inch input connector. You use the LFO Rate control to vary the sweep rate from 0.01 to 2.5 Hz or from 1 to 250 Hz; the LFO Amount control defines how much the LFO signal sweeps the phasing. The Sweep control lets you sweep the phasing over a six-octave range, and the Resonance control introduces internal feedback that sharpens the overall sound of the phaser. Additional controls include input-drive and output-level knobs, a bypass switch, and rocker switches that let you choose between low and high LFO range and between 6- and 12-stage phasing.

There are two audio outputs, each on unbalanced ¼-inch connectors. An LFO output lets you use the unit to modulate other voltage-controlled devices. The MF-103 is powered by a standard 9V adapter (included). Big Briar, Inc.; tel. (800) 948-1990 or (828) 251-0090; fax (828) 254-6233; e-mail info@bigbriar.com; Web www.bigbriar.com.

Circle #407 on Reader Service Card

OPCODE FERMATA

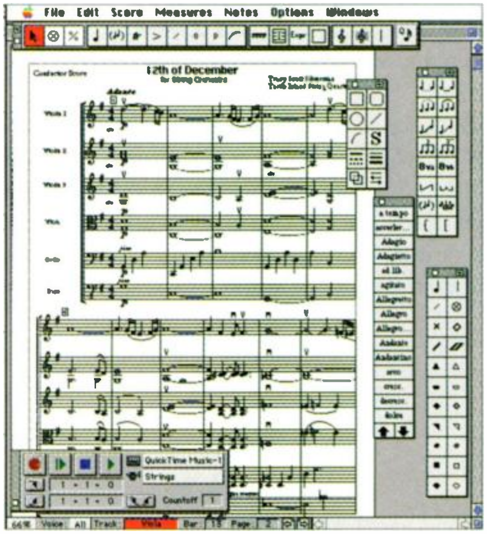
Opcode's *Fermata* notation software package (Mac; $59.95) is available only as a download from the company's Web site. The program's interface is designed for simplicity, and you can access most functions with a single keystroke or mouse click.

The program features OMS support, MIDI recording and playback, and graphic editing of MIDI data using the Strip Chart (a feature familiar to users of the company's *Vision* series software). You can change MIDI Velocity settings so that they follow dynamics markings.

With the program's Jazz Articulation function, you can notate glissandos, bends, slurs, grace notes, and falloffs. An EPS file-capture function lets you save your work as high-resolution EPS files for quality printing. Support for Standard MIDI Files and lyric writing is included.

The package also provides the complete Bach Chorales and several files that can be used as templates. *Fermata* comes with a manual in Adobe PDF format but is sold without technical support of any kind. A 68020 or faster Mac running Mac OS 7 or later is required. Opcode Systems; tel. (650) 429-2400; fax (650) 429-2401; e-mail info@opcode.com; Web www.opcode.com.

Circle #408 on Reader Service Card



▶ KORG D16

Korg's D16 portable digital studio ($2,399) expands and improves upon the capabilities of the company's popular D8. The new device samples at 44.1 kHz and employs 24-bit A/D/A converters and 32-bit internal processing. It can record 8 16-bit tracks simultaneously and play back 16 tracks, or record 4 and play back 4 24-bit tracks. Each physical track is associated with eight virtual tracks. The D16 ships with a 2.1 GB hard drive, and a SCSI connector lets you link up to seven external hard drives.

The 8-bus digital mixer can handle up to 24 tracks, so you can play back 16 recorded tracks while routing, mixing, and adding effects to eight other signals. Each of the 16 channels has a 3-band EQ with sweepable mid. You get scene automation as well as automatable faders, pan knobs, EQ settings, and effects.

Eight mono insert effects, two stereo master effects, and one stereo final effect are available at any time. There are 128 preset and 128 user-programmable insert effects, and 64 preset and 64 user locations are available for master effects. Korg's REMS (Resonant structure and Electronic circuit Modeling System) effects, which emulate speakers, cabinets, instrument bodies, and more, are included. There is also a chromatic tuner, a built-in condenser mic, and an 8-voice PCM drum machine with 200 preset rhythm patterns.

Input on the D16 includes eight balanced ¼-inch inputs, two Neutrik balanced ¼-inch/XLR combination connectors, and an unbalanced, high-impedance guitar input on a ¼-inch connector. Analog outputs include stereo RCA master and monitor outs, two unbalanced ¼-inch aux sends, and a ¼-inch stereo headphone jack. Stereo S/PDIF optical digital outputs are also provided.

Korg rates the unit's frequency response at 10 Hz to 20 kHz (±1 dB), S/N ratio at 100 dB, and TDH+N at 0.02%. Korg USA, Inc.; tel. (516) 333-9100; fax (516) 333-9108; e-mail product_support@korgusa.com; Web www.korg.com.

Circle #409 on Reader Service Card

▶ SONIC FOUNDRY VEGAS PRO

Sonic Foundry is shipping its long-awaited *Vegas Pro* 24-bit, 96 kHz, multichannel digital audio recording and editing software (Win; $699). The program syncs to MTC, supports multiple sound cards, and uses asynchronous I/O, which is designed to allow for better throughput of data to and from disk. The software's multithreaded architecture allows for the use of computers with more than one processor and for simultaneous processing of multiple tasks.

Vegas Pro supports multiple file formats, including AIFF, AVI, WAV, and MP3; can import QuickTime movies and BMP graphics files; and can export Internet-specific files such as ASF, WMA, and RealNetworks' RM files. The program's real-time resampling allows you to mix files with multiple bit depths and sampling rates without a separate conversion stage. You can even use AIFF and WAV files within a single track. The software offers unlimited undo and redo capabilities, and you can simultaneously record and play back as many tracks as your computer and audio interface can handle.

DirectX plug-ins are organized in a Plug-In Manager window. You get 32 effects sends, 26 output buses, and 26 aux sends; dithering and noise-shaping plug-ins can be applied on each bus. The program features 4-band graphic EQ and compressors as insert effects on every track. You can choose from several crossfade types and can scrub through audio and video.

Vegas Pro requires a 200 MHz or faster Pentium; Windows 95, 98, or NT 4.0; and 32 MB of RAM. Sonic Foundry; tel. (800) 577-6642 or (608) 256-3133; fax (608) 256-7300; e-mail sales@sonicfoundry.com; Web www.sonicfoundry.com.

Circle #410 on Reader Service Card

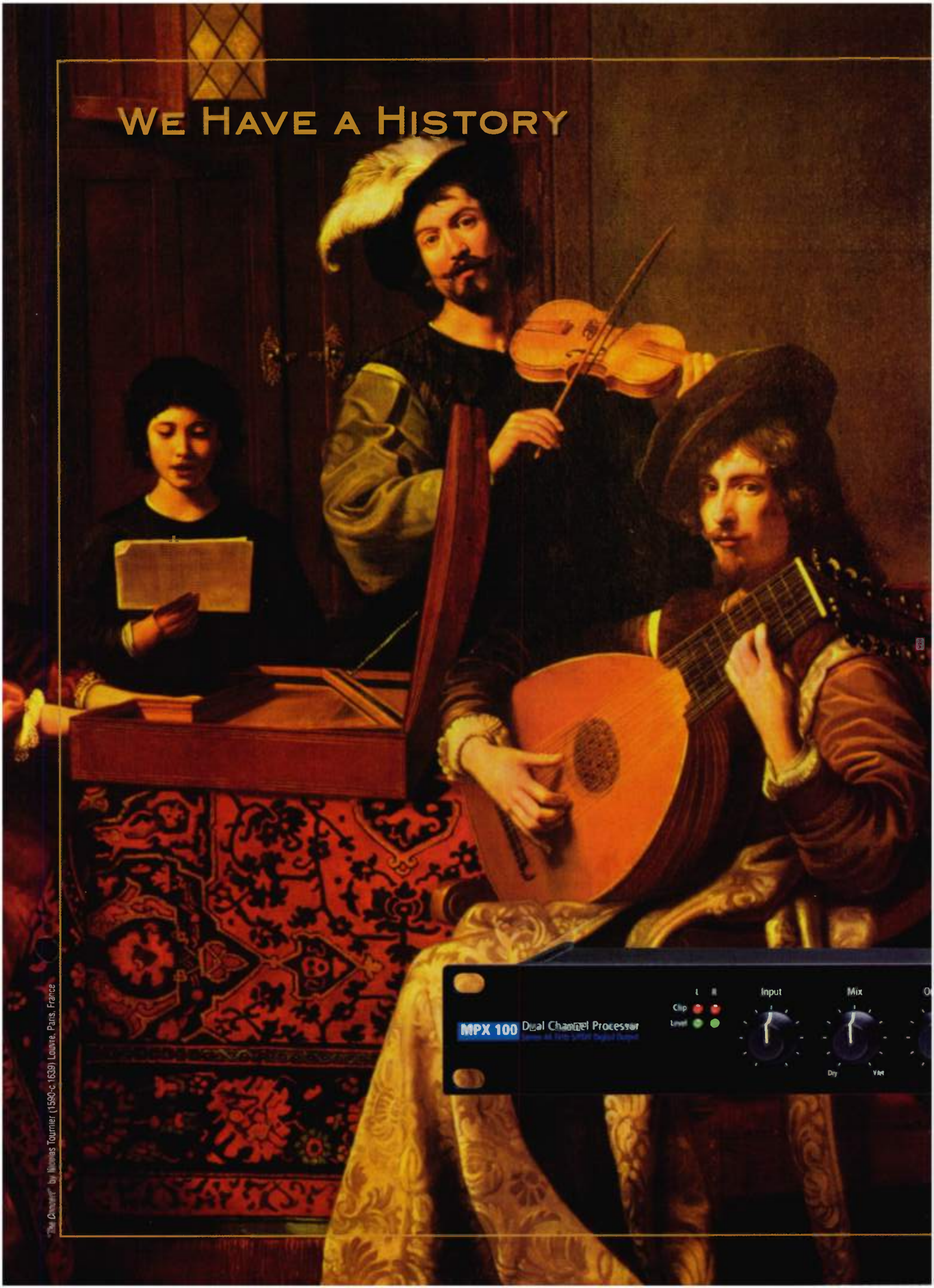
▼ MOTU 308

Mark of the Unicorn's 308 ($695) is a 1U rack-mount, 24-channel digital audio expansion interface for MOTU's 2408 and 1224 recording systems. It can be used with any digital audio software that supports WAV or ASIO drivers.

The 308 offers eight channels each of AES/EBU (on XLR connectors), electrical S/PDIF (on RCA), and optical S/PDIF I/O. All 24 inputs and outputs are simultaneously available for transfer of up to 24-bit audio at 44.1 or 48 kHz. The unit provides wordclock I/O on BNC connectors. You can connect up to three 308 interfaces via the MOTU PCI-324 card's Audio Wire jacks.

The front panel of the unit has sampling-rate indicators and offers a signal-present LED for each input and output. In addition to its main application as an expander, the 308 can be used as a stand-alone format converter. Mark of the Unicorn, Inc.; tel. (617) 576-2760; fax (617) 576-3609; e-mail info@motu.com; Web www.motu.com.

Circle #411 on Reader Service Card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OF MAKING GREAT MUSIC

Lexicon's award-winning MPX processors have made history with multi-effects that work the way you want them to – giving you professional studio quality effects at an affordable price. Whether you're looking for the best presets on the market, easy editing, or total control over your effects, the MPX Series delivers it all, without compromising the superior sonic quality you expect from Lexicon.

MPX 1

With high-quality audio effects, extensive control and easy operation, the MPX 1 has set a new standard for sound, flexibility and feature depth. 56 Pitch, Chorus, EQ, Modulation, Delay and Reverb effects give you an enormous array of sounds to combine and configure in any order you want – as many as five at once, including uncompromised reverb available at all times.

200 superbly-crafted presets show off the MPX 1's versatility, and an intuitive, interactive front panel lets you make quick edits, or completely re-structure any program. Packed with innovative design features and equipped with professional connections, the MPX 1 is ready to plug and play in any situation: in a stage rig, in a PA rack, in the studio or plugged into a digital workstation.

$899 List

MPX 100

With true stereo dual-channel processing, 24-bit internal processing, 20-bit A/D-D/A and S/PDIF digital output, the affordable MPX 100 packs more features than any box in its class. 240 presets exploit classic reverb programs such as Ambience, Plate, Chamber and Inverse, as well as Tremolo, Rotary, Chorus, Flange, Pitch, Detune, 5.7 second Delay and Echo. Dual-channel processing gives you completely independent effects on the left and right channels.

A front panel Adjust knob allows instant manipulation of each effect's critical parameters and an Effects Lvl/Balance knob lets you control effect level or the balance of dual effect combinations. Tempo-controlled delays lock to Tap or MIDI clock, and Tap tempos can be controlled by audio input, the front panel Tap button, dual footswitch, external MIDI controller or MIDI Program Change.

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PERFORMANCE TOOLS ▲▲▲▲

▲ ZOOM

Two new multi-effects processors from Zoom provide new choices for guitarists and bassists looking to add to their stompbox arsenal. The effects on the GFX-707 and BFX-708 ($229 each) are optimized for guitar and bass, respectively, but otherwise are identical. The processors each combine a drum machine, a 6-second riff sampler with half-speed playback capability, a 2-second phrase sampler, and 48 DSP effects. Each unit offers an expression pedal that controls volume, wah wah, and other effect parameters. The phrase sampler lets you play along over a sampled phrase, reverse it, and use the expression pedal to simulate turntable scratching.

The drum machine has 45 preset rhythm patterns using PCM samples. Internal samplers let you sample sounds and create multilayered effects in live performance. There are 60 patch memory locations (30 user, 30 preset) for storing effects settings.

The processors deliver 48 effects, including distortion, reverb, delay, ring modulator, doubler, and more. Onboard knobs let you adjust effect parameters in real time. You can use up to nine effects simultaneously. Other features include amp and acoustic-guitar tone simulators and a chromatic tuner.

An LED shows program and bank information. There is a stereo headphone output on a ¼-inch TRS connector. Zoom/Samson Technologies Corp. (distributor); tel. (800) 328-2882; fax (516) 932-3815; e-mail sales@samsontech.com; Web www.samsontech.com.

Circle #413 on Reader Service Card

▼ PRO MUSIC

Editable karaoke, anyone? Italian hardware and software manufacturer Pro Music has released *WinLive 3.0 Professional* (Win; $215), a program that lets you play and edit format 0 and 1 Standard MIDI Files on the fly. Editing features include the ability to adjust tempo, pan, and Velocity settings; send Program Change messages; transpose tunes; mute channels; and add chorus and reverb.

You can create and manage playlists of up to 40 songs each and view the files in a number of ways. One window lets you view songs by their order in a playlist, providing information on the artist and the MIDI file's name. Another shows default and saved tempos, transposition by a positive or negative number of semitones, and muted channels.

Three windows are provided for karaoke performance. Each shows lyrics in large type. One shows lyrics only; background images can be inserted behind the text in this window. A second window provides large lyrics in the upper half of the screen, with the current chords and a syllabic breakdown of the lyrics on a time line shown below. A third shows lyrics and MIDI instrument information by channel.

The program is compatible with GS, GM, and XG synths and Creative Labs Sound Blaster cards. *WinLive 3.0 Professional* requires a PC with at least a 386 processor and 4 MB of RAM, running Windows 3.1, 95, 98, or NT. Pro Music; tel. 39-080-395-8811; fax 39-080-395-8812; e-mail webmaster@promusic.net; Web www.promusic.net.

Circle #414 on Reader Service Card

▼ MAXON

Reissues of several 1970s-era Maxon effects pedals are now shipping. The Japanese pedals can be powered by a 9V battery or with an optional AC power adapter ($24.95). All feature silent FET electronic switching and an on/off status LED.

The OD808 overdrive pedal ($195) is a solid-state device that emulates tube overdrive as it boosts your signal level. The AD80 analog delay ($250) provides delay of up to 300 ms and is designed to impart a rich, warm analog tone. It has two outputs and controls for delay time, feedback, and blend. The PT999 Phase Tone ($199) features a single knob to vary the speed of phasing. The D&S distortion and sustain pedal ($189) also emulates tube distortion while adding to your instrument's sustain. It features controls for distortion, tone, and balance. A less aggressive version, the D&S II ($149), is also available.

Other reissued pedals include the CS-505 stereo chorus ($219) and FL-301 flanger ($219). Godlyke (distributor); tel. (973) 835-2100; fax (973) 835-2100; e-mail godlykehq@aol.com; Web www.maxonfx.com.

Circle #412 on Reader Service Card

Let the "critics" tell you how easy the Spirit Digital 328 mixer is to use...

Spirit's Digital 328 represents a new way of thinking in digital console design—it bridges the gap between analog ease-of-use and digital sound quality and features.

George Petersen of* Mix *says: "There are more than a dozen entries in the 'low cost' category of digital consoles, but in terms of pricing, performance and fast, logical interface, the Digital 328 clearly sets itself apart from the pack."

Take a few moments to read what he and other "critics" say about the Digital 328. Then, go to *www.spiritbysoundcraft.com* **on the web for more information. If you're in the market for an *affordable* digital console, you need look no further.**

On 328's user interface:

"The 328 is a real console interface that immediately feels as close to your comfortable old analog board as you could want... the consideration that has gone into every single button, knob and interconnect is striking." – Recording

"I liked the user interface a lot, and given that the most-requested features and digital interfaces are all included, the price is excellent." – Electronic Musician

"I like this board. It has a logical interface and enough knobs for fast operation (as such it could be ideal in a live performance or broadcast situation) while its audio performance is clean enough for any recording application." – Mix

On 328's E-Strip:

"The invention of the E-Strip is a stroke of genius, [giving] instant access to all controls at once on the selected channel." – Audio Media

"The 328 is fast and intuitive, thanks in large part to its 'E-Strip' interface. There are no subroutines or hidden pages; anyone familiar with an analog console can sit down at a 328 and be working in a matter of minutes." – Mix

"With Spirit's clever E-Strip design, this digital desk has the feel of an analogue." – The Mix (UK)

On 328's equalization:

"...To my ears, this is one of the most musical sounding digital EQs I've ever heard." – Recording

"[One] of the best features of the desk: carefully tailored to provide control ranges similar to those on a top-notch analogue console, it is (dare I say) very musical." – Audio Media

On 328's effects:

"A strong selling point for this unit is the pair of built-in stereo Lexicon effects... Having quality effects in the digital domain makes for clean sounds." – Electronic Musician

On 328's automation:

"The automation is straightforward to set up and works well." – Audio Media

"Between the user setups, snapshots and dynamic automation, the 328 remembers everything except the line-input trims and 100Hz rolloff switches. It's easy to get used to this way of working." – Electronic Musician

On 328's connectability:

"Clearly, the Digital 328 provides a multitude of configuration options suitable for project studios, post-production facilities, radio stations and even live applications." – Electronic Musician

"The 328 interfaces to practically anything digital." – Recording

On 328's unbeatable value:

"All in all, the British have indeed landed with a winner. The more you use this board, the more you will discover its depth and power. With one of these consoles, you could start a musical revolution of your own." – Electronic Musician

"This mixer packs a mighty punch for $5,000 [suggested list price]. It sounds excellent, does an excellent job of untangling all the various digital formats in use, and has an excellent interface. A bold step forward in digital console design." – Recording

"I like this desk! There's nothing better out there right now than the 328." – The Mix (UK)

On 328's mic preamps:

"The mic preamps have plenty of headroom... I was surprised at the clarity of the most subtle nuances of the percussion, including the last hint of sound from the bell trees and chimes." – Electronic Musician

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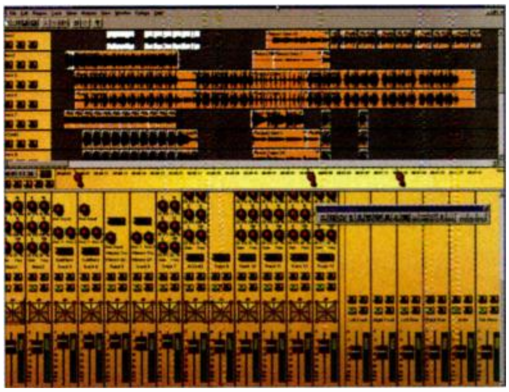
MINNETONKA MX51

Minnetonka's *Mx51* v. 2.0 software allows you to use a Yamaha DSP Factory or Digital Audio Labs V8 to author Dolby Digital 5.1 surround sound. The V8 version lists for $1,500 (upgrades from *MxTrax* $901), and the DSP Factory version costs $895 (upgrades from *MxTrax* $396). The program retains all of the editing and mixing features of *MxTrax* and adds 5.1 busing, panning, and mastering.

Mx51's main window, the Surround Mixer, lets you custom-build your mixing surface: you can drag and drop faders; pan controls; insert effects; solo, mute, and automate buttons; aux sends; and other controls. You can drag an automatable Surround Panner window onto any mixer channel and use it to localize sounds in a 5.1 soundstage. You can also



drag a SubBass Crossover filter onto any channel, which allows you to send the lows to a subwoofer or remove them from the surround channels. You can set each filter's crossover frequency separately.

The V8 version supports V8 plug-ins, and the DSP Factory version supports DirectX plug-ins. You can link *Mx51* to the company's optional *SurCode* 5.1 surround-sound encoder application ($995) in order to save a 6-track master as a Dolby AC-3 file.

Mx51 generates and slaves to MIDI Time Code. The V8 version also syncs to video as either a master or slave. An optional Microsoft Force-Feedback joystick ($138) can be used to position sounds.

The program requires a Yamaha DSP Factory or DAL V8 system, a Pentium 200 MHz or faster PC, 64 MB of RAM, and Windows 95, 98, or NT. Minnetonka Audio Software, Inc.; tel. (612) 449-0187; fax (612) 449-0318; e-mail info@minnetonkaaudio.com; Web www.minnetonkaaudio.com.

Circle #415 on Reader Service Card

FOSTEX VM-200/VR-800

Fostex has released two complementary products: the VM-200 digital mixer ($1,499) and the VR-800 digital multitrack recorder ($749 without media; $999 with external 250 MB Iomega Zip drive; $1,099 with internal 5.1 GB IDE hard drive).

The VM-200 is a 20-input digital mixer that offers 32-bit, RISC-based signal processing and mixing of 16-bit, 44.1 kHz digital audio signals. The top surface of the VM-200 gives you direct access to pan and EQ functions, providing 12 rotary controls and an interactive, 256 x 64–dot, backlit LCD. It has eight 60 mm, motorized channel faders and one stereo master fader.

Each channel features a 4-band parametric EQ, analog input-level control, and illuminated buttons that edit different functions depending on which mode you are in. Fostex's proprietary Advanced Signal Processing effects-processing technology is designed to offer higher gain in the early reflection characteristics of reverbs and to be free of artifacts. Effects include reverb, delays, chorus, flange, pitch shift, and preset combinations of these. Two stereo effects can be used simultaneously on each input. The effects, faders, pan, and EQ settings can be automated, and 100 scenes can be stored in memory. There is also a library in which to store 50 EQ presets.

There are eight analog inputs; channels 1 to 4 use balanced XLR connectors and channels 5 to 8 use balanced ¼-inch connectors. The XLR inputs provide 48V phantom power. Input for channels 9 to 16 is via an ADAT Optical input, and channels 17 to 20 are accessed by using the balanced ¼-inch effects return jacks. Each mixer can be completely controlled via MIDI.

Stereo S/PDIF digital I/O is available on RCA jacks and can be used to link two units in Cascade mode. Eight-channel ADAT Optical digital I/O is also provided.

Four effects sends and four returns are provided on unbalanced ¼-inch connectors. Stereo analog master output is on two RCA connectors. You get MIDI In, Out, and Thru connectors and wordclock I/O on BNC connectors.

The VR-800 digital multitrack recorder can simultaneously record up to 8 tracks of digital audio, and 16 virtual tracks can be stored. Each track is recorded uncompressed at 16-bit, 44.1 kHz resolution. The unit does not have analog I/O; all audio input and output is in the digital domain.

The recorder offers nondestructive editing with copy, paste, move, and erase; audio scrubbing via a jog/shuttle wheel; autopunch recording; vari pitch of ±6%; and one level of undo/redo. There are 99 edit and 99 locate points.

The unit syncs to word clock (on BNC connectors) and MIDI Clock, slaves to MTC, and outputs MMC. The recorder has a SCSI port for storage and backup. Fostex Corporation of America; tel. (562) 921-1112; fax (562) 802-1964; e-mail info@fostex.com; Web www.fostex.com.

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PRO FILE

Hell Is for Chillin'

Ming + FS supply dense, funked up hip-hop.

By Rick Weldon

Based in New York's dense Hell's Kitchen neighborhood, Madhattan Studios is home to Ming + FS (aka Aaron Albano and Fred Sargolini). Aside from remixing for the likes of Biggy Smalls, Coolio, and Li'l Kim, the two also release albums for other artists on their own Madhattan label.

On *Hell's Kitchen,* their debut release for Om records, Ming + FS hot-rod their own hip-hop with Rhodes piano, drum 'n' bass trappings, and vocal performances from a variety of New York–based MCs. The album was recorded at Madhattan, an apartment that the two musicians share.

Each room is used in a particular way. According to Ming, "Most of what we do is sample, sequence stuff, or play live to taped stuff. We have DAT machines, samplers, two DA-88s, and a bunch of keyboards." All of their beats come from the samplers and the sequenced synths, and the DA-88s are used mainly for vocals and live instrumentation. A second mixing board resides in another room and is used mostly for live tracking. "That would be my bedroom," laughs FS. "I can record drums or sample breaks from records there; I just move the DA-88s in to track drums. Sometimes I'll go directly from the board to a DAT. For a lot of stuff, like bass lines and drum loops, I don't bother going to an 8-track because I can go straight to DAT, put it in the sampler, and tweak it from there.

When miking drums, FS often uses just one mic, positioned on the floor. "We almost never use a sampled break by itself. We just add some room sound to the mix of the kicks and snares and hi-hats that we've already sampled from records."

Ming adds that, overall, the process sounds better "because you maintain the feel by playing along live. But we layer all the drum sounds. Sometimes we'll want to change the sound of the beat throughout the song, and we can do that by changing the layering of the sound. We'll also try compressing and overdriving the channel again to bring out the harmonics, make the sound even thicker, or make it sound like a keyboard or a bass tone."

When recording the MCs who appeared on *Hell's Kitchen,* FS doubled each vocal track. But the intricacies of a doubled rap performance didn't pose a problem. "For singers, there's pitch and rhythm. For rap, it's just rhythm, so it's actually easier," he says. Ming interjects, "You'd be surprised to see a guy smoke a bag of the craziest dope you've ever seen and then hit the same off-time rhythm perfectly, twice in a row."

"Also, there's a different mentality," says FS. "It's not like doing pop music or rock. When you double a rap, you're going for more of a gang feeling; it's a reinforcement. So sometimes it's good to have some messiness—a little bit, anyway; it helps cover up their bad breath control from smoking all that weed."

To give MCs a starting point, Ming + FS prepared a rough version of the songs. "It would basically be a loop of something for four minutes, with a couple of drum dropouts just so you don't get bored out of your skull," says FS. "They write their lyrics, and then we formulate the arrangement together."

This give and take makes for more interesting music, but it also requires a greater amount of time and energy. "I don't think people comprehend the kind of effort it takes to really switch it up," Ming comments. "We're looking at compositions, not necessarily at individual samples. After all, we still have to listen to it in the end, we still have to be able to have conversations like this about it. If there isn't much to talk about, then there's a problem."

For further information, contact Om Records: tel. (415) 575-1800; e-mail info@madhattanstudios.com; Web www.madhattanstudios.com.

Ming + FS bring new school to *Hell's Kitchen.*

electro-h

Big Muff PI

The pedal that started it all! The distortion countless musicians such as Hendrix and Santana relied on for its rich, creamy, violin-like sustain. A timeless piece, the Big Muff has been defining the sound of rock guitar for the past 30 years. From Pink Floyd to the Chemical Brothers to Korn, everyone wants a piece of the 'Pi!

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BassBalls

Back by popular demand! One of the most sought-after units in the Electro-Harmonix line, Bassballs is available again! Bassballs' twin dynamic filters track picking dynamics at slightly different frequencies, creating shifting peaks and notches that simulate human speech! Distortion mode further emphasizes vowel-like overtones and dynamic responsiveness, making Bassballs the ideal accessory for Bass or Guitar.

Stereo Polychorus

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Deluxe Electric Mistress

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Frequency Analyzer

Ring Modulator! One of the most sophisticated and unusual effects in the Electro-Harmonix line, the Frequency Analyzer, used by Devo and others, adds moving harmonies to the original note while controllable high order filters reduce cross product distortion. From tunable three-voice harmony to anarchic microtonal sounds, the Frequency Analyzer is an esoteric accessory for all instruments.

armonix

Stereo Memory Man

True Stereo Chorus! Dual outputs, 90 degrees out of phase with each other, provide ultra lush "phased" chorus with remarkable clarity and depth as well as 300ms of the warm, clean analog delay the Memory Man line is famous for. Slapback echo, controlled feedback and "bathtub" reverb are some of the other effects this versatile unit affords.

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Q-Tron+

The ultimate envelope follower! Same as the original Q-Tron but with added Effects Loop and Attack Response Switch. The Effects Loop lets you place an additional effect between the Q-Tron+'s preamp and filter section without changing the envelope drive. The added Response Switch lets the player choose between a slow, smooth, vowel-like attack and the fast, snaked response of the original Q-Tron.

Bass Micro Synthesizer

Analog synth for bass. The Bass Micro Synthesizer has the same features as the famed Micro Synthesizer but with a filter sweep range tailored for bass guitar.

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16, 20 or 24 bits, no matter how fine you chop, soulful guitarists feel the irregularities of digitally processed sound. That's why countless musicians rely on the flowing warmth of Electro-Harmonix analog effects. Available at leading music stores throughout the world.

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TECH PAGE

XML Marks the Spot

New Web languages may boost online music distribution.

By Scott Wilkinson

The importance of the World Wide Web for musicians should be obvious by now. MP3 sites abound, and anyone with a computer can distribute music to Net-nauts around the world without a major record label.

The Web is based on a computer language called HTML (HyperText Markup Language). This language uses special codes that indicate how text and graphics should look when they are accessed by a Web browser. It also supports the use of *hyperlinks,* those underlined words and phrases within a Web page that let you jump to other pages when you click on them.

Unfortunately, HTML suffers from one major drawback: it describes how information should look, but it offers no way to represent what that information means. As a result, a lot of bandwidth is wasted on appearances. For example, if a customer changes an online order, the remote server must resend all the text and graphics to update a few numbers while your computer sits idly waiting because it knows nothing about prices or shipping.

One solution to this problem is to use a more advanced markup language that represents what the information means, rather than how it looks. To create such a language, Internet designers use a *metalanguage*—that is, a language that is used to define other languages. For instance, HTML was created using the SGML (Standard Generalized Markup Language) metalanguage. However, SGML is quite complex, so a simpler metalanguage called XML (Extensible Markup Language) was developed from it. XML is said to offer 80 percent of the benefit of SGML with 20 percent of the effort.

Using XML, any industry can create its own markup language by establishing a *document type definition* (DTD) that represents the industry's information in a meaningful way. In addition, XML incorporates the concept of *stylesheets,* which allow the information to be presented in any number of ways on a wide variety of devices while retaining its basic meaning. For example, you could display a piece of musical data as notation on a standard computer screen or personal digital assistant (PDA), or it could be played on a Web-savvy audio system, such as a digital telephone. In addition, the remote server need not concern itself with appearances, which should speed up the World Wide Wait significantly.

In the music world, one promising development is MML (Music Markup Language) from the University of Pretoria in South Africa (http://is.up.ac.za/mml). MML uses tags to represent musical content (see **Fig. 1**), which can be presented as text, a piano-roll graphic (as in many sequencers), standard music notation, or MIDI files, depending on the receiving device and its associated stylesheet.

A number of other music markup languages also exist. One of the oldest of these is SMDL (Standard Music Description Language), a specific application of HyTime (Hypermedia/Time-based Structuring Language). HyTime was developed using SGML. Another example is MNML (Music Notation Markup Language), which is designed for Internet delivery of Western music notation.

The bottom line is that a music markup language can greatly facilitate the visual and audible distribution of music on the Internet. As these languages become more commonplace, we should see an explosion of Web-based music sites that make today's offerings look positively quaint.

Thanks to Steve Mounce, Sami Nybacka, Tim Bray, Steve Newcomb, and Jacques Steyn for their help with this article.

```
<clef>C          <bar 1>3[E A] [3E B] [3E:8 C]F:8 [3G C]
<octave>4        <bar 2>[A D] [3G D] ([3G E])2
<note>:4         <bar 3>[A D] [3E E] ([3E C])2
<upbeat>[A E]    <bar 4>[3E:2 B:2] R [B E]
```

FIG. 1: In this short MML file excerpt, simultaneous notes are enclosed in square brackets. The default rhythmic value is a quarter note; other values are indicated with a colon and number after the note name. The default octave is 4 (middle C is identified as 4C); other octaves are indicated with a number before the note name. Repeated notes are enclosed in parentheses followed by the number of times they repeat.

What's up with Dat?

An update on the little format that could.

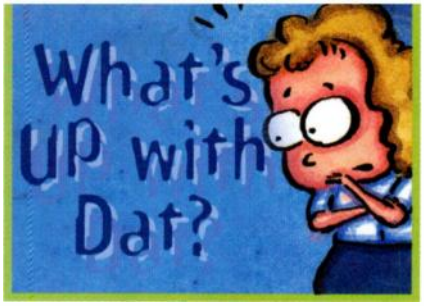
By Gino Robair

The use of affordable multitrack hard-disk recorders is sweeping the globe, but there's a tape-based technology that's taken for granted yet rarely used to its full potential: DAT.

DAT has become a universal medium for sharing digital audio. Despite stiff competition from CD-R and CD-RW, DAT continues to outperform other mastering formats and shows no signs of stopping.

The suggested retail prices of DAT machines varies widely—from $900 for a no-frills portable to $11,000 for a fully loaded, time-code-ready studio unit. In this article, I will examine a handful of familiar recorders that are within the budget constraints of most personal studios. But first, let's have a look at where DAT came from and the interesting things you can do with it.

Illustration by Charles Stubbs



WHERE DID DAT COME FROM?

In 1987, the year DAT machines began to appear at industry trade shows, home recording was much different than it is today. The Amiga, Atari, and Macintosh computers were au courant for electronic music, the cassette multitrack recorder was king, and most people didn't have a .com address on their business cards.

The choices for an affordable 2-track mastering deck were also different. Besides analog reel-to-reel and cassette formats, there was Beta and VHS Hi-Fi, as well as the Sony PCM-F1 digital format. With the less expensive analog formats, you had to use noise reduction to get optimal sound; with VCRs, you were stuck with an unpleasant compression/expansion noise-reduction scheme that the user couldn't disable; and PCM-F1 was sometimes unreliable and sonically not too desirable.

AND ALONG CAME DAT

DAT machines combine a couple of technologies familiar to our circa-1987 home recordist—the videocassette recorder (VCR) and a pulse-code modulation (PCM) processor. DAT machines store digital information on tape via helical scanning using a miniaturized rotating head. This technology was originally called Rotary-head DAT, or R-DAT for short. (Also in development at the time, though never commercially released, was a system called Stationary-head DAT, or S-DAT.)

Although DAT was originally intended as a consumer-oriented digital replacement of the ubiquitous Philips analog cassette tape, its success was nipped in the bud primarily because it could be used to be make direct digital copies of compact discs. The U.S. recording industry, with the help of Congress, immediately sued Sony to prohibit sales of DAT gear in the United States and lobbied hard for a copy-protection scheme. The result was that individuals who wanted an early DAT machine had to acquire a unit through the gray market if they wanted to avoid the copy-protection circuitry. This, of course, meant that you'd have to take your machine to Japan or Europe if it needed servicing—not very practical for most consumers.

FIG. 2: Sony PCM-R500

As it turned out, audio professionals found DAT to be more stable, easier to use and maintain, and better-sounding than the PCM-F1 format, and thus DAT survived. In fact, "professional" units even allowed the user to defeat the copy protection. Today DAT is still the most common 2-track recording medium used by pros, not to mention the legions of music fans that occupy the "taping section" at rock concerts.

In the beginning, DAT brought promises of better-than-CD audio quality, an end to wow and flutter, and a reliable digital format that didn't require a separate VCR. Of course, in these days of 24-bit, 96 kHz hard-disk recording, mere "better-than-CD quality" is not the draw it used to be. Still, DAT continues to provide a cost-effective medium for high-quality 2-track recording.

STAYING POWER

To many home recordists in the early days, DAT seemed like a transitional format. The biggest question was, "How long before I have to change formats again?" The question is still valid today, because everyone can see the writing on the wall regarding tape-based media: affordable, random-access, disk-based recording formats are everywhere, so why invest a grand or two in a deck that will be obsolete in just a few years?

For starters, digital audio tape has proven to be a robust, yet inexpensive and compact, medium for storing sound. Thanks to the popularity of the DTRS (which uses Hi-8 video cassettes) and ADAT formats, tape will be with us for a while longer—especially those of us with huge DAT, ADAT, and DTRS archives. And in the age of hard-disk recording and CD-RW, the mighty DAT machine still offers features that are unavailable in any other format: up to two hours of full-frequency digital audio without compression; high-quality, low-cost reusable media; and compatibility and portability (every 16-bit, 44.1 kHz digital audio tape can play in any DAT machine, including the portable ones).

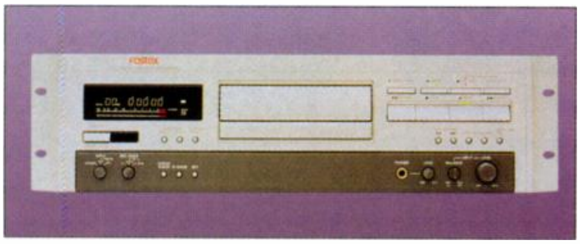


FIG. 1: Fostex D-5

LENGTH IS EVERYTHING

With DAT you can record up to two uninterrupted hours of music, at a higher resolution than the average CD. If you want to double the amount of record time (and audio quality isn't an issue), the ability to record in Long Play (LP) mode may be a determining factor in your choice of a DAT deck.

Although there are 180-minute DAT media on the market, manufacturers of DAT players for audio use recommend against using such tapes (see the sidebar "All DAT Media Are Not Created Equal"). If you need four hours of 44.1 kHz, 16-bit audio record time, Tascam's DA-302 dual DAT machine ($1,875) is the ticket. In Continuous Record mode, it will automatically switch to the second tape once the first has been fully recorded. It can also do this in Long Play at 32 kHz, yielding a total of eight hours of record time. Need more than that? Tascam claims you can hook up several DA-302 units in series to get an amount of record time limited only by your budget. In addition, each DA-302 tape well can be used independently thanks to the separate S/PDIF input/output, allowing you to dub off a copy from one side of the machine while recording with the other. The unit can also handle high-speed dubbing duties.

DAT FACE

That traditional-looking tape-deck exterior of the DAT unit (with play,

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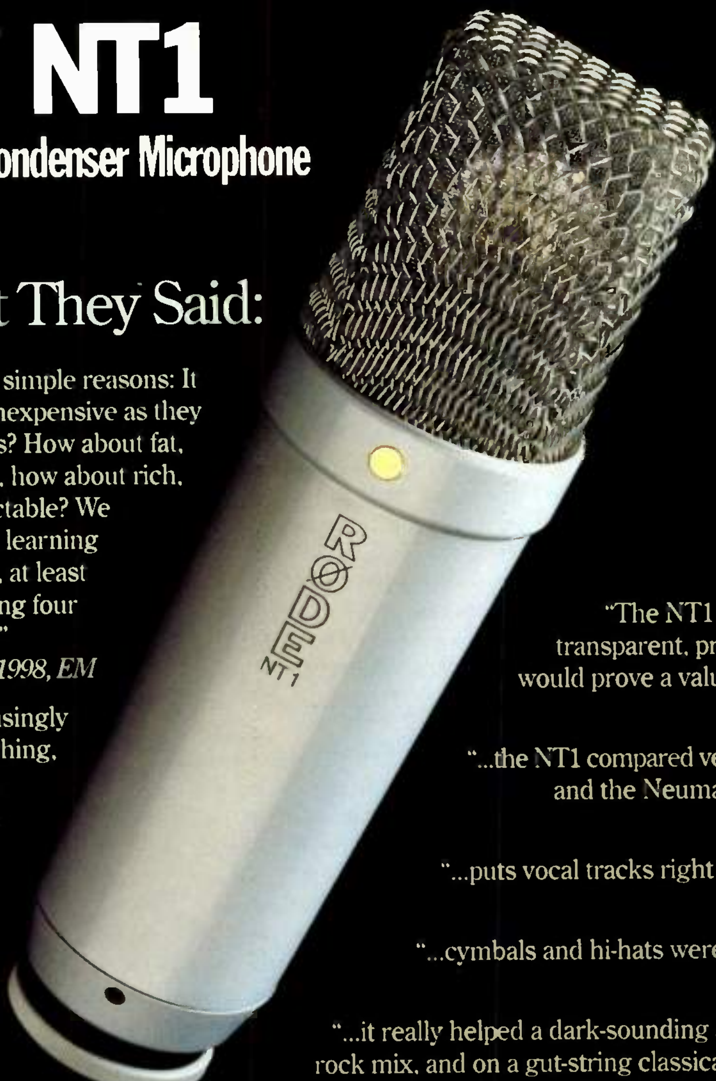

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—Brian Knave, April 1998, EM

"The NT1 has a rich, stunning sound—very transparent, present, and brightly detailed—that would prove a valuable addition to any mic cabinet."
—Brian Knave, April 1998, EM

"...the NT1 compared very favorably to both the AKG C414 and the Neumann U 87—and that's saying a lot!"
—Brian Knave, April 1998, EM

"...puts vocal tracks right in your face with startling clarity."
—Brian Knave, April 1998, EM

"...cymbals and hi-hats were reproduced exceptionally well..."
—Brian Knave, April 1998, EM

"...it really helped a dark-sounding acoustic guitar cut through a busy rock mix, and on a gut-string classical guitar, it captured the warmth of the instrument while detailing the high end and minimizing boominess."
—Brian Knave, April 1998, EM

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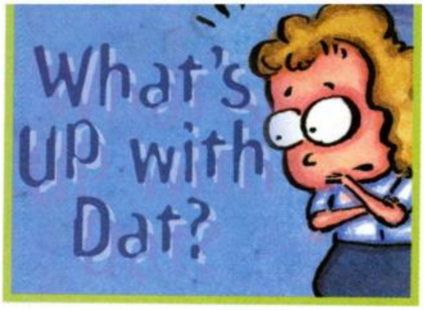


pause, record, fast forward, and rewind controls) belies its plentiful yet easy-to-use features that make DAT a powerful 2-track format. For instance, you can search for songs on your cassette deck using the Automatic Music Search feature, but DAT allows you to search for specific songs using program ID numbers.

As with other gear, the overall design of the DAT machine's user interface determines the way the product can be used. For example, if a particularly useful feature is embedded deep in the menus, it's unlikely that the feature will get much use. Therefore, it's important to know which features you will need in a DAT deck before you go shopping for one.

THE RESOLUTION ENVELOPE

One factor that will determine the "sound" of a particular digital recorder are its A/D converters. Most of the units I tested for this article have comparable-sounding converters (I'll discuss the exceptions below). But even though you're stuck with 16-bit resolution with a standard DAT deck, there's still room for improvement in sound quality.

External high-resolution A/D converters and their proprietary dithering schemes, such as Apogee's UV22 and Sony's Super Bit Mapping, yield results that are superior to those of standard 16-bit converters. Any time you use a higher-resolution converter, you increase your chances of using the full 16 bits you're allowed on DAT media.

Using outboard converters with a DAT machine is easy. The first step is to send your analog signal to the converter for translation into a high-resolution digital signal. Next, using proprietary circuitry, the digital word length is dithered to the 16 bits that your DAT can handle. Finally, this 16-bit digital material is sent to your recorder via a digital cable. This process lets you maximize the number of bits you're using in the 16-bit word, resulting in greater resolution and sonic transparency than you would get otherwise.

FIG. 4: Tascam DA-45HR

Hearing the difference that high-resolution converters can make will immediately open your ears to the world beyond 16 bits. Fortunately, the cost of these converters is dropping while the quality is increasing. No matter which DAT machine you choose, you'll be able to improve on the audio quality without having to buy a whole new unit.

SPEED OF SOUND

The choices of sampling rates available to the consumer haven't changed much since the first DAT machines hit the market. The biggest difference is that you can now easily purchase a deck domestically that records at 44.1 kHz without Serial Copy Management System (SCMS) encoding (more on that later). The widespread—and largely unfounded—fear in the record industry that average citizens would use DAT to illegally copy CDs (as well as prerecorded DATs) kept the earliest units from being able to record at 44.1 kHz. Thankfully, that nonsense is now history.

For those times when you need more than two hours of record time, many machines have a Long Play mode that allows you to double that time. In LP mode the tape speed is halved (4.075 mm per second), as is the head-rotation speed (1,000 rpm); and the word length is reduced to 12-bit nonlinear quantization. In addition, LP mode uses a lower sampling rate of 32 kHz—which results in a reduced upper frequency response of 14.5 kHz due to the Nyquist theorem. It is possible in some machines to record 16-bit words in Standard Play at 32 kHz, although I have difficulty imagining a use for this scenario in the personal studio.

FIG. 3: Panasonic SV-3800

MALLEABLE SUBCODE

Each DAT recorder allows the user to write and erase Start, Skip, and End IDs any time after the audio has been recorded. That means you can change misaligned IDs as well as erase erroneous ones: the IDs are part of the subcode; therefore, you never have to worry about accidentally erasing part of a song when you edit ID information.

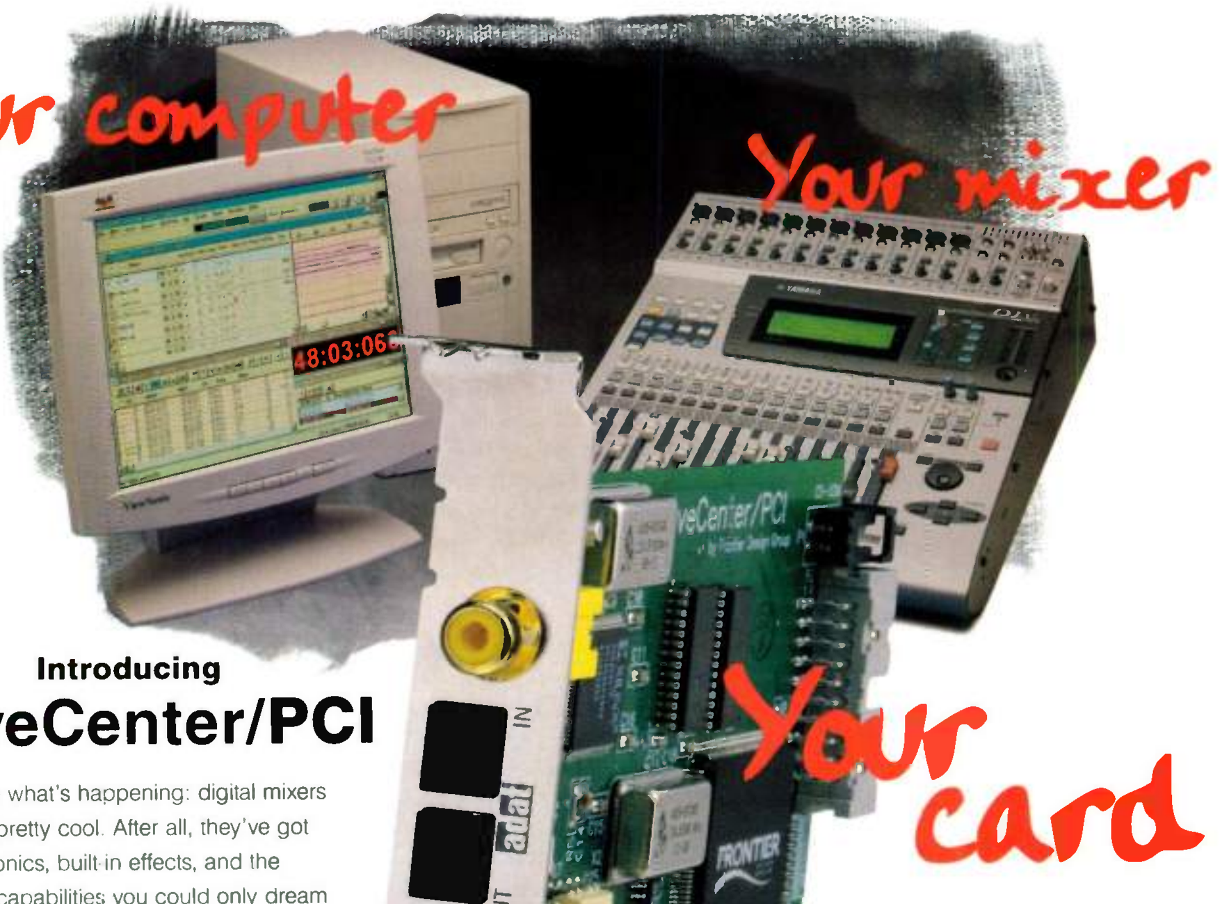
Because I never know when inspiration will strike (and because tape is cheap), I always have the DAT rolling when I'm playing. When I'm done, I listen back and put Start IDs at the points where there is music I want to access, and I add Skip IDs at the beginnings of sections I don't need. Then I press Renumber, and my DAT machine rewinds to the beginning of the tape and renumbers the Start IDs sequentially. On a unit like the Fostex D-5, this step also lets you create a Table of Contents on the tape so that you can easily access songs and timing information.

I also make a habit of adding an End ID when I finish recording for the day. This way, I can do an End Search and easily pickup where I left off.

One thing I take note of immediately is whether a particular DAT unit goes directly into Play when I do an ID search from Stop. The SV-3800 and the DA-45HR, for example, go into Pause in this instance. If you press a search button while these units are playing, they automatically go into Play when the correct ID is found. I prefer a deck that goes directly into Play every time I do a search; it saves me from having to reach over and push Play as well.

WHEELS OF FIRE

Yet another wonderful feature of the DAT machine is the shuttle wheel. On many machines, this handy device serves double duty: its primary function is to shuttle the tape forward and backward—from one-half to 15 times normal speed, depending on how far you turn the knob—so you can quickly locate points on the tape by ear. Shuttle wheels are sometimes used for data selection, because they are perfectly suited for



Introducing

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OK, you see what's happening: digital mixers are looking pretty cool. After all, they've got incredible sonics, built-in effects, and the automation capabilities you could only dream about before. But if you hook that puppy up to the NoiseRacket analog soundcard that came with your computer, you're right back in ****ville. (Rhymes with "Snapville.")

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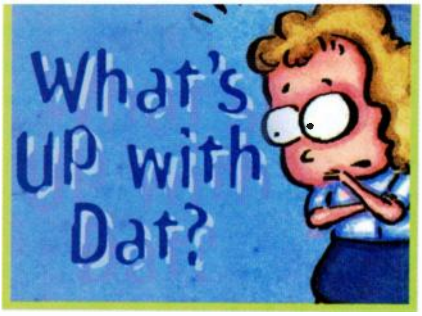


scrolling quickly through parameters.

The downside of the shuttle wheel is that it increases the cost of the recorder; you can easily save a couple of hundred dollars by eschewing this particular control. If you're looking for a second DAT machine (you *do* want to make backup copies of your work, don't you?), you can probably get by without a shuttle wheel if your primary deck already has one. However, I opted for a shuttle wheel on my second deck so that I could use it as a backup if my primary deck was down.

SCMS OF THE UNIVERSE

The solution that was adopted to control the number of "serial" digital copies allowed is called the Serial Copy Management System. SCMS controls the number of second-generation digital copies that can be produced with the IEC-958 Type II "consumer" digital interface format (commonly known as Sony/Philips Digital Interface Format, or S/PDIF). The setting of ID6 in the tape's subcode—not to be confused with Program number ID6—determines whether serial copying will be allowed.

Although SCMS is still with us, anyone with a pro-quality DAT recorder can relax because AES/EBU digital connections ignore this subcode information. However, it's good to examine the setup of your recorder so that you don't inadvertently put a "copy prohibited" code onto your tape; you never know when you may need to make a digital dub of your music using a S/PDIF connector, so make sure your recordings are "copy free."

THE PLAYERS

Currently, five manufacturers are distributing DAT machines in the United States—Fostex, HHB, Panasonic, Sony, and Tascam—for a total of 15 different models priced between $899 and $3,000. I chose four representative studio models within this range, as well as a couple of portable units.

Rather than use this section to run through all the features of each machine (you can examine the table for that information), I'll focus on specific features that make each unit unique among the group, as well as common features that may differ on some units or be missing altogether.

FIG. 5: Sony PCM-M1

Fostex D-5. Even though it was the least expensive studio recorder of the group, the D-5 ($1,029) performed as well as any of the units. It has one of the easiest interfaces to navigate (see **Fig. 1**). Everything you need is on the top level, so you won't find yourself descending deep into menus.

The D-5's analog input controls include a single input volume knob that

ALL DAT MEDIA ARE NOT CREATED EQUAL

If you've been around the computer industry at all, you know that DAT tape is also used as a back-up storage medium for data other than digital audio. Although both kinds of machines use digital tape, data-storage machines and tapes are substantially different in design than those made for recording audio. Many engineers I know won't use data-backup tapes for audio production due to the increased risk of dropouts and other problems.

These cassettes are often referred to as Data DATs, and they come in lengths that can yield up to 180 minutes of recording time. This makes them attractive to engineers needing three hours of recording time. However, manufacturers of audio DAT equipment recommend against using tapes that play longer than 120 minutes, and I'm going to recommend against using Data DAT tapes in general for music. To see why, we need to look closely at how DAT tapes are made, including the tape and the shell.

Like analog tape, digital audio tape has a magnetic coating that is attached to a plastic backing using an adhesive binder. Meter-wide sheets of this tape are manufactured and cut down into strips that are 4 millimeters in width for use in DAT cassettes. The most consistent layer of magnetic coating happens to be in the center of these large sheets, and this is the portion that is used for high-quality tape, especially for Data tapes. In addition, tape stocks come in different thicknesses, and that's where we begin to get into trouble.

The biggest problem with 180-minute tapes is that they are a thinner stock than that which is in tapes of up to 120 minutes, since there's only so much room inside a cassette shell. Audio DAT decks are optimized for a specific thickness and tension that shouldn't exceed that found in the 120-minute variety.

According to some manufacturers, one of the biggest reasons that Data DAT tapes don't particularly work well in audio decks has to do with the cassette housing or shell. Data DATs use a different, less expensive housing (which effects tension and tape movement) that isn't up to the standards that audio DAT machines require. Data machines and tapes have a wider tolerance for movement within the cassette than audio DATs—a tolerance that is beyond what audio recorders can safely handle. The cheaper shells can cause the tape to not track properly.

The common misconception is that tape formulation alone determines DAT-tape quality. However, it's the total package—tape and shell—that determines the ultimate stability of the DAT media.

controls both channels, as well as a center-detented left/right balance knob. Although I prefer a separate volume control for each input, a single-knob setup is not a problem if you're able to regulate audio levels before they get to the deck.

The D-5 is also the only studio unit of the bunch that doesn't come with a shuttle wheel. (However, pressing play and rewind or fast-forward simultaneously allows you to scan the audio in a similar way.)

One feature I found quite useful on the D-5 is that you can create a Table of Contents (TOC) in the subcode of a tape. The TOC lists the number of IDs on the tape, the total time of the recorded portion of the tape, the length of each piece identified by an ID, and the start time of each piece. This is a handy feature if you're sharing tapes with people who have the capability of reading TOC material.

Immediate access to all the basic features is at your fingertips on the front panel, including the choice of 48, 44.1, and 32 kHz (LP) sampling frequencies. But many of the buttons are tiny, such as the ID search buttons, Counter Mode and Reset, Renumber, and Wireless Off (which turns off the receiver for the wireless remote control). Fortunately, the buttons have enough space between them that you won't accidentally push two at the same time.

The only thing I found inconvenient on the D-5 was using the Margin Reset. The Margin is a common feature of DAT machines that gives you a reading of the input level in decibels. On the D-5, the display for the Margin numbers shares the same characters as the program ID numbers. Unfortunately, you can't just press the Margin Reset button to get a reading; you have to hold it down, because the display goes back to the program number when the button is released. I use the Margin reading to set record levels with a DAT deck, and keeping a finger on the Margin Reset button while adjusting levels can be a nuisance.

Like the front panel, the rear panel of the D-5 is a model of simplicity: XLR inputs and outputs for analog connection, and AES/EBU and optical digital I/O. Analog I/O level (–10 or +4) is designated using a back panel switch. (You select the type of digital I/O you want to use from the front panel.) There is also a 5-pin DIN plug that accepts a parallel remote (GPI) input connector, though the supplied wireless remote works just fine.

Sony PCM-R500. The PCM-R500 ($1,695), Sony's midprice studio DAT recorder, fell within our price range and features a motor transport with four direct drives (see **Fig. 2**). The R300 ($995) is similar in many ways to the R500 but doesn't include a shuttle wheel, while the R700 has everything the R500 has and more. In fact, the latter two models are so closely matched that they share a manual.

The R500's shuttle wheel is actually two controllers in one. The inner dial is a data wheel that lets you specify parameters as you make adjustments in

the menus. The outer dial performs regular shuttle duties with audio, but it also selects the menu you want to edit in data mode. Once I got used to this setup, I found it a convenient way to work.

The R500 also adds a time stamp to your recordings: once you set the date, day of the week, and time, the unit will put this information into the subcode of the tape every time you record. This, of course, gives you the ability to identify a tape that is otherwise unlabled. I only wish that this feature were standard on every DAT machine.

The best news about the Sony studio DAT decks is that they include the company's Super Bit Mapping (SBM) technology. SBM, which is engaged or defeated using a front-panel switch, converts the analog signal to 20 bits; then it reduces the word length to 16 bits when recording to tape, using a proprietary noise-shaping algorithm. You're not using any sort of encoding/decoding scheme, so recordings made with SBM can be played on any other DAT machine without losing the increased resolution. SBM is used only when recording analog sources. It comes standard on the R500, making this unit a great value for the money.

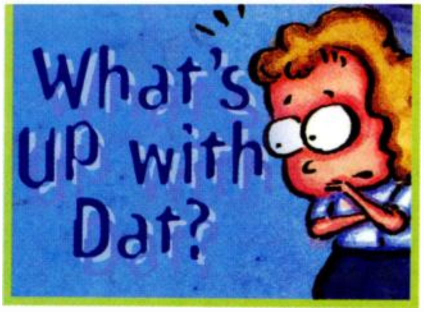
Like the D-5, the PCM-R500 has a handy "wireless off" feature and can play and record all three sampling rates, including 32 kHz Long Play. But the R500 also lets you vary the length of Start IDs; adjust the threshold for automatic ID writing (ranging from −12 to −60 dB, in 1 dB units); and adjust the amount of time that a low-level signal must play (from 1 to 10 seconds in 1-second intervals) before an ID is automatically written.

One thing I miss on the R500 is programmable playback. It has none. Nevertheless, the R500 makes great-sounding recordings, with or without Super Bit Mapping. I left SBM on all the time because I couldn't come up with a reason to leave it off.

FIG. 6: HHB PDR1000

Panasonic SV-3800. To my eyes, the front-panel layout of Panasonic's SV-3800 ($1,695) is the most logically arranged of the bunch (see **Fig. 3**). The features I need most when working are all on the front panel, including the A/D input selector, End Search button, and fade buttons. The shuttle wheel lets you search up to 15 times the normal tape speed—almost twice as fast as the other decks in this survey. It's also the only studio deck that doesn't have built-in rack ears (although the parts are supplied with the unit).

The nominal output of the SV-3800 is incrementally variable over an 11 dB range from +4 dB to −6 dB, then directly to −10 dB. And like the R500, it lets



you monitor an audio input without a tape in the drive. The SV-3800 doesn't, however, read 32 kHz material, in either Standard or Long Play mode.

One of the things that sets the SV-3800 apart from the pack is that it can interface in three digital I/O types: AES/EBU (XLR), S/PDIF coaxial, and S/PDIF optical. To switch among them you have to descend a level or two into the menus. But that's okay, because it also gives you a chance to select *what* kind of digital signal is sent out over each of the digital connectors. For example, on the digital XLR connectors you can select either standard AES/EBU or the consumer-oriented IEC-958 Type II format (which usually comes over optical or coaxial jacks). This feature gives the SV-3800 an even greater degree of digital-dubbing flexibility.

Thankfully, the rear-panel DIP switches that were part of the SV-3700 are gone from the SV-3800, and their functions are now embedded in the menus. This means that you can change the SCMS/ID6 status or digital I/O format without having to go behind the unit to do it.

The next step up in a Panasonic DAT player is the pro-oriented SV-4100 ($2,650). It comes with a buffer memory and can lock to word clock and blackburst.

Tascam DA-45HR. Although Tascam has two studio DAT models that are less expensive (the DA-20 MKII at $975 and the DA-40 at $1,399), I chose to look at the DA-45HR ($2,165) because it can record in both 24 and 16 bits (see **Fig. 4**). The unit records at double-speed to get the 24-bit word length onto tape—which means your maximum record time is one hour in High Resolution mode. In 16-bit Standard Play mode, it operates as you would expect. The DA-45HR, however, lacks an LP mode and it is the only studio DAT player I used that doesn't come with a wireless remote control. You'll need to spring for the RC-D45 remote ($99) if you want to work with this deck from a distance.

In sound quality, the 16-bit recordings made on the DA-45HR surpassed those of other 16-bit units, primarily because the DA-45HR uses a 24-bit A/D converter and rounds down to 16 bits. I was able to improve on the sound even further by using an Apogee PSX-100

That's DAT

	Fostex D-5	HHB PDR1000	Panasonic SV-3800
Price	$1,029	$2,995	$1,695
Analog Inputs	(2) XLR	(2) XLR	(2) XLR
Analog Outputs	(2) XLR	(2) RCA	(2) XLR
Digital I/O	AES/EBU; S/PDIF optical	AES/EBU; S/PDIF coaxial	AES/EBU; S/PDIF coaxial and optical
Sampling Frequency	48, 44.1, 32 kHz	48, 44.1, 32 kHz	48, 44.1 kHz
ADC Resolution	16-bit linear (12-bit nonlinear @ 32 kHz LP mode)	16-bit linear (12-bit nonlinear @ 32 kHz LP mode)	16-bit linear
DAC Resolution	16-bit linear (12-bit nonlinear @ 32 kHz LP mode)	16-bit linear (12-bit nonlinear @ 32 kHz LP mode)	16-bit linear
Dynamic Range	> 90 dB	90 dB	>92 dB
Signal to Noise	> 90 dB	90 dB	>92 dB
THD	<0.05% (1 kHz)	0.015%	<0.03% (1 kHz)
Nominal Input Level	–10/+4	+4	+4
Nominal Output Level	–10/+4	–10	–10 to +4
Frequency Response			
LP (12 bit @ 32 kHz)	20 Hz–14.5 kHz, ± 1 dB	20 Hz–14.5 kHz, ± 3 dB	10 Hz–14.5 kHz, ± 0.5 dB
SP (16 bit @ 44.1 kHz)	20 Hz–20 kHz, ± 1 dB	20 Hz–20 kHz, ± 3dB	10 Hz–20 kHz, ± 0.5 dB
SP (16 bit @ 48 kHz)	20 Hz–20 kHz, ± 1 dB	20 Hz–22 kHz, ± 3dB	10 Hz–22 kHz, ± 0.5 dB
HR (24-bit record)	n/a	n/a	n/a
HR (20-bit playback)	n/a	n/a	n/a
Shuttle Wheel	no	no	yes
Programmed Playback	yes	no	yes
Wireless Remote	yes	no	yes
Mic Preamp	no	yes	no
Dimensions	19" (W) x 5.3 (H)" x 15" (D)	16.93" (W) x 4.8 (H)" x 12.4" (D)	19" (W) x 5.5 (H)" x 14.5" (D)
Weight	15 lbs.	4.3 lbs.	13 lbs.

with UV22. (Designed in part to let you create a DAT backup, the PSX-100 can make 24-bit and 16-bit recordings simultaneously, which was very helpful in this particular reviewing situation.)

In HR mode, the DA-45HR's 24-bit recordings sounded far superior to the 16-bit tracks made with the same recorder. The 16-bit recordings seemed harsher, while the HR tracks had an improved dynamic range and an evenly distributed frequency range. Even with the deck's 20-bit D/A converters, the greater resolution was apparent.

The DA-45HR lets you title each program ID with up to 60 characters for viewing in the display. You create the titles, which are added to the subcode, by using the unit's data/shuttle wheels to choose alphanumeric characters. Because the titles are written into the subcode, they do not affect the audio portion of the tape, no matter which machine you use for playback. The feature also allows you to cut and paste characters from one title to the next, which is convenient when you're doing mixes of the same piece. Note that these titles (as well as 24-bit recordings) can be read only on a DA-45HR; they won't be of any use if you play the tape in another model of DAT player. (For further details on the DA45-HR, see the Quick Pick review in the August 1999 issue of **EM**.)

IN THE FIELD

For many recordists, the ability to make high-quality recordings outside of the studio is one of the biggest draws of DAT technology. For these people, the state of the portable DAT recorder has never been better.

The big difference between the portables and the studio machines is that the portables include mic preamps and sometimes phantom power as well. The mic inputs range from ⅛-inch miniplugs to XLR jacks on the more expensive models. The quality of the preamps and connections keeps getting better, and the durability of the units is also increasing.

One important thing to consider in a portable is its battery time. When you combine mic preamps, phantom power, a headphone amp, and the regular mechanics of recording and playback, you can see why battery life is a major issue with portable units. (See "Gear to Go" in the November 1997 issue of **EM** for more on field-recording accessories.) But the manufacturers are on the case: the most recent portable decks use battery power a lot more wisely than previous models did.

	Sony PCM-R500	Tascam DA-45HR	Sony PCM-M1	Tascam DA-P1
	$1,695	$2,165	$1,000	$2,060
	(2) XLR; (2) RCA	(2) XLR; (2) RCA	stereo minijack	(2) XLR; (2) RCA
	(2) XLR; (2) RCA	(2) XLR; (2) RCA	stereo minijack	(2) RCA
	AES/EBU; S/PDIF coaxial	AES/EBU; S/PDIF coaxial	S/PDIF (requires a special connector)	S/PDIF coaxial
	48, 44.1, 32 kHz	48, 44.1 kHz	48, 44.1, 32 kHz	48, 44.1 kHz
	16-bit linear (12-bit nonlinear @ 32 kHz LP mode)	16-bit linear 24 bit (HR Mode)	16-bit linear (12-bit nonlinear @ 32 kHz LP mode)	16-bit linear
	16-bit linear (12-bit nonlinear @ 32 kHz LP mode)	16-bit linear 20 bit (HR Mode)	16-bit linear (12-bit nonlinear @ 32 kHz LP mode)	16-bit linear
	>90 dB	>100 dB (SP); >105 dB (HR)	>87 dB	>93 dB
	>90 dB	>105 dB (SP); >105 dB (HR)	>87 dB	>92 dB
	<0.05% (1 kHz)	<0.004% (1 kHz)	<0.008% (1 kHz)	<0.004%
	–12 to +4	+4/–10	–10	+4/–10
	–12 to +4	+4/–10	–10	–10
	20 Hz–14.5 kHz, ± 0.5 dB	n/a	20 Hz–14.5 kHz, ± 1 dB	10 Hz–14.5 kHz, ± 0.5 dB
	20 Hz–20 kHz, ± 0.5 dB	20 Hz–20 kHz	20 Hz–20 kHz, ± 1 dB	10 Hz–20 kHz, ± 0.5 dB
	20 Hz–20 kHz, ± 0.5 dB	20 Hz–20 kHz	20 Hz–22 kHz, ± 1 dB	10 Hz–22 kHz, ± 0.5 dB
	n/a	20 Hz–20 kHz	n/a	n/a
	n/a	20 Hz–20 kHz	n/a	n/a
	yes	yes	no	no
	no	yes	no	no
	yes	no	no	no
	no	no	yes	yes
	11.93 " (W) x 7.72 (H)" x 2.2" (D)	9.45" (W) x 3.74" (H) x 6.97" (D)	19 " (W) x 5.75 (H)" x 14" (D)	3.75" (W) x 4.62 (H)" x 1.15" (D)
	15.87 lbs.	18.25 lbs.	.88 lbs.	3.10 lbs.

With these issues in mind, let's look at three popular DAT portables.

Sony PCM-M1. The latest in a line of highly popular, low-cost portable DAT recorders from Sony, the PCM-M1 ($1,000; see **Fig. 5**) includes a number of key improvements over Sony's lowest-priced portable, the D8 ($899). The main differences between the two units is that the M1 takes fewer batteries (two), has a longer playing time using rechargeable batteries supplied with the unit (three hours and 45 minutes), allows you to defeat SCMS, and has improved mic preamps. The PCM-M1, unlike the D8, comes with an AC adapter.

The Walkman-style interface of the M1 helps you get up and running very quickly. The various buttons and switches are easy to understand. However, because they're scattered around the body of the recorder, accessing some of them is difficult when the unit is inside its protective case.

The M1 sounds better and is easier to operate than earlier Sony portables. To further improve the M1's sound and interfacing capabilities, Sony sells the SBM-1 Super Bit Mapping attachment ($550), which not only gives you the increased sound quality of 20-bit A/D converters but also includes a pair of ¼-inch inputs, a pair of RCA inputs, a 3.5 mm stereo miniconnector for line input and headphone output, independent level adjustment for the two channels, and even a 20 dB mic pad. The price of the SBM-1 attachment may seem a little steep at first, but keep in mind that it gives you both increased digital resolution and pro connectors—all in a unit that's roughly the same size as the M1 itself.

Interestingly, the M1 doesn't have a power switch. Also, it wouldn't let me exercise a blank tape before recording (even though the manual suggested doing this). Exercising the tape is an important step: you fast-forward to the end of the tape and then rewind to the beginning. This helps the tape wind evenly throughout and removes any extraneous elements that are clinging to it. Unfortunately, because the M1 wouldn't allow it, I had to use the Sony PCM-R500 for this task.

Tascam DA-P1. Yes, it does cost twice as much as the PCM-M1, but Tascam's DA-P1 ($2,060) is a fully professional unit that could just as easily serve as a studio DAT machine when you're not in the field. The DA-P1 has a pair of balanced XLR mic/line inputs, phantom power, RCA jacks for line-level analog I/O, SCMS-free coaxial digital I/O, and separate level controls for each analog input channel. The unit comes with all of the ID functions you would expect from a pro DAT deck.

The DA-P1 also includes a Hold switch; once you're in Record or Play mode, use Hold to disable the other controls so that you won't accidentally stop the machine or change a setting. You can also illuminate the display, though only for 10-second intervals; that means you will be hitting the display button over and over again when setting levels in the dark.

That point aside, I've used several DA-P1 units over the years for various projects and have never been disappointed by their sound or function. (For more

For manufacturer information, please see Contact Sheet, p. 179.

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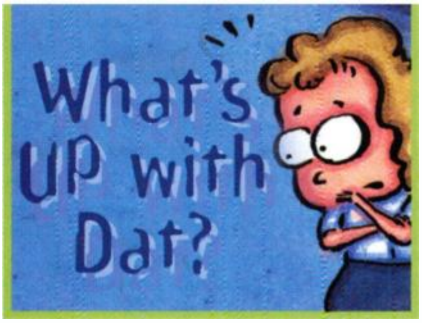

information on the DA-P1, see the review in the July 1997 issue of **EM**.)

HHB PDR1000. The next step up in price takes us to the HHB PDR1000 ($2,995; see **Fig. 6**). The unit I received was actually the fully loaded time-code model known as the PDR1000/TC Plus ($6,995). On its own, however, the PDR1000 comes suited up with a soft case, an RB110 dual-bay battery recharger (which also serves as the AC power supply), and an MHB220 nickel–metal-hydride battery. This particular battery is purported to show fewer signs of "memory effect" trouble than nickel-cadmium rechargeable batteries. To extend the battery life, the recharger has a handy "refresh" feature that completely discharges the battery before recharging it.

What's remarkable about the PDR-1000 is that it's so light (4.3 pounds), even with the addition of the time-code unit. And I really appreciate the design of the carrying case, which allows easy access to the sides of the unit (where connections are made) as well as to the top and front.

The PDR1000 features that I was most thrilled about include a 4-head/4-motor transport, a pair of balanced XLR mic/+4 dB line inputs, a built-in speaker for field monitoring, and the ability to read and record in 32 kHz/LP or 32 kHz/SP modes. Because the PDR1000 has four heads, you can monitor either the source or the tape. The unit can also add a time stamp to the subcode of a tape for identification.

The PDR1000 allows you to lock the input-level controls once you have set them. In addition, a separate Key Hold button disables all of the controls that could potentially disrupt a recording, so you can further secure your player in a field-recording situation.

The mic preamps sound very good and were perhaps a little hotter than those of the other portable units I used. The recording experience was further enhanced by the HM1000 ($350), a 5-position mid-side headphone matrix switch that allowed me to choose among several headphone monitoring settings: stereo, mono left, mono right, M-S stereo, and mono sum. The headphone matrix (which does not come standard but is essential for the serious concert recordist), combined with an AKG C 426 stereo mic, enabled me to dial in the perfect stereo-mic setting, since I could switch between monitoring the stereo spread, the sound of the individual channels, and the overall mono compatibility.

DAT'S ALL, FOLKS

Whether it's a stereo recorder or a mastering or mixdown device you need, DAT is still a strong option, and one that will be with us for some time to come. After 12 years of service, it has shown itself to be a rugged, durable technology with a lot of flexibility. Even within DAT's 16-bit limit, you can now get better results than ever before with the advent of affordable outboard digital converters.

Gino Robair, an associate editor at EM, formally apologizes for all the puns on the DAT acronym. Special thanks to AKG and Whirlwind/US Audio.

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The Ins and Outs of Producing Radio Promos.

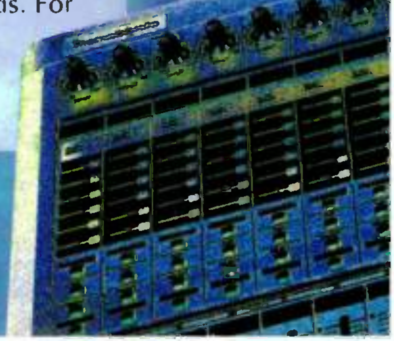
By Roger Maycock

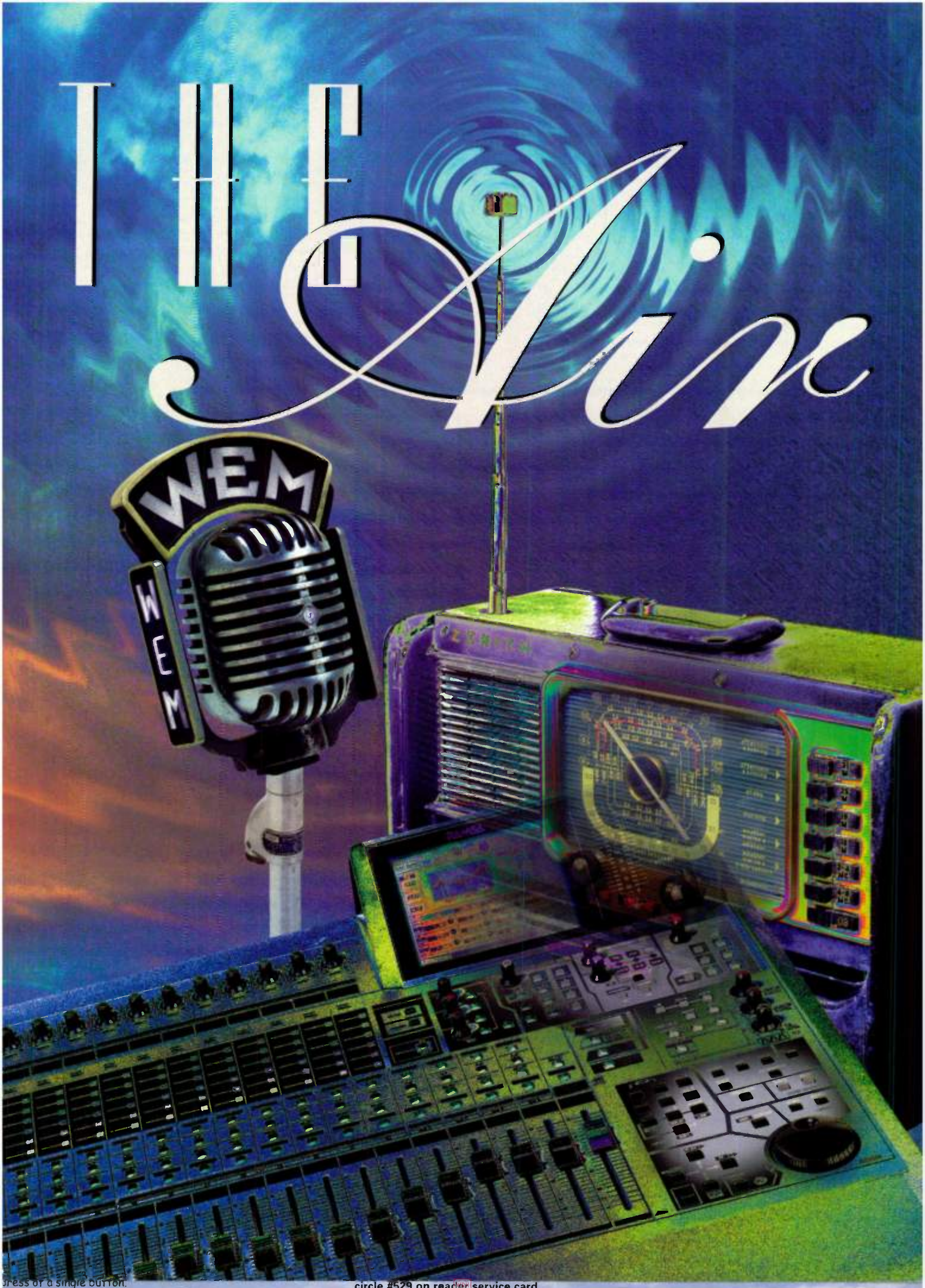
As the tools for music production become increasingly more sophisticated and affordable, the home studio continues to evolve as a facility where we can produce income-generating projects of all types. We now live in a world where record projects, commercials, and music scoring or sound-effects design for TV and film are commonly produced off-site—a practice that was inconceivable ten years ago.

Not to be left behind, the radio-broadcast community has jumped on the bandwagon. Many of the same tools that musicians and recording engineers use for music production are also used in the construction of radio material. The difference with producing material expressly for radio broadcast (such as in-house station IDs, commercial spots, and station promos) is not the equipment but the attitude. These spots are usually relatively short, tightly edited, loaded with snippets of music, and often heavy on sound effects, either to create an aural environment (say, making the dialog sound as though it were recorded on a downtown street corner) or to generate a sonically aggressive promo.

Most important, you rarely have more than a minute to get the message across, so the announcement takes precedence over all else. Ron Shapiro, imaging/creative services director at KCMG Mega 100 FM in Los Angeles, puts it this way: "Brevity is best. If a promo goes beyond 60 seconds, you're likely to lose the listener. Most of the producers I know generally attempt to keep these promos between 30 and 40 seconds. For the most part, people perceive the promo as just another commercial and will tune out if it's too long."

Illustration by Ron Brown





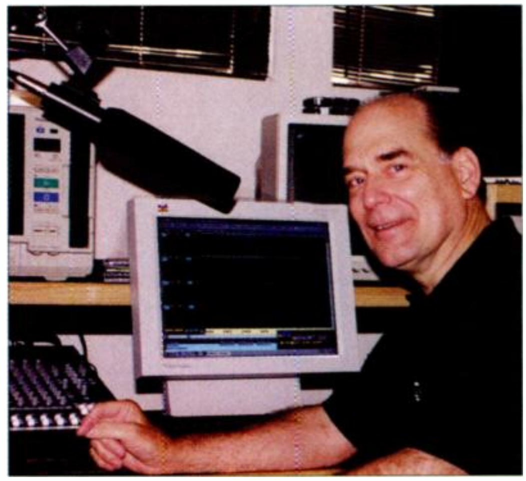
press of a single button.

the number of tracks that are found in a typical project.

For the musician or composer, if you have only eight audio tracks, you probably use a lot of MIDI tracks. For an audio-only project, most musicians prefer to have 16 to 24 audio tracks or more, and it's common to find 96-track production rooms. For radio spots, though, things are very different.

The radio promo usually consists of between eight and ten audio tracks at the multitrack stage of production, and rarely more than a dozen. Why is that? According to Shapiro, "A typical radio promo has two tracks for voices. This frequently takes the form of a primary VO and a secondary voice that responds, or otherwise interacts, with the principal voice. Music and effects usually consist of stereo pairs, each separated for better control over the mix. My multitrack environment frequently consists of primary and secondary VOs on tracks 1 and 2. Music bed number 1 goes on tracks 3 and 4, while music bed number 2 resides on tracks 5 and 6. Ambient crowd noise or sound effects will follow on tracks 7 and 8 and perhaps tracks 9 and 10." **Figure 1** shows this track arrangement.

There are no hard-and-fast rules when it comes to radio production. If you find that you need a certain type of EQ or effect on a particular segment of a track, you will most likely find it easier to accomplish your goal if you isolate that section of material. With some simple cut-and-paste editing, you can move a segment of a particular track onto a dedicated track. In doing so, you immediately have the freedom to process that segment without the fear of affecting the remaining material.



Don Elliot, house voice for Los Angeles–based KFI AM and KOST FM, uses a PC running IQS *Software Audio Workshop (SAW)* as his primary recording and editing tool. His home studio includes JBL 4411 monitor speakers, a Sennheiser 416 microphone, and a Mackie 1604 VLZ compact mixer.

LIBRARY RESOURCES

In the overwhelming majority of instances, radio producers rely on libraries to provide music and sound effects. Why not create your own? Simply put, the answer is lack of time. In order to feed that 24-hour, seven-day pipeline, it's infinitely more practical to use libraries. In the time that it takes you to compose, record, and mix a 30-second music bed, another radio producer will have finished several spots. According to Don Elliot, house voice for KFI AM and KOST FM in Los Angeles, "Radio is a very fast-paced business. The demands of a 24-hour, seven-day system force us to deliver program material in sizable quantities. Music and sound-effects libraries are the only realistic way to meet the demand."

One of the best-known music providers is a company called Brown Bag. It is considered the Rolls Royce of the radio-promo business, and its music is available as CD Red Book audio, so you usually play the music bed into your recording system. (Most multitrack editors don't let you rip audio from CD in the digital domain, and radio engineers rarely use 2-track editing programs, which often have this feature.) Other companies make their libraries available as WAV files, allowing you to import the music data directly into your digital audio workstation. Music and sound-effects libraries are also available on hard disk.

Because the music and sound-effects libraries are professionally mixed and mastered elsewhere, they generally occupy one or two stereo pairs of tracks. If composing is your passion, perhaps you should look into producing music libraries. But if bringing all the elements together and creating something that leaps out of the speaker and drives the point home in 30 seconds would be a welcome challenge, producing radio promos might be for you.

FIG. 1: Here is a good example of how the various tracks of a radio spot are typically organized: VOs on tracks 1 and 2, in addition to two stereo music beds. Note the crossfade in the middle of tracks 5 and 6. This provides a smoother transition from one music segment to another.

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TOOLS OF THE TRADE

Due to its random-access editing capability, the computer has become the principal force in radio production. Although you could do the job on a linear system (such as an MDM), a computer-based system provides far more visual feedback. Which multitrack editing software you use really doesn't matter; the important thing is to have the level of editing capability that is found only in a software editor.

You'll also need a good sound card or other computer audio interface. The important issue here is to select a system that effectively communicates with the other audio hardware you own. (**EM** has covered this subject repeatedly over the years. For a look at PC sound cards, check out "Playing the Slots" in the March 1998 issue. "Digital Pipelines," in the April 1999 issue, offers an overview of digital connectivity, including sound cards. Finally, **EM**'s *Personal Studio Buyer's Guide* features a huge listing of digital audio workstations, including audio interfaces and sound cards.)

You will certainly want the ability to mix digitally to DAT, CD-R, or MiniDisc, so either S/PDIF or AES/EBU digital output capability will be a must. In many cases, a simple 2-channel digital I/O system will serve you well. Because source music and sound effects from libraries are already professionally mixed and mastered, you can play the program material into your system and slide it around to fit your needs later.

Sometimes you will need to sample a sound for use as an effect. In the radio world, the two most popular tools for this purpose are portable DAT and MiniDisc (MD) recorders. MD has experienced less than a wonderful reception in most sectors of the audio community, mostly because it employs ATRAC, a data-compression algorithm that sacrifices a calculated amount of the audio signal in order to record smaller files. However, the format has done quite well in radio broadcast, where the signal will be compressed several times anyway. (I'll discuss this in detail later.)

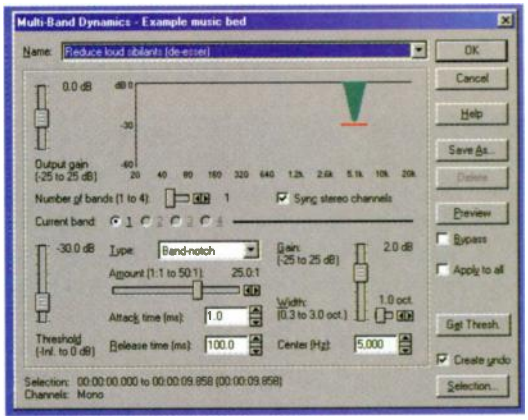

FIG. 2: We're reducing and eliminating sibilant sounds from a dialog track using a compressor and notch filter in Sonic Foundry *Sound Forge*'s Multi-Band Dynamics area.

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When Elliot was asked why he prefers to sample with MD as opposed to tape, he put it this way: "With tape, locating a specific point is far more cumbersome than it is with MD. When I want to check my work to ensure that I've captured the sound I'm after, the rewinding process is time-consuming. With random-access capability, MD provides a faster and easier way of getting things done."

THE VENERABLE VO

It's not uncommon for an announcer to serve as the "voice" of a radio network, giving that chain of radio stations a unique, identifiable sound. To circumvent the physical limitation of having to reside near any particular broadcast facility, many of the better-known announcers have their own home studios equipped with ISDN lines that enable them to "phone in" their part to the production facility. Since we all have to start somewhere, however, we'll continue to focus on getting the job done locally, with local talent.

It's important to recognize that no particular microphone is ideal for every recording job—or for every voice. Choosing a microphone that is best suited for recording voice-overs is largely a matter of personal preference. Of course, every effort should be made to find a microphone that is a good fit for the voice-over talent.

Due mainly to their low cost and durability, dynamic mics are more common than condensers. Most radio stations tend to lean toward the Electro-Voice RE20 and RE27, Sennheiser MD 421, and Shure SM7. However, you'll also find mics like the Neumann U 87 and U 89 and the Sennheiser MD 416 in more elaborate production facilities. That doesn't mean you can't use a mic from, say, AKG, Audio-Technica, or Røde; radio producers who read music-technology magazines sometimes choose these mics as well.

As is the case with every recording application, positioning a microphone requires considerable attention to detail. Close-proximity miking, in particular, requires careful positioning in order to minimize problems caused by sibilants and plosives. Sibilance—a hissing *s* sound—commonly occurs when recording dialog. Plosives are popping sounds caused by words that include the letter *p* and, to a lesser extent, *t* and *d*.

The more you perfect the art of close-proximity dialog recording, the less time you will spend editing the recorded announcement—and as you already know, time is in short supply when it comes to radio production. As is the case with so many aspects of recording, the old saying "Garbage in, garbage out" rings true here. So learning good mic technique is critical.

One of the most common techniques for close-miking dialog is to take a side-angle approach. By positioning the microphone from the side but very close to the speaker's mouth, you can significantly reduce the occurrence of plosive sounds. With this technique, the voice-over talent speaks across the front of the microphone as opposed to speaking directly into it. "If you're experiencing too much plosive and sibilant noise, go for the side-angle approach," recommends Shapiro. "With our station's E-V RE27 N/D, I've found the side approach works best."

If the microphone is positioned at the side, a pop filter might not be necessary. However, if you're miking the talent directly from the front, you'll need to place a pop filter between the speaker and the microphone. Pop filters are available from a number of sources, including Popless Voice Screens, Middle Atlantic Products (Popper Stopper), and Windtech Microphone & Windscreens. You also can build your own; see "DIY: Build the **EM** Pop Filter" in the October 1998 issue.

NAKED DIALOG

Although the finished promo will most likely have music accompaniment, the VO is generally recorded without music cues because the music can become distracting to the talent as they read. It's not uncommon, however, to play the music for the talent beforehand and have them rehearse with the music to get a feel for how the dialog needs to fit.

But as with any creative endeavor, there are no rules. Some VO talent prefer to hear the music in their headphones, so it all boils down to a matter of personal taste. Shapiro notes, "I find that it's really not that crucial to hear the music bed when you're recording a promo. Obviously, if the talent needs to match the tempo of the music, then we'll use it."

How much dialog is used in relation to the rest of the promo spot? Good question. I doubt anyone has ever conducted any research specific to this point, but the general consensus appears to be that the dialog will constitute a good 80 percent or more of the actual promo.

Whatever you do, don't waste time getting the announcement started. More often than not, roughly half a second of music or sound effects will "establish" the promo prior to the actual dialog. Sometimes the dialog even begins right on the music or slightly ahead of it. Remember, the purpose of the promo is to inform, not to entertain—so forget about that four-bar intro.

TROUBLESHOOTING

As mentioned earlier, proper microphone placement and good recording techniques can minimize your editing load, and this is an extremely important point. However, you'll still need to polish even the best of takes. Virtually all professional recording programs offer tools for dealing with plosives, sibilants, and other common artifacts. Learn to use them. Many of these algorithms do a wonderful job of cleaning up dialog. Also, numerous plug-ins for this purpose are available from third-party software developers.

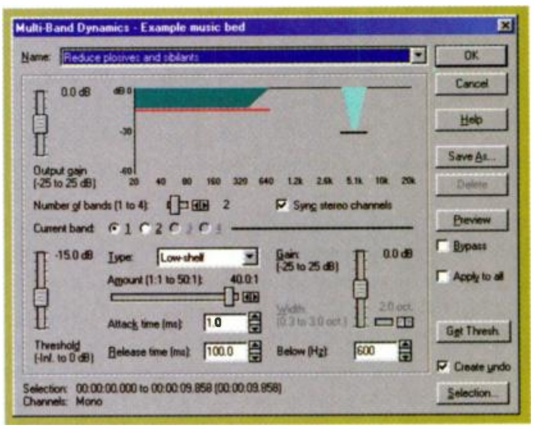
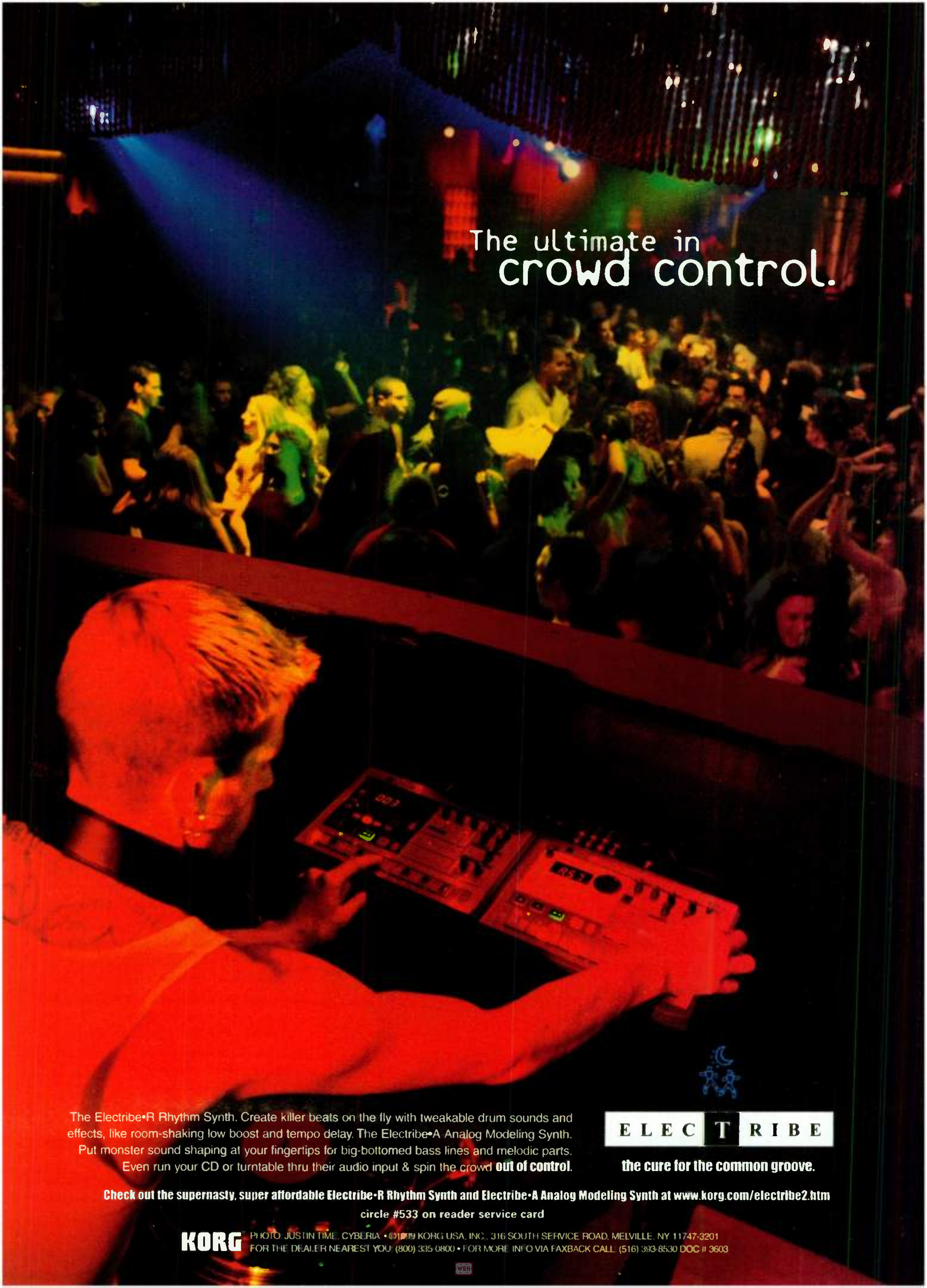


FIG. 3: These parameters can be used to reduce or eliminate plosive as well as sibilant sounds from a dialog track in a single step.



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Use de-essers to reduce the occurrence of sibilant sounds, and use a similar combination of compression and EQ—which can be very effective—for eliminating plosives. These tools commonly fall under the category of dynamics processors and frequently have several parameters for fine-tuning the process. **Figures 2** and **3** illustrate *Sound Forge*'s capabilities in this department. These tools are located in the program's Multi-Band Dynamics area.

In many cases, you'll need to do some more tweaking in one or two stubborn segments, including editing right down to the individual sample. Many programs have provisions for pulling the offending audio segment's waveform down or even redrawing the audio data. And let's not forget what EQ can do. With parametric equalization, you should be able to notch the offending sound into submission.

An important consideration to bear in mind is not to overdo it. Ask yourself, "Just how important is this?" I'm not advocating sloppy work, but try not to lose sight of the fact that the dialog will probably be compressed a good three times or more before the listener even hears it. That ever-so-brief sound that annoys the heck out of you might well be masked by the time music, sound effects, and compression all work their magic. If you overdo things, the dialog can easily become unintelligible—and that can't be allowed to happen.

Experience is a key factor in knowing when to let go. "Sometimes you know what's not likely to be heard once it hits the air, so you simply don't dwell on it," says Elliot. "I know that sounds lazy, and I don't mean to come across that way. The reality of the situation, however, is that there is so much material to produce in order to fill the schedule that if the offending clip is so minute we know it will never be heard, we just let it pass. We always try to deliver the highest-quality product, but there's a limit when so much material needs to be produced."

A CUT ABOVE

The promo's copy is almost always timed prior to the recording session so the engineer knows that it will fit. However, in order to get the message across in a designated time frame, the dialog occasionally needs to move considerably faster than any announcer could realistically speak. Time compression is a truly wonderful aspect of working with software editors. In the days of tape, speeding up the audio playback invariably meant transforming your announcer into a chipmunk. Nowadays, using time compression won't alter the pitch of the program material unless you want it to.

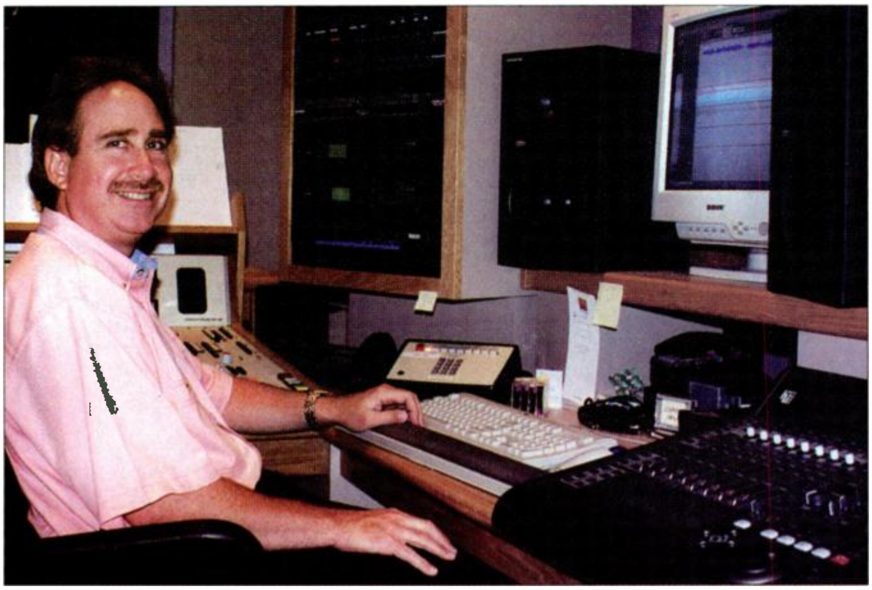
Most programs can handle at least 10 percent time compression without noticeable artifacts. This is usually sufficient because, as noted, most recordings are "timed out" prior to the session. Hence, the original performance and the desired change are usually fairly close—typically not more than about 5 percent apart.

Another common task when working with dialog is to edit out all breath marks. That's right, folks—radio announcers don't breathe! The subtle "aaah" that invariably gets recorded prior to any spoken phrase simply gets deleted. When it comes to removing all those blank spaces in the dialog track, many programs (including Innovative Quality Software's *Software Audio Workshop*, or *SAW;* see **Fig. 4**) enable you to set a threshold level at which any data beneath the specified value will be deleted. This sort of function can be a tremendous time-saver.

Normally, the dialog that follows the deleted breath mark remains where it originally was in time. But if you need to speed things up, here's another tip: rather than deleting the breath mark and leaving the dialog in its original location, you can slip the audio forward and butt one spoken phrase up against the preceding phrase. This alone can shave a considerable amount of time off of any audio segment. With relatively little effort, this technique, combined with time compression, can give the effect of someone speaking a mile a minute.

Another related practice for dialog editing is to position a critical portion of the announcement precisely with a sound effect or other occurrence in the promo. This technique is frequently referred to as "back-timing to music." Perhaps the phrase "You too could win a million dollars!" needs to coincide with a cymbal crash, horn blast, or similar effect. By "slipping" the track into position, or editing out a portion of a pause or breath mark, you can accomplish this easily.

Yet another common technique is to simply duplicate the dialog track and paste it to a new, open track. By offsetting the two tracks ever so slightly, by about 10 or 15 milliseconds,



Ron Shapiro, imaging/creative services director for Mega 100 FM in Los Angeles, creates his radio spots using Digidesign's Pro Tools with a Mackie HUI control surface. Also included in his setup are Mackie's HR824 monitors.

you can easily create a much bigger-sounding voice than you had originally. Even better, pan the tracks hard left and right in the mix for that *really* big sound. Try this yourself, and you'll immediately realize how many times you've heard this technique used.

SQUASHED

Compression and normalization can be effective tools when you're massaging an announcement into shape. However, you must always pay careful attention not to distort the audio. Here again, good recording technique is more than half the battle. A few peaks here and there are natural occurrences, but if the original audio file's waveforms are all over the place and have a lot of widely varying peaks, you may need to normalize the track or segment. Similarly, if your audio levels range from very low to very high, you should also consider normalizing and compressing the track. So what exactly is normalization, as opposed to compression, and how do they interact? Let's take a look.

When you normalize a file, you alter its overall volume so that the highest-level peak in the file reaches a defined limit. The entire signal is adjusted in the same proportion, which does not change the relative amplitudes of the frequencies within the signal. If the highest peak is at the selected limit, the rest of the signal will also be at its highest possible level without *clipping* (distortion) and without affecting the overall sound quality.

Compression, in contrast, is used to fit a large signal into a small dynamic space by squashing the signal's dynamic range so that the peaks are not excessive. But this does not affect the whole signal in a uniform way, as with normalization, so the sound is altered. In radio, where the dynamic range of the audio signal is usually larger than the receiving equipment can handle, compression can effectively fit the audio neatly within the limits of the equipment's capabilities. (For more on compressors, see "Square One: Dynamic Duos, Part 1" in the December 1994 issue of **EM**.)

So how do compression and normalization work in practice? If the original audio file has wide level fluctuations, ideally you would want to rerecord the track. But as this is not always possible, you might consider normalizing the file to achieve a better overall "average"

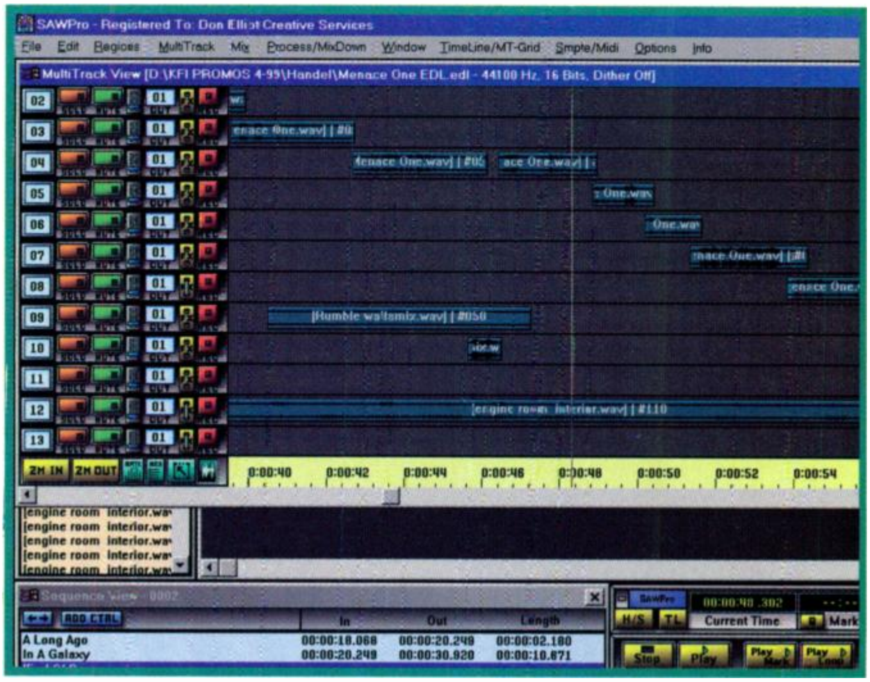


FIG. 4: Innovative Quality Software's *SAW* is a leading Windows program in the radio-production community.

level. Once the level of the VO is more consistent, you can then apply a gentle compression (say, a 3:1 ratio) to tighten up the sound. In the same way that compressing a bass track tends to give more punch to the part, a compressed VO will typically yield a greater degree of articulation, making it easier to understand the words—provided you don't overdo it. In radio production, you typically apply compression a few times, with this stage of production being the first, so don't get carried away.

Normalization can be extremely effective if, for example, you have primary and secondary VO parts in which one voice is considerably louder than the other. By normalizing the two tracks, you can effectively put the two voices on a more equal footing.

Do you always need to normalize the VO? No; if the original recording is clear and doesn't exhibit many peak level fluctuations, don't mess with it. Do you need to use compression? Except with very experienced announcers, the human voice is all over the place in terms of dynamics, so compression is usually a good idea.

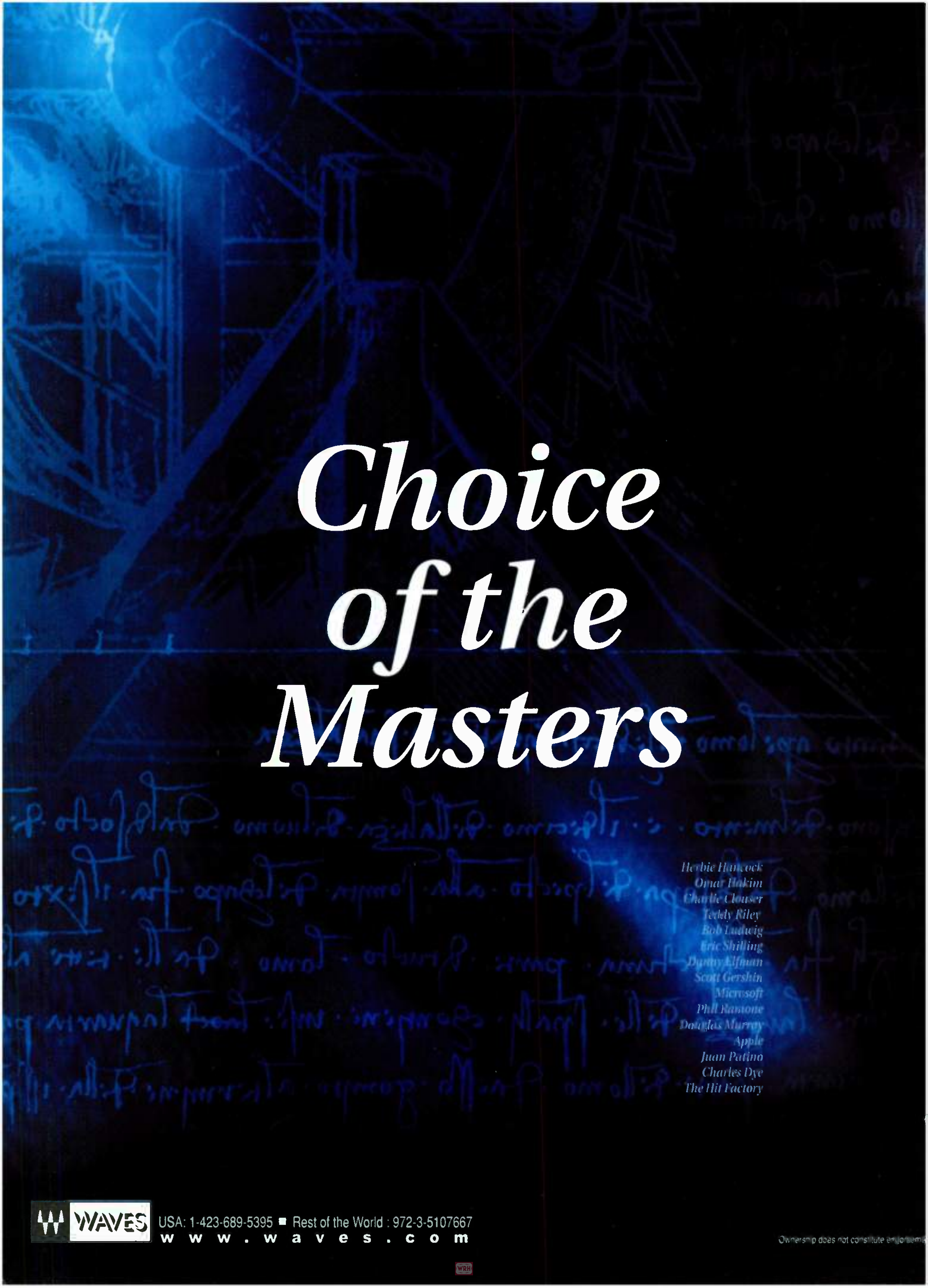
PLAYING MUSIC

Even though most radio facilities use library music, chances are you'll still have to perform some editing on the music bed. For many musicians, perhaps the most annoying aspect of listening to radio or television spots is the proliferation of *stone cut*, or hard, edits that have little if any regard for musical phrasing. I never cease to be amazed at how music gets chopped to fit the time frame of the spot. Audio editors are now beginning to pay closer attention to musical phrasing and, whenever possible, are attempting to soften the blow.

Here's a good tip to make your edits more palatable. If you need to segue from one music style to another, try to make the transition smoother by using your audio program's crossfade function. In **Figure 1**, you'll notice there is a crossfade on tracks 5 and 6. In this case, the music makes a quick, but musically acceptable, transition from a section that contained vocal tracks to one that is purely instrumental. Taking advantage of your editor's crossfade capability can work wonders in this type of situation.

THE MIXDOWN

In the process of creating a mix that translates well to a variety of portable audio systems, home systems, and car systems, it's important to check your mix at low volume levels and at a normal studio level. "My biggest concern," Shapiro says, "is how the mix will sound on a small radio, such as a mono radio alarm clock, as opposed to a big home system.



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I always test my mixes by listening at a very low level to ensure the voice is intelligible over all the music and other background material. If the material passes that test, I know it will sound fine once it hits the air. I'm most concerned with getting the message across."

Because most music and sound effects come from libraries that are already professionally mixed and mastered, the bulk of your focus will continue to be on the VO. When you're equalizing the VO, you will generally find that the majority of your EQ tweaking will occur in the midrange, as this is where the human voice resides.

In order to give some added emphasis to the dialog, you should experiment with the midrange frequencies. As is usually the case, moderation is the key to success. Don't forget that if you boost the high frequencies too much, you're likely to increase the sibilance in the track. If you overdo the low frequencies, you're likely to "muddy" the track and perhaps make plosive sounds more noticeable. Generally speaking, applying a slight boost to the midrange, around 3 or 4 kHz, will bring the announcement straight to the front of your mix.

When you're adding signal processing to the music and sound-effects tracks, your ear must make the final decisions. With explosions and other sound effects, a good, long reverb tail can make all the difference as the sound tapers off into oblivion. With music beds and effects, subtlety tends to be the key factor.

Once the announcement, music, and sound effects reach the point where you feel they are just as you want them, it's time to mix. To expedite your mixdowns, you may find it helpful to place certain types of elements on the tracks in a consistent manner. Get into the habit of placing dialog on tracks 1 and 2 and stereo music beds on tracks 3/4 and 5/6. Sound effects can go on tracks 7 and 8 and, if necessary, on 9 and 10. As Shapiro notes, "Developing a consistent work approach has the benefit of enabling you to work faster with less error. In doing so, you may not even find it necessary to get out your Sharpie to label so relatively few tracks—you just know what goes where."

DUCK, DUCK, GOOSE

When listening to a promo spot, you've probably noticed how the music bed tends to drop down beneath the VO once the announcer begins speaking. This process of lowering the music level is known as *ducking*, and it's one of the most common techniques employed in radio production. These days, the ducking process is generally handled by the editing program's automation system and level changes are generally visible onscreen (see **Fig. 5**). By taking advantage of automation, you can achieve perfectly adjusted levels on a consistent basis. In addition, by writing these level changes into automation, it's easier to edit and fine-tune the mix later.

Typically, you ride the faders that govern the music bed, lowering and raising them as the spot requires. Generally, a change of a few decibels at most is all that's required, though your ears and the person who's paying you will be the final judge. The change in level should be enough to get out of the way of the voice, but not enough to make the music sound like it has moved across the street. If your computer program supports such editing tasks, you may prefer to draw the level changes. Some engineers find that doing so provides them with a greater degree of control.

When you're ducking the music for a promo, knowing whether to make a rapid or a gradual change in level is something that comes with experience. For the most part, your ears will tell you what works and what doesn't. "Sometimes a sudden level change does the job, while other times it sounds too abrupt," says Shapiro. "I generally prefer to make these changes gradually,

FIG. 6: Don Elliot's *Producing Radio Promos Digitally* is an interactive CD-ROM that teaches the art of radio production. For more information, contact him at voiceovers@earthlink.net.

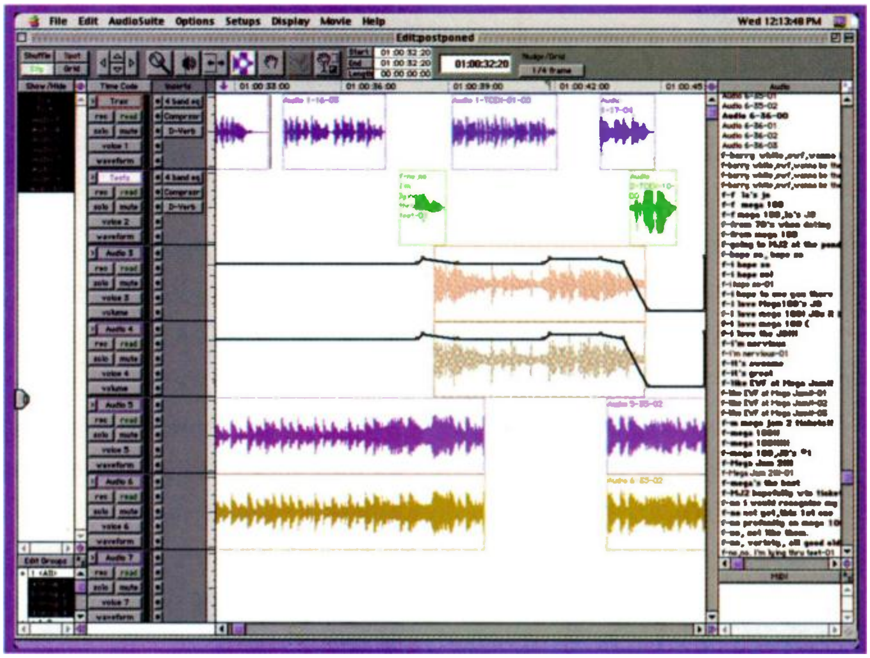

FIG. 5: This screen shot from *Pro Tools* shows a reduction in level for the stereo music bed that is occupying tracks 3 and 4. Being able to inspect changes in level visually can be a tremendous help when you're mixing.

especially when it comes to raising the music bed toward the end of the announcer's dialog. The beauty of using a computer for this task is that you can see where the VO begins and ends, so you can drop your levels and gradually bring them back up to coincide with the end of the announcement."

READY TO PRINT

Generally, the second application of compression will occur during the mix to stereo. For this, TC Electronic's Finalizer, inserted in-line between the workstation and the mastering deck, has proven to be a popular tool in radio production. Shapiro notes, "The Finalizer is a truly wonderful tool for radio-broadcast work. It contains an excellent assortment of presets that, without any editing, are ideal for compressing the mix on its way to the DAT recorder."

As a general rule of thumb, the compression ratio at this stage of production is in the 4:1 range. As before, this is fairly gentle compression, the goal being to even out the entire stereo mix, perhaps making it sound louder. It can't be overemphasized that while compression is common in radio production, you must use moderation, because the station's broadcast technicians will compress the audio signals yet again on the way to the transmitter. Too much compression, and you're likely to end up with something that is unintelligible and without any real dynamics.

While DAT and CD-R are certainly common mixdown media, it should be noted that many random-access systems have provisions that enable you to mix internally to a pair of open tracks. The delivery medium of the final product is not really that important, as long as it is common. Stations will transfer your material to whatever medium they want.

However, it is important to recognize that in many smaller U.S. markets, as well as in Latin America and Asia, you may need to deliver the finished product on MiniDisc. Due to its near-CD sound quality and instant start capability, MiniDisc has had its fair share of success in the broadcast market, partially replacing the conventional NAB (National Association of Broadcasters) cartridge that was used for years. Whatever you do, *never* supply a station with a cartridge. It is a standard in the industry, but alignment is rarely the same from station to station.

At Mega 100 and many other large radio stations, the finished spot will typically be archived to an audio server where all files are numbered and categorized. This is generally accomplished by way of a real-time transfer via a DAT deck or CD player digital output. Once on the server, the promo most often resides as a broadcast WAV file. From there, the audio can easily be recalled at any time for broadcast.

AIR IT

At this point, you should have a fairly good understanding of what you will probably encounter when you land that first radio gig. However, I have a few closing thoughts.

For starters, I can't stress enough just how fast-paced this business is. Since time tends to exist in far too little supply, do yourself a big favor and learn how to use your equipment *before* you commit to a project. If you miss your deadline or in some other way fail to deliver, you will most likely never hear from that employer again. Your first shot will probably be your last if you jump the gun before you are prepared.

Second, listen and learn. Emulation is in many ways the best method of learning a new skill. Listen to radio: know what's being done and, more important, what isn't done. If your work doesn't fit with the station's style, you'll end up wasting more than just your own time and effort.

Many vocational schools for the creative arts as well as colleges and universities have programs to help you get started. If you prefer, try Don Elliot's interactive CD-ROM, which teaches the ins and outs of radio production (see **Fig. 6**).

Radio production is a fascinating field that provides a wealth of challenges and the opportunity to stay steadily employed. So don't just sit there—go for it!

A percussionist, **Roger Maycock** *spent years hitting things before realizing he should be holding sticks. He now sits at a computer attempting to find the Off button.*

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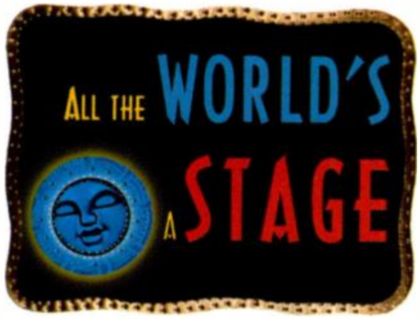
All the World's a Stage

Practical applications for miking world percussion on stage.

By Karen Stackpole

Books about miking techniques typically cover the subject of drum-set miking but rarely focus on percussion and almost never on world percussion. If an ensemble of African, East Indian, or Brazilian hand drummers walked on stage (an increasingly likely occurrence these days), many club sound engineers would have little idea about which microphones to position where. This article covers miking common percussion instruments, as well as many esoteric ones that you may encounter.

I queried three professional sound engineers—all veterans of the San Francisco Bay Area's bustling multi-ethnic music scene—about their approaches to miking percussion. Cuba native Oscar Autié is the engineer of choice for Conjunto Cespedes, Jesús Diaz and Qba, Munequitos de Matanza, John Santos and Omar Sosa, and Talking Drums. Jeff Cressman's credits include sound reinforcement for Keith Terry and Crosspulse, Jai Uttal, and Sheila E. Finally, Greg Landau, who engineered and produced two Grammy-nominated albums for Ritmo Y Candela, works with performers such as Eddy Palmieri, Susanna Baca, Cubanismo, and the Puerto Rican group Plena Libra.



BEGIN THE BEGUINE

When miking percussion, you are usually best off taking a naturalistic approach. Miking in these instances is not about sound design or heavy processing of signals. Rather, the goal is to capture and reproduce the true sound of each instrument and then to fashion a natural-sounding mix in which each voice can be heard clearly among the rest.

Most percussion instruments have a distinctive, identifiable voice. Staying true to this voice—characterized by the instrument's timbre and relative pitch—doesn't mean that you can't equalize or process the sound later; indeed, emphasizing each instrument's identifying characteristics can help clarify and distinguish among multiple percussion voices in the mix. As always, though, it's smart to emphasize the sound first through proper mic selection and positioning.

Begin by listening critically to each instrument to determine its primary tonal characteristics. Listen close up and from a few feet back to find the sweet spot, which is the point or area from which the sound resonates and projects most effectively. The sweet spot is usually where you'll want to position the mic or mics so as to best present the instrument through the speakers.

Of course, many other factors can influence mic selection and placement, and whether you use EQ, compression, or other processing. These wild cards include the size of the room and stage, style of music, number of members in the group versus number of channel inputs, available microphones, and quality of the P.A. system. With so many variables, your ears and judgment are the sole constants. Only by listening carefully to each instrument—and to the overall sound—can you best determine which mics to use and where to position them.

But regardless of the variables, some rules hold true much of the time. First, let's look at different types of microphones and which are most appropriate to use for the stage.

LAY OF THE LAND

The preferred microphones for live-sound reinforcement are usually dynamic mics because they are more rugged, less expensive, and less sensitive than condensers—a quality that frequently translates into less feedback. Live-music venues with their own sound systems typically have a cache of dynamics on hand, such as Shure SM57s and SM58s. The SM57 is an excellent all-around choice for miking live percussion; it can be used on just about any instrument with good results. Another ubiquitous dynamic mic for live sound is the Sennheiser MD 421, which is typically used on toms and other medium- to low-pitched drums.

Of course, numerous other good-quality dynamics are on the market, including many engineered to perform similarly to these two long-standing workhorses. I refer repeatedly to the Shure SM57 and Sennheiser MD 421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All the World's a Stage

in this article, in part because they are the mics most often mentioned by the three engineers I interviewed, and also because they represent two functionally different takes on the dynamic: the small diaphragm and the large. But don't let that lead you to overlook other excellent microphones from Shure and Sennheiser, as well as from companies such as AKG, Audix, Audio-Technica, Beyerdynamic, and Electro-Voice.

In addition, many clubs also own at least one pair of small-diaphragm cardioid condensers, such as Shure SM81s or AKG 451s. Although often positioned as drum overheads, these mics are also suitable for other percussion-miking duties, especially those involving instruments with lots of high-frequency content.

Increasingly common, too, are the clip-on small-diaphragm condenser "micro mics" such as the Shure SM98a, Audio Technica ATM35xcW, and AKG 418. Thanks to their small profile, these microphones can be positioned unobtrusively in tight spots (good for jam-packed percussion arrays) and attached to instruments that get moved around on stage during a performance—for example, djembe, ashiko, and bombo—because they are strapped to the player.

Regardless of what type of microphone you use—dynamic, condenser, or both—make sure that it has a unidirectional polar-pattern response. There are three standard unidirectional patterns: cardioid, supercardioid, and hypercardioid. The cardioid pattern gives you the greatest rejection of sound from the rear of the mic. The supercardioid provides a tighter pattern than the cardioid but is open to some leakage from directly behind the mic. The hypercardioid pattern offers even better side rejection but picks up slightly more sound from the rear than the supercardioid.

Mics designed specifically for drums and percussion—for example, Audix's D and Electro-Voice's N/DYM series—tend to use supercardioid or hypercardioid patterns. Properly positioned, mics with these patterns can provide the best isolation when you're miking densely arranged, multiple sound sources (such as percussion setups).

CLOSER, PLEASE

When you're miking live sound, close-miking is the way to go. It allows greater gain before feedback, and it facilitates good pickup of each instrument and better rejection of unwanted sound.

Mic each instrument individually whenever possible. If you don't have enough channels, you can capture small groups of similar instruments (for example, two conga drums or a cowbell cluster) with a single strategically placed mic. If you have the luxury of extra channels, experiment with miking different parts of the instruments to broaden the scope of the sound.

Be careful of phase problems when you are using two or more microphones close to one another. If you hear phasing, first try repositioning one of the mics, typically moving it

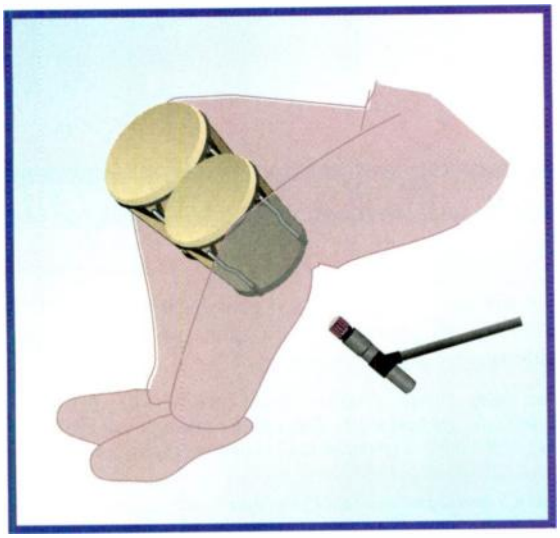

FIG. 1: If the player is sitting, bongos can be effectively miked from underneath, with the mic positioned evenly between the two drums, aimed up into the cavities. This not only picks up more resonance from the bongos, but it also ensures that the microphone does not hinder the musician's arm and hand movements.

away from other mics and closer to the sound source. If that doesn't solve the problem, try reversing the phase of one of the microphones. If neither of these approaches works, stick to a single mic.

In general, low-frequency instruments (for example, tumba, surdo, and bodhrán) are best captured by a microphone with a good low-end response, such as the large-diaphragm Sennheiser MD 421. Higher-pitched instruments (say, bongos and wood blocks) are better complemented by smaller-diaphragm mics such as the Shure SM57 and Audix D2, which can handle sharp transients and high SPLs while still providing a full, natural sound. In addition, the SM57 has a 4 kHz presence peak that helps emphasize attack, making it an appropriate choice for many percussion applications. Most mics designed specifically for drums also have an upper-mid presence peak.

PERCUSS THIS

The world is replete with percussion—after all, practically anything that makes a sound when struck can be construed as a percussion instrument (excepting, perhaps, your kid brother). And even among "officially recognized" percussion the selection is vast, encompassing a wide range of shapes, sizes, materials, and sounds. Obviously, attempting to cover every known percussion instrument would be impractical. Most of them, however, can be categorized by design and tonal characteristics. And once you get the hang of miking a particular type, you can approach other instruments of that ilk in a like manner.

Virtually all percussion instruments fall under one of two categories: *membranophones* and *idiophones*. Anything with a head is generally considered to be a membranophone, and anything without is an idiophone. (A headed tambourine or rattle-outfitted djembe could be said to blur the distinction, but you get the idea.) For the purposes of this article, I have further divided the membranophones into single-headed and double-headed drums. The emphasis is on hand drums, but I've also included timbales. I'll discuss other divisions under each category.

SINGLE-HEADED DRUMS

Single-headed hand drums span an array of sizes and shapes, from cylindrical and conical drums (for example, congas, bongos, and requinto) to goblet-type drums (such as dumbek and djembe) to frame drums (bodhrán, tar, various Native American drums, and so on). They can also produce a wide variety of sounds, from bright, crisp sounds executed near the rims to lower and more resonant tones produced by center hits (not to mention slaps, pops, and finger ruffs, among other actions). I'll start with the hand drums that you're most likely to encounter on stage—bongos and congas—and then group instruments roughly by region.

Common Single-Headed Drums

Bongos. Bongos are typically played with the drums mounted on a stand or supported between the performer's knees. If the player is sitting, Autié suggests miking the bongos from underneath with an SM57, an Audix D1, or Sennheiser MD 421. Position the mic under the chair with the capsule angled up at the bottom of the drums at a distance of three to five inches (see **Fig. 1**).

You can also mic bongos from above, whether the player is standing or sitting. Position the mic between the two drums, facing the heads at a distance of about three to six inches.

Congas. In general, place the mic at the top of the drum opposite the player, about two inches in from the rim and two to six inches from the head. Position the capsule so that it faces downward between 45 and 90 degrees, adjusting the angle to get the sound that you desire. You can boost bass response through proximity effect by moving the capsule closer to the drum head. To pick up more attack, move the mic back a bit and aim the capsule more toward the area where the player's hands hit the head.

Congas are often played in sets of two or three, with each drum pitched differently. From low to high, the drums are called tumba, conga, and quinto. The tumba, as I mentioned previously, is best miked by a large-diaphragm dynamic mic, whereas the conga and quinto are better complemented by small-diaphragm dynamics. To obtain separation between the drums, use one microphone on each drum and angle the mics away from one another (see **Fig. 2**). If the congas are on stands and you have enough extra inputs and mics, you can pick up more low end by positioning a mic beneath each drum, facing up into the cavity. On the other hand, if you are limited to one channel, you can capture a balance of two drums by placing one microphone between them about four to six inches above and angled toward the center area of the heads.

Timbales. Timbales typically consist of two drums, a bell/block cluster, and a cymbal. You can mic this array of instruments in a number of ways. If you're short on channels, you can position a single small-diaphragm condenser mic above the array, angled down, to capture the whole setup. The more common approach, however, is to use three mics: an overhead small-diaphragm condenser positioned to capture the bell/block cluster and cymbal, and a dynamic on each drum. Each dynamic is positioned over the rim of the drum, pointing at the head, as with a rack tom. But with this approach, the penetrating sounds of the bells and blocks sometimes bleed into the dynamic mics, overwhelming the sound and spoiling any separation.

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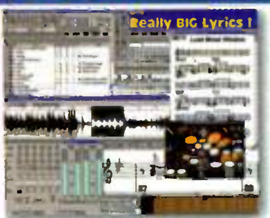

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ALL THE WORLD'S A STAGE

from Brazil, differs from its cousins in that it is stroked, not struck. The player firmly rubs a wooden post attached to the center of the head inside the drum cavity with a cloth to create a unique, voicelike groan that sounds at times like a chimpanzee chattering. Landau recommends miking the cuica from the top at a distance of three inches to a foot to capture the sound coming off the head. (In this case, you won't have to worry about getting in the way of the player's hands.)

Mic the bass cuica from underneath with a large-diaphragm dynamic to better capture the low frequencies. Here, though, you will have to take the player's arm motions into account.

Pandeiro. Also from Brazil, the pandeiro is used in samba and bossa nova. This frame drum comes in different sizes and typically has a single row of dry-sounding jingles, like a tambourine. For a large pandeiro, use an MD 421 to better represent the bass frequencies. An SM57 or similar mic will work fine for smaller pandeiros. Landau recommends placing the mic in front of the pandeiro at a distance that the player can work with—generally, about six inches to a foot. The player can then lean into the mic for emphasis and back away to lower the level.

Requinto, seguidora, buleador. From Puerto Rico comes this family of single-headed, barrel-shaped drums. The requinto is the small drum, the seguidora the medium-size, and the buleador the largest. Mic all three as you would congas.

Tabla. The tabla, used in East Indian music, consists of two drums. The smaller drum is called the tabla; the larger, called the baya, produces the characteristic low-end modulations. A single microphone positioned strategically above and between the drums can be used to pick up the sound of the set. Individual miking, however, better represents the different sound qualities of the small and large drums, allowing separate equalizing as well.

Because the tabla are fairly quiet instruments, getting enough gain before feedback in loud situations is your biggest challenge. I recommend using a mic with a tight pattern—and, again, the SM57 is a good choice.

Kanjira. The kanjira is a small, South Indian tambourine with a thin, lizard-skin head. To accentuate this instrument's crisp highs, mic it from the front with the capsule angled toward the center of the head. The same technique can be used for other small frame drums, such as tamborim.

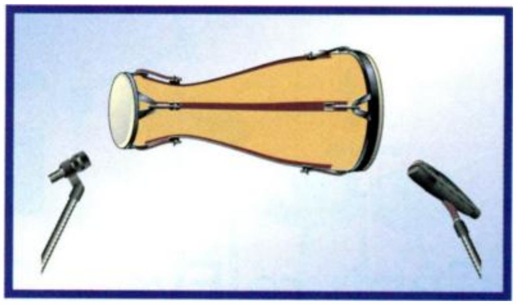

FIG. 5: Most double-headed hand drums require the use of two microphones to capture the full range of tones. To mic an iya, aim a large-diaphragm dynamic between the rim and center of the larger head and a small-diaphragm dynamic toward the rim of the smaller head, with both mics positioned three to six inches from the drum.

DOUBLE-HEADED DRUMS

Double-headed drums project sound differently from single-headed drums. Because they have no open end, you must mic both heads to get a full sound from double-headed drums.

Batá. Batá drums, from Cuba by way of Yoruba (now Nigeria), have a distinctive asymmetrical, hourglass shape. The drums come in three sizes—iya (large), itotele (medium), and okonkolo (small)—and are characteristically played by a group of three musicians sitting together (see **Fig. 4**). For each drum, the larger end is called the enu, and the smaller end the chacha.

Your goal is to represent six distinct voices—two from each drum—so as to highlight the melodic interplay between the three players. Accentuate bass frequencies from the large end of the iya and itotele using a large-diaphragm dynamic like the MD 421 or a mic designed for use on low-frequency sound sources, such as the Audix D4. To represent the high end from the smaller heads use an SM57 or Audix D3 (see **Fig. 5**).

The okonkolo's pitch is on the high side, so SM57s and D3s are appropriate choices for miking both the enu and chacha. Place the enu mic about three to five inches from the head, angled at the edge of the drum, and the chacha mic approximately three to six inches from the head, angled toward the center.

Bombo. Played with a stick and a mallet, the South American bombo produces two distinct sounds: a deep bass

OSCAR AUTIE

FIG. 4: Although they are traditionally played for religious ceremonies, batá drums and rhythms are increasingly seen and heard in secular contexts. Here, AfroCuba de Matanzas performs at the Calvin Simmons Theater in Oakland, California. From left to right, the drums are itotele, iya, and okonkolo.


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from the soft-beater mallet hitting the head, and a "click" sound from the stick on the rim. According to Landau, the bombo can be difficult to mic because the goatskin head has hair on it, which dampens the drum's resonance. In addition, bombo players tend to move around a lot.

Again, the MD 421 is a good choice for capturing the lows, but an MD 504 will work too. Mic the bombo from the top, placing the mic about a foot away to allow for the movement of the musician. Work with the position at this distance to pick up a good balance of both the sound of the stick hitting the rim and the boom from the mallet hits. If more stick sound is required, angle the capsule more toward the rim.

Mrdangam. The mrdangam, commonly heard in the music of South India, has a large head on one end and a smaller head on the other. This drum should be miked like the iya or itotele to capture the lows from the larger head and the highs from the smaller.

Surdo. The Brazilian surdo is a deep-sounding drum with lots of resonance, making it a great candidate for the MD 421. Mic the surdo as you would a floor tom, with the mic capsule situated an inch or two in from the rim, angled down and pointed at the center of the head to capture the most low end.

Talking drums. Talking drums, which are of African origin, are part of a larger family called pressure drums, which are distinguished by an hourglass-shape shell with heads at either end laced evenly together across the length of the drum. Pressure drums are typically held between the body and upper arm and struck with a curved mallet. The pitch is modulated by squeezing the lacing, which increases tension on the heads.

Landau recommends an SM57 for miking talking drums. Aim the mic at the top head to capture the mallet attack and head tone, and leave plenty of room for the player's broad arm movements.

IDIOPHONES

Miking the various idiophones—hand percussion such as shakers, maracas, caxixi, cabasa, guiro, reco reco, cowbells, wood blocks, temple blocks, claves, cua, and tambourines—is fairly straightforward for live performance. Put the mic directly in front of the instrument, positioned to allow a working distance of three inches to a foot. The player, of course, determines the specific distance based on the volume levels that he or she desires. In general, you'll need to bring shakers, maracas, and other quiet instruments in closer, whereas you can capture cowbells and other loud instruments from farther back. As for which mic to use, again, you can't lose with the SM57.

Berimbau (and caxixi). The berimbau, from Brazil, is a wire-strung wooden bow attached to a resonating gourd. The player uses one hand to rhythmically strike the wire with a slim stick while using a flat rock or coin in the other hand to apply pressure to the wire, thus altering the pitch. The musician can also vary the pitch by moving the gourd against his or her midsection.

The caxixi, a basket-type shaker with a loop handle and hard leather base, is typically held in the stick hand to add a rhythmic rattle effect.

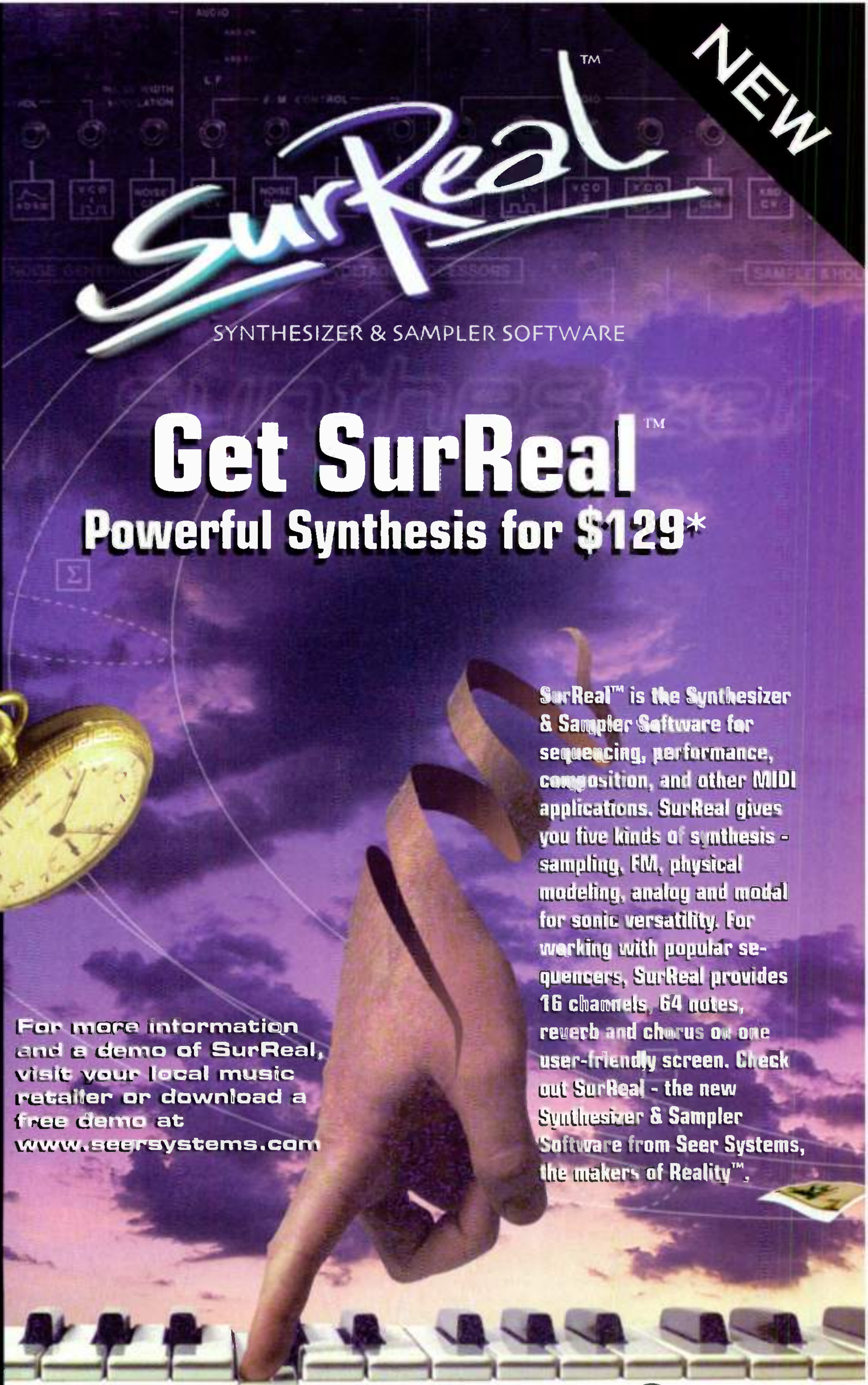
To capture a balance of the various sounds—the metal string being struck, the low resonance of the gourd, and the rattling of the caxixi—Landau recommends placing a small-diaphragm condenser (for example, a Shure SM81) in front of the player above the gourd level. Angle the mic 45 degrees downward, toward the gourd. In this position, the mic will pick up the string sound as well as the sound of the gourd lifting off the player's stomach.

It's also important to capture the full tone of the caxixi and the impact of the beads hitting the leather base inside the basket, so make sure to aim the mic to capture the sound of the basket part of the caxixi.

Cajón. Originally from Peru, the cajón is a square or trapezoid-shaped wooden box with an opening on the bottom or a sound hole in the back. It is played much like a drum, although the larger models typically also serve as a seat in which case the player may also rock the cajón back and forth to vary the tone.

Generally, the cajón is miked from the back at the sound hole (like a bass drum) with a mic that is well suited to accentuate bass frequencies. Autié

For manufacturer information, please see Contact Sheet, p.179.



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Miking Percussion Onstage

This chart provides a quick reference for miking a variety of percussion instruments. The recommendations are not meant to limit your choices, but rather to reflect the preferences of the live-sound engineers interviewed for this article.

Instrument	Recommended Microphones	Basic Placement
Ashiko	Shure SM57, Beta 57 or 56, Audix D2, Electro-Voice N/D408 or N/D468 Clip-on minicondenser mics: Shure SM98a, AKG C 418, Audio-Technica ATM35xcW	2 to 4 inches from head, angled 45 degrees toward rim
Batá: Itotele	Large end: Sennheiser MD 421, Audix D2 Small end: Shure SM57, Audix D3	 Same as for iya
Batá: Iya	Large end: Sennheiser MD 421, Audix D4 Small end: Shure SM57, Audix D3	Large end: 3 to 5 inches from head, aimed between rim and center Small end: 3 to 6 inches from drum, angled toward rim
Batá: Okonkolo	Both ends: Shure SM57, Audix D3	3 inches from drum, aimed toward center
Berimbau/ Caxixi	Neumann KM 184, Shure SM81, AKG C 451 or C 460; Audix SCX-1, Audio-Technica AT3528, Shure SM57	Approximately 1 foot in front of player and 1 foot above gourd, angled down about 45 degrees between gourd and caxixi
Bodhrán	Sennheiser MD 421, Shure SM98a, AKG C 418, Audio-Technica ATM35xcW	Behind drum, aimed between center and rim, 4 to 6 inches from head; or 2 to 6 inches from front head, angled between rim and center
Bombo	Sennheiser MD 421 or MD 504; Audix D4	Approximately 1 foot from top head, angled between rim (at the point where stick strikes) and center
Bongos	Shure SM57, Audix D1, Sennheiser MD 421	3 to 6 inches from instrument, either from above or beneath
Buleador, Requinta, Sequidora	Sennheiser MD 421, Shure SM57, Beta 57 or 56; Audix D2, Electro-Voice N/D408 or N/D468 Clip-on minicondenser mics: Shure SM98a, AKG C 418, Audio-Technica ATM35xcW	2 to 3 inches from head, angled 45 degrees toward rim Optional: MD 421 underneath
Cajón	Top surface: Shure SM57 or Beta 57 Opening (or sound hole): Sennheiser MD 504 or MD 421, Audix D4, AKG D 112 Clip-on minicondenser mics: Shure SM98a, AKG C 418, Audio-Technica ATM35xcW	6 inches from top surface, about 2 to 3 inches in from edge, angled toward player's hands Underneath: place mic just outside opening (as you would mic a bass drum)
Conga	Shure SM57, Beta 57 or 56, Audix D2, Electro-Voice N/D408 or N/D468 Clip-on minicondenser mics: Shure SM98a, AKG C 418, Audio-Technica ATM35xcW	3 inches from head, angled 45 degrees toward rim Optional: MD 421 beneath
Cuica	Top: Shure SM57, Beta 57 or 56, Audix D2 Underneath (helpful for bass cuica): Sennheiser MD 421	Approximately 2 to 5 inches from the head, angled toward center Optional: MD 421 underneath, just outside cavity, pointing up toward head
Cowbell, Cymbal, Woodblock	Shure SM81, AKG C 451 or C 460, Audix SCX-1, Audio-Technica AT3528	3 to 12 inches above instrument(s)
Djembe	Shure SM57, Beta 57 or 56, Audix D2, Electro-Voice N/D408 or N/D468 Clip-on minicondenser mics: Shure SM98a, AKG C 418, Audio-Technica ATM35xcW	2 to 4 inches from head, angled 45 degrees toward rim Optional: MD 421 underneath
Dumbek	Shure SM57, Audix D2	2 to 3 inches from head, angled 45 degrees toward center Optional: MD 421 or Audix D4 underneath, 2 inches from opening
Kanjira	Shure SM57 or Beta 57, Audix D2 or D1	3 to 6 inches in front of drum, angled toward center of head
Mrdangam	Large end: Sennheiser MD 421, Audix D4 Small end: Shure SM57, Audix D3	Large end: 3 to 5 inches from head, aimed between rim and center Small end: 3 to 6 inches from drum, angled toward rim
Pandeiro	Shure SM57, Audix D2, Sennheiser MD 421 (especially for larger pandeiro)	6 to 12 inches in front of instrument
Quinto	Shure SM57, Beta 57 or 56, Audix D1 or D2, Electro-Voice N/D408 or N/D468 Clip-on minicondenser mics: Shure SM98a, AKG C 418, Audio-Technica ATM35xcW	6 inches from head between center and rim, angled 45 degrees Optional: MD 421 underneath
Shekeré	Side: Shure SM57, Audix D1 Mouth: Sennheiser MD 421	3 to 6 inches from side of gourd 3 to 6 inches from mouth of gourd, aimed inside
Surdo	Sennheiser MD 421, Audix D4	2 to 3 inches from head, in 2 inches from rim, angled down a head (as you would mic a floor tom)
Tabla	Shure SM57, Audix D2	2 to 4 inches from head, angled between edge and center
Talking Drum	Shure SM57, Audix D2	3 to 5 inches from head, angled toward center
Timbales	Shure SM57, Audix D3	Top: 2 inches from head, angled 45 degrees Underneath: 2 to 3 inches from head, angled toward shell
Tumba	Sennheiser MD 421, Audix D2, Shure SM57, Electro-Voice N/D408 or N/D468 Clip-on minicondenser mics: Shure SM98a, AKG C 418, Audio-Technica ATM35xcW	2 inches from head, angled 45 degrees toward center Optional: MD 421 underneath, pointing up into drum

ALL THE WORLD'S A STAGE

prefers the Sennheiser MD 504 for this purpose. For smaller cajóns, Landau uses clip-on microphones at the sound hole. In situations where you want more attack (to help the sound cut through the mix), Autié recommends placing an SM57 about six inches from the top surface and angled toward the player's hands.

Shekeré. Originally from Africa, the shekeré is a dried, hollowed-out gourd that is wrapped in a beaded net. This drum projects a combination of high-frequency transients (from the beads slapping against the gourd) and low boom from the mouth of the gourd when the instrument is struck at the base.

Position a Shure SM57, an Audix D2, or a similar microphone three to six inches from the instrument. Again, the musician will determine the distance during the performance. If you have enough inputs available, Autié suggests also using an MD 421, angled downward and aimed into the mouth of the instrument, to better capture the low note. If several shekerés are being played, a melodic conversation between the gourds can be captured with this technique.

TO SQUEEZE OR NOT TO SQUEEZE

Using compression on percussion instruments is a subject of debate among live-sound engineers. Some believe that avoiding it when possible is best, while others use compression prodigiously to control the wide dynamics of an energetic percussion performance.

If you opt to use compression, moderation is the key. Compression increases the risk of feedback and can also alter attack characteristics—an essential element in the sonic nature of percussion. In general, try to maintain as much transient punch from percussion instruments as possible, so as to keep the sound natural and to help the instruments cut through the mix.

Usually, a compression ratio of 2:1 to 4:1 with a moderate threshold setting and gain reduction of about 2 to 4 dB is adequate for situations in which you feel you need compression. Low-frequency instruments such as the surdo and bodhrán may require higher ratio and/or lower threshold settings, to keep the sound from overwhelming the mix.

THE PDQ OF EQ

What you end up doing with EQ depends on a variety of factors, including the acoustics of the space, the proximity of the musicians to the monitors, and how much the sound from the floor monitors is affecting the sound from the mains. Seldom does a venue have ideal acoustics or a premium P.A. system, so a bit of work on your part may be required to bring certain instruments out in the mix. It's hard to generalize because each situation is different. As a rule, though, always try to enhance the sound first by working with mic placement rather than equalization. Only after you have achieved the best sound possible from mic selection and positioning should you reach for the EQ.

When using EQ, make sure you are clarifying the mix rather than muddying it. Listen carefully to the results while the full ensemble is playing and adjust the various channels accordingly. Hand drums can typically benefit from a little boosting in the 2 to 5 kHz frequency range, to bring out the "slap" and "pop." To reduce muddiness, try cutting a couple of decibels around 250 to 350 Hz. If an instrument sounds too bassy, roll off the lows below 100 Hz with a high-pass filter. (You can also back the mic off a bit to lessen the proximity effect.)

EXIT STAGE RIGHT

Many percussion instruments—taiko drums and gamelan, for example—are obviously not covered in this article. Hopefully, though, you'll be able to extrapolate from the techniques detailed above to prepare yourself for miking almost any percussion instrument that comes your way.

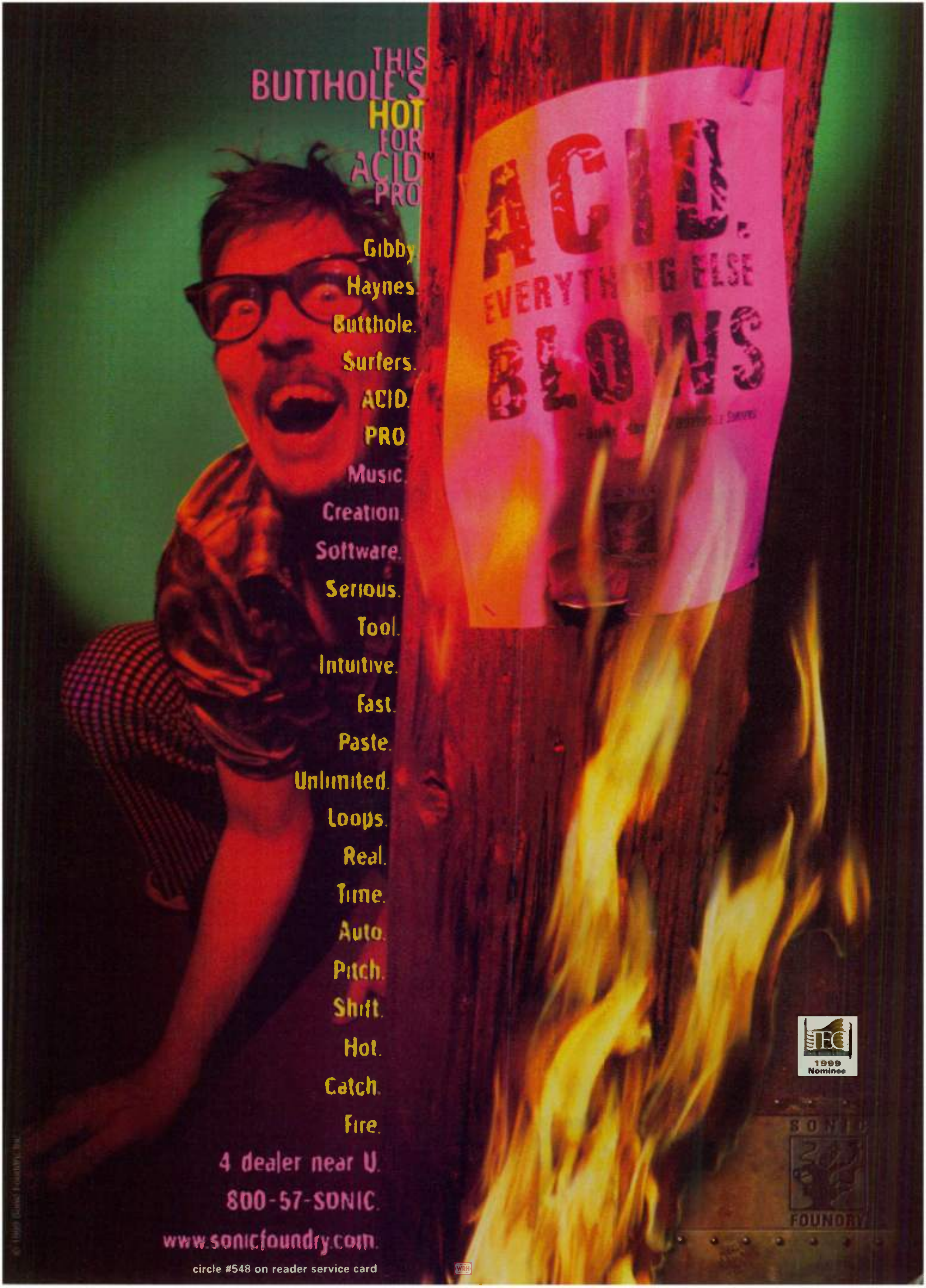
Always remember to use unidirectional mics, selecting the particular polar pattern according to your needs for off-axis rejection. Dynamic mics are usually best for the job because of their durability, ready availability, and decreased sensitivity (as compared with condensers). On the other hand, small-diaphragm condensers are often preferable for use on cymbals, bells, and blocks. You can also use clip-on condensers on a variety of drums to minimize clutter and allow the musicians to move freely around the stage.

Finally, always listen critically so as to select the best mics for the job, and position them to capture the most natural sound from the instrument without inhibiting the player's range of motion.

OSCAR AUTIE

Dynamic mics are commonly used to capture the powerful tones of Latin hand percussion. Pictured are members of the Puerto Rican group Los Hermanos Cepeda playing (left to right) the requinto, seguidora, buleador, and cua.

Karen Stackpole *is a recording and mastering engineer and an active drummer and percussionist. Many thanks to Oscar Autié, Jeff Cressman, and Greg Landau for their valuable assistance.*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Comping a Vocal Track

Copy and paste your way to the perfect vocal track.

By Scott R. Garrigus

No matter what kind of songwriting you do, the vocal tracks are typically the most important element for connecting with the listener and conveying emotional content. It's therefore crucial that you produce the best possible vocal tracks, even when the singer doesn't deliver that elusive "perfect" take.

The easiest way to get a great vocal recording is to create a *composite* track. In other words, you record multiple takes—each on its own track—and then combine the best parts of each take into a single "perfect" performance.

Using multitrack analog tape for this process has always been a hassle. Bouncing the takes by muting bad sections and soloing good sections can be time-consuming and inaccurate. Mix automation helps, but you still have the problem of generation loss, the deterioration in audio quality that happens each time you rerecord an analog track. Digital tape recording has made the process quite a bit less painful. With the ADAT's seamless, automated punch-in and punch-out recording, edits are more precise and digital recordings don't suffer from generation loss.

Hard-disk recording, however, offers the greatest flexibility and makes comping even easier because you can visually edit audio waveforms. In some cases, you can edit right down to the individual sample level. Even so, comping a vocal track with digital audio editing software is not just a matter of cutting and pasting. You have a number of options during recording and editing that can make the difference between a professional track and a patchwork one.

RECORDING TIPS

During the recording session, your object, of course, is to make each take sound as good as possible. But you also want them to sound as much alike as

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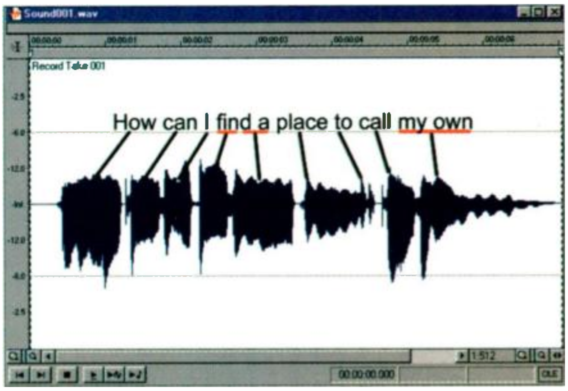


FIG. 1: In the phrase "How can I find a place to call my own?" the words "find a" and "my own" blend together, so there's no way to do a clean edit between them. The best place to edit is just before a hard consonant. In this example, the best edit points are right before the words "can," "to," and "call."

possible. If all the takes sound similar, later edits will blend seamlessly.

To begin with, don't use time-based effects (such as reverb, chorus, or echo) during the recording. When you start pasting in sections from different tracks, things like reverb tails and LFO modulation for chorusing won't match up correctly. Dry tracks provide a more uniform sound. On the other hand, if you add effects after editing, they can help mask the edits between audio segments.

Depending on your singer's mic technique, you may want to use a bit of compression while recording each take. That will give each take a similar amplitude range. A modest setting with a fairly high threshold, fast attack, and 2:1 or 3:1 ratio should keep levels under control without squashing the life out of the recording.

In case your singer gets a little too rambunctious with those high notes, it's a good idea to patch a limiter into the recording chain to prevent signal overloading. Set up the limiter with a fast attack and release and a threshold that's a couple of decibels below the loudest peak your sound card can handle. With these settings, the limiter will simply block out distortion without otherwise altering the signal.

Noise can sometimes be a problem during recording. Even though there are methods for eliminating some noises after they occur (which I'll discuss later), prevention is always the best way to handle the problem. For example, always use a pop filter on your mic to eliminate *plosives* (sudden bursts of air) that your singer may produce.

Furthermore, if your mic has a low-cut switch, you might want to activate it. The human voice has very little low-frequency content, and you don't want your mic picking up things like ventilation noise or room rumble. You might also consider using a noise gate. They can produce abrupt level changes, but if you set a very low threshold and a decay time of about 250 to 400 ms, the changes won't be so obvious. Moreover, the reverb you'll add to the vocal track later should smooth out the transitions.

EDITING BASICS

Composite tracks are typically created using one of three methods: cutting; copying and pasting; or using amplitude envelopes. Cutting involves destructively removing all of the bad sections from each take. You are then left with a number of tracks that contain scattered audio segments, which make up the final composite take. Cutting is easy, and it's available in any editing program, but it's not very flexible because you can't go back and make changes.

Instead of seeking out the bad sections, the copy-and-paste method involves finding and marking all of the good sections in each take. You then copy the best sections and paste them into a composite track. This method is easy, and it's flexible because it doesn't alter your source tracks.

But using amplitude envelopes is the best method by far. With this technique, all of your edits are fully adjustable at all times, and there's no need for cutting or copying and pasting. You simply assign amplitude envelopes to each take, draw in the volume changes needed to mute the bad sections, and then mix down to a new composite track. If you decide that you need to make changes, you simply adjust the envelope points and remix. Unfortunately, amplitude envelopes aren't available in all audio-editing programs, so you might have to use one of the other methods.

Whichever method you choose, always be sure to use the "snap to zero crossing" function in your editing program. By ensuring that all of your edit points land on the nearest zero-point crossing in the audio waveform, this function minimizes glitches that can occur between segments. To map out the good (or bad) sections of each take, use start and end markers. Then you can easily select the audio between each set of markers for cutting or copying and pasting. When using amplitude envelopes, however, you don't need markers because drawing the envelopes visually marks each section.

WHERE TO CUT

Where you make your edits is just as important as how you make them. Finding the right points on a waveform display can be difficult unless they all occur in the silence between lyrical phrases. Within a phrase, however, many of the words flow into one another. For example, in the phrase "How can I find a place to call my own?" the words "find a" and "my own" blend together, so there's really no way to do a clean edit between them (see **Fig. 1**).

The best place to edit is immediately before a hard consonant. The quick burst of air used to pronounce hard consonants separates them from the word before, so the edit will work even if there isn't a pause between the

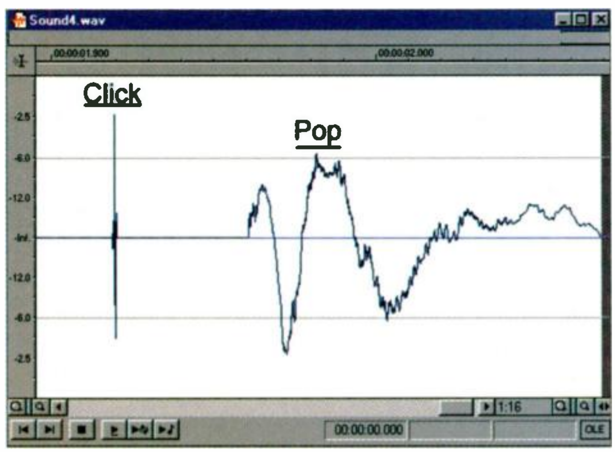


FIG. 2: In a waveform fully zoomed out, pops and clicks have a similar "spike" appearance. When zoomed in, however, clicks retain the spike look, whereas pops resemble distorted waveforms.

words. In our example, the best edit points are right before the words "can," "to," and "call." Hard consonants are relatively easy to find on a waveform display because of their sharp peaks.

Another useful, but more difficult, technique is to place your edit points on *fricatives,* consonants pronounced by forcing breath through a constricted mouth formation. The letters *f* and *s* are fricatives. Because of the noise they produce, it's possible to hide an edit point within them. You'll be editing within the word itself, so you should use this technique only when absolutely necessary. Smoothly matching up audio segments this way is often difficult, especially when the words are being sung.

KILL THE NOISE

Even if you take preventative measures, noise can make its way into your tracks. But a few clicks and pops shouldn't deter you from using an otherwise great section of audio. These transient disruptions can easily be removed, and you don't need any special tools to do it. Most of the pops your singer produces while performing plosive lyrics can be filtered out by a pop filter. Clicks, however, arise from a variety of sources, including saliva snapping in the singer's mouth.

In a waveform fully zoomed out, pops and clicks have a similar "spike" look. It's best to locate them by first listening to the audio segment and then zooming in on the suspected area. When zoomed in, a click still looks like a spike, but a pop resembles a distorted waveform (see **Fig. 2**).

You don't want to simply delete these transient noises. That could upset the rhythm of the lyrics. Instead, reduce the volume of the pop or click by about 10 to 15 dB. You may also want to try replacing the click or pop with a tiny bit of the waveform that comes immediately before or after it. You'll have to experiment a little to see what sounds best. You also have to be cautious when making a selection. Clicks are easy to select, but pops usually come right before the start of a word. Take care not to cut any part of the word itself.

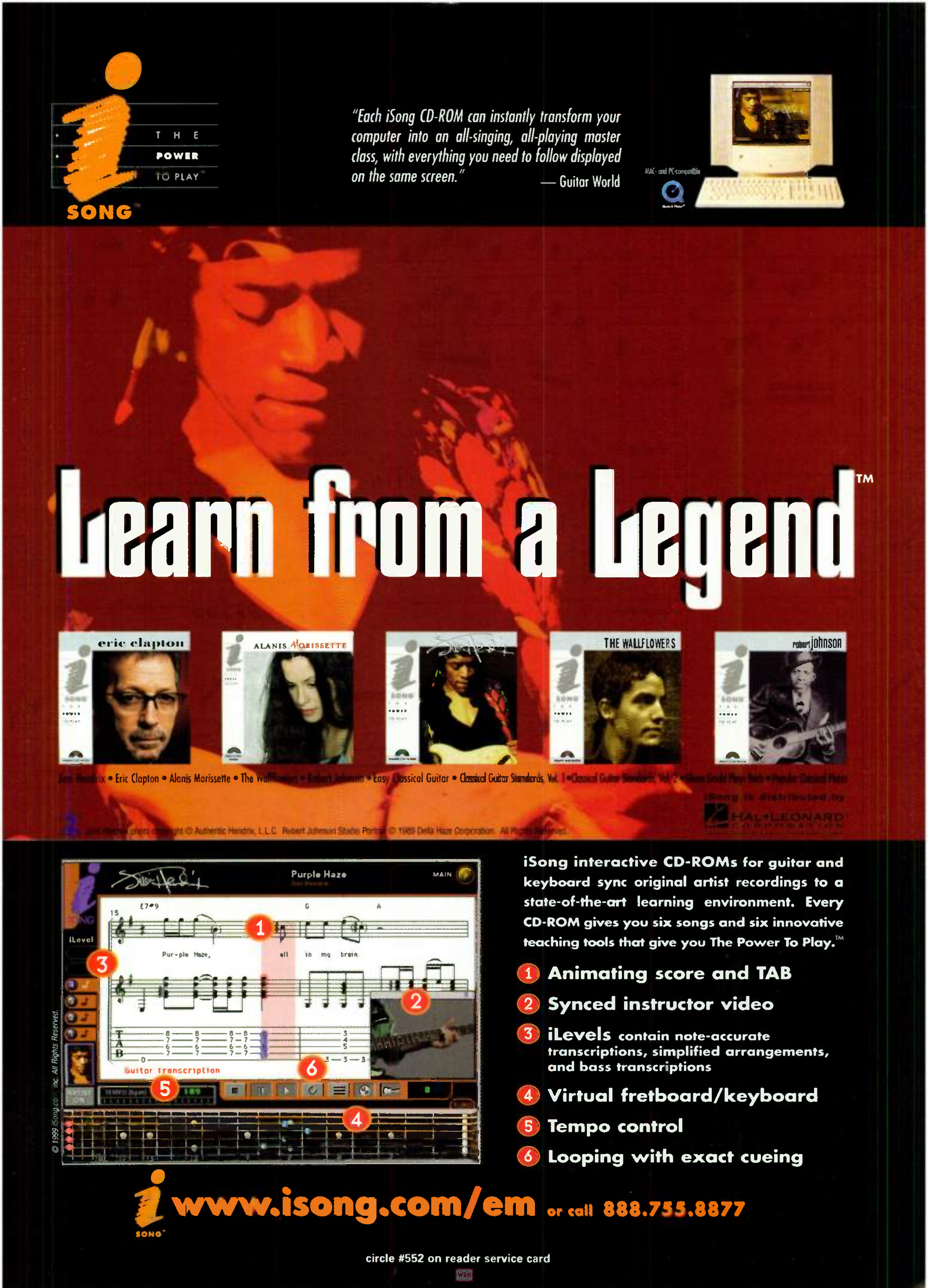
Always use a pop filter on your mic to eliminate sudden bursts of air.

FINAL CUT

If vocals are a big part of your music, making them sound as good as possible is a high priority. Creating a composite vocal track is one of the best ways to do that. It may take a little time for you to get the knack of making clean hard-consonant edits or recognizing and removing pops and clicks, but it's definitely worth the effort.

There may not be such a thing as a perfect vocal recording. There always seems to be something that could be done just a little bit better. But if you take the time to apply the techniques outlined here, your vocal tracks can step a bit closer to perfection.

Scott R. Garrigus *is an author, musician, multimedia expert, singer, and voice-over performer. You can contact him at www.garrigus.com or via e-mail at scott@garrigus.com.*



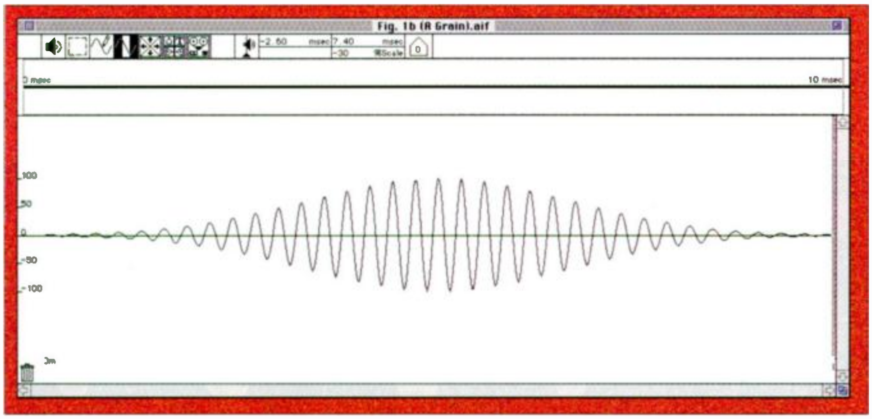


FIG. 1: This figure shows a 10 ms sine-wave grain with a Gaussian envelope.

Roads has also published numerous landmark articles on the subject.

Since Roads's first work, many others have contributed to the art. Especially notable is composer Barry Truax, who pioneered real-time granular synthesis in the 1980s. To accomplish this, Truax had to microprogram a DSP device and write additional software for a host computer. Today, a variety of granular synthesis software is available to any desktop musician, with little or no programming involved. (See the sidebar "Granular Synthesis Tools.")

Granular synthesis, unlike subtractive or modulation synthesis, doesn't derive from conventional signal processing techniques and has no standard implementation. Curtis Roads's technical vocabulary is the closest thing to an accepted terminology for granular synthesis, and I'll use it in this article.

STRUCTURE OF A GRAIN

Let's start by looking at how a single grain is generated. It's a fairly trivial operation. The classic recipe calls for an oscillator signal with a simple amplitude envelope. An oscillator-based grain of this type is termed *synthetic*. Any waveform can be used in a synthetic grain, but a sine wave is probably the most common. Another approach, called *sound-file granulation*, uses audio samples as the source signal. Sample-based grains are called *file grains*.

For a grain's amplitude envelope, a smooth curve is often desirable because it doesn't produce sharp transients in the grain. The most common grain envelope is the bell-shaped curve of a *Gaussian* function. Shapes that approximate this curve, such as triangular

GRANULAR SYNTHESIS TOOLS

The good news is that granular synthesis software abounds on both the Mac and PC platforms. The better news is that much of it is free or very inexpensive.

If you're a Macintosh user, you'll definitely want to get ahold of Chris Rolfe's award-winning *MacPOD* (www3.bc.sympatico.ca/thirdmonk), which is a real-time, MIDI-controllable file-granulation program. *MetaSynth*, by U&I Software, offers granular synthesis–based effects in its Effects palette. A demo is available at www.uisoftware.com. *ThOnk_0+2* (www.audioease.com) is a file-granulation program by Arjen van der Schoot, who boasts that composers can use the free program "without having to think at all." Just provide an input file, and *thOnk_0+2* will generate reams of material. Despite its name, Mike Berry's *GrainWave* 3 (www.nmol.com/users/mikeb) does much more than granular synthesis. It's a real-time software synthesizer with a grain operator that generates both synthetic and file grains. James McCartney's *SuperCollider* (www.audiosynth.com) also supports real-time granulation; some coding is required. Curtis Roads's classic *Cloud Generator*, which is not a real-time application, is available through Tom Erbe's Mac computer-music Web site (shoko.calarts.edu/~tre/CompMusMac).

The Windows platform also has some excellent granular synthesis programs. Rasmus Ekman's real-time *Granulab* (hem.passagen.se/rasmuse/Granny.htm) handles both file and synthetic granulation and includes refinements such as patch storage, MIDI control, and mouse-controlled crossfades between patches. The full registered version can generate up to eight grain streams at once. Ross Bencina's *AudioMulch* (www.audiomulch.com) is a full-featured, real-time software synthesizer with a granulator that takes its input from a delay line. *Chaosynth* (www.nyrsound.com) does real-time granular synthesis, filtering, and reverb with grains generated by cellular automata. It will also produce a MIDI file from the granular data. (A new version is NT-compatible.) Finally, the U.K.-based CDP (www.bath.ac.uk/~masjpf/CDP/CDP.htm) has recently added *GrainMill*, a file- and synthetic-granulation program, to its collection of inexpensive synthesis programs. CDP's *Groucho* signal-processing package also contains a set of grain-manipulation tools.

The Csound language (Mac, Windows, Linux, BeOS, other platforms) includes several granular unit generators. Csound is available from http://mitpress.mit.edu/e-books/csound/frontpage.html. You might start with the relatively simple *grain* generator, then graduate to the more complex *granule*, which accepts no fewer than 22 parameters! Tom Erbe's *CornBucket* utility takes another approach: it generates a Csound score file consisting of zillions of grain events. Get it at Erbe's site (see above).

One final cross-platform option is the Kyma System from Symbolic Sound. This hardware-accelerated synthesis workstation has so many built-in tools for granular synthesis that it definitely deserves a look. You'll find numerous presets ready to splice and dice any sound you throw at them, plus built-in granular sound generators that can employ any basic waveform you want. Kyma can do granular synthesis in real time, with all the parameters live and tweakable. Have a look at www.symbolicsound.com/kyma.html for an introduction to the system.

or simple three-stage envelopes, are also commonly used.

Figure 1 shows a typical sine-wave grain with a Gaussian envelope. Because of the grain's short duration, the ear registers it as a "blip" of ill-defined pitch and timbre. Taken by itself, it's a rather boring sound. However, the objective in granular synthesis is not to produce grains that are interesting individually, but to sequence these neutral-sounding blips into larger events that will interest the listener. In this context, the anonymous character of the grain is actually an asset.

Grain parameters can vary on a grain-by-grain basis. You can vary many aspects of the grain, including the waveform or sample source, and its amplitude, frequency, duration, envelope shape, and pan position. The effect of changing some of these parameters will be obvious. For example, if you synthesize 100 grains over 2 seconds and increase the frequency of each grain by a small amount, you will get some sort of pitch-sliding effect. Other parameters, such as duration and envelope shape, have effects that cannot be fully understood until we consider what happens when we arrange grains in time.

SEQUENCED GRAINS AND AM

Amplitude modulation (AM) is an important component of granular synthesis. If you're not familiar with the basics of AM synthesis, you may want to look over a previous "Square One" article on that topic, which appeared in the March 1999 **EM.**

The easiest way to understand the relationship between granular synthesis and AM is to consider a stream of evenly spaced sine-wave grains, like that in the top track of **Figure 2.** This simple example, where the time intervals between grains are equal, is called *synchronous granular synthesis* (SGS). If the rate of grain generation, or *grain density,* is lower than about 20 grains per second, the grains will be perceived as a metronomic sequence of sounds. If the grain density is higher, listeners will hear a continuous signal.

SGS is equivalent to amplitude modulation of a carrier signal by a periodic modulator. The carrier, in this example, is a sine wave, and the modulator is the repeated envelopes. To figure out the modulator frequency, take the inverse of the period between one grain and the next. For example, given a period of 1/30 of a second, the fundamental modulator frequency would be 30 Hz. As in conventional AM, sidebands (sums and differences of the carrier and modulator frequencies) appear in the output spectrum.

The shape and duration of the envelopes determine the modulator waveform and therefore its spectrum. In general, envelopes with sharp rise and fall times will generate stronger high-frequency content than smooth envelopes. Thus the choice of envelope can have a considerable effect on the number and strength of the sidebands.

You can approximate granular synthesis with your favorite sequencer and synthesizer.

SGS isn't the most typical application of granular synthesis; it's a rather cumbersome way to produce AM effects. (To get ordinary AM, use ordinary signal generators.) However, SGS does let you vary one or more parameters per grain, leading to complexities difficult to produce with AM alone. In the bottom track of **Figure 2,** the frequency of each grain is random within a 50 Hz to 2 kHz range. We could analyze this as amplitude modulation of a carrier that is also being frequency modulated by a quasi-random step waveform.

SONIC SPRAY GUN

Asynchronous granular synthesis (AGS), in which grains are distributed randomly over some period of time, is a more widely used form of granular synthesis. AGS works like a sonic spray gun, scattering droplets of sound instead of paint. By definition, AGS involves some degree of aperiodicity in the timing of the grains. The effect is often similar to AM with band-limited noise as the modulator signal; there is random sideband energy, which is perceived as noise.

The term *cloud* is particularly appropriate for AGS-generated events. Like natural clouds, an AGS cloud is made up of randomly configured particles. Rather than arranging all details of the grains that make up the cloud, you specify the duration of the cloud and the way various grain parameters will change over that period of time. Grain density, frequency, amplitude, panning or multichannel dispersion, waveform, and envelope shape may all be variable over the life of a cloud, depending on what software you use. Non-real-time programs like *Cloud Generator* require you to give initial and final values for the parameters you want to change, before the software generates the grains. Real-time programs, like Chris Rolfe's *MacPOD* and Rasmus Ekman's *Granulab,* let you conduct the cloud-generating process as you listen.

The frequency content of a cloud determines whether it will have a strong pitch center. Clouds in which all grain frequencies are the same, or fall within a narrow range, tend to have a perceptible pitch, somewhat like bandpass-filtered noise. If grain frequencies are dispersed over a wide range, the cloud will sound more like broadband noise. Low and high boundaries for grain frequency may be specified in terms of bandwidth or as a

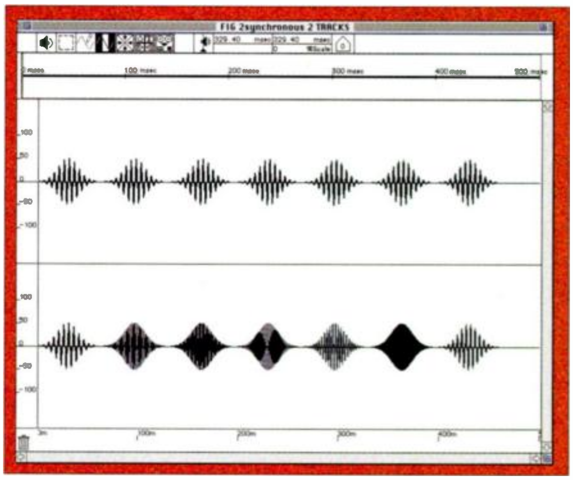


FIG. 2: These two tracks are examples of synchronous granular synthesis (SGS). The top track shows constant grain frequencies, and the bottom track shows grain frequencies randomized.

frequency deviation from a center value.

Figure 3 illustrates a simple 1-second cloud in which the grain density increases while the grain amplitude decreases. For this cloud, the initial and final values for grain density were 30 and 60 grains per second. The initial and final amplitude values were 60 percent full-scale and 0. Frequency and all other parameters were held constant. This cloud sounds like a series of rattling pitched-noise sounds, gradually accelerating while diminishing in volume. The rate of acceleration and the volume fade are random. This unpredictability imparts a certain liveness to the sound. Note that some of the grains in **Figure 3** overlap, which is common in AGS and can easily lead to clipping. To avoid clipping, you may find it necessary to lower the grain amplitude range.

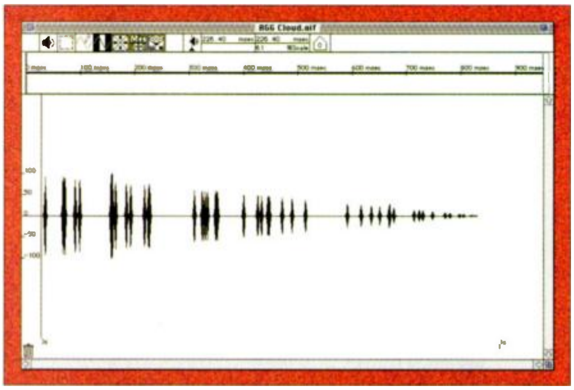


FIG. 3: This is a 1-second example of asynchronous granular synthesis (AGS). As the grain density increases, the amplitude decreases.

OTHER RESOURCES

If you want to find out more about granular synthesis, the first place to look is the writings of Curtis Roads; see the discussion and bibliography in his *Computer Music Tutorial* (MIT Press, 1996). Also look at the excellent and practical *Cloud Generator* manual. *Representations of Musical Signals* (MIT Press, 1991) also has a chapter on granular synthesis, though much of it is highly technical. Barry Truax describes his mid-1980s work programming for real-time granular synthesis in *Computer Music Journal* 12, no. 2 (summer 1988). Also worth noting is Michael Berry's in-depth description of his *GrainWave* program, published in *Computer Music Journal* 23, no. 1 (spring 1999).

Commercial recordings featuring granular synthesis are not widely available. *Riverrun*, by Barry Truax, is included on his *Digital Soundscapes* CD (Wergo 2017-50). Curtis Roads's *nscor*, which uses granular and many other techniques, is found on *New Computer Music* (Wergo 2010-50). *Concret PH*, an early work by Iannis Xenakis, incorporated granular textures and is available from the Electronic Music Foundation at www.emf.org.

YOUR GRAIN, MY GRAIN

Generating grains from prerecorded audio files brings additional complexity to granular synthesis. A grain's waveform may be read from any point in a sound file, in any order. Thus, a huge number of different waveforms may be derived from a single file, simply by varying the read pointer. There are many different algorithms for selecting samples from a source file. The simplest approach would be to use samples as source material for SGS. This can be done by repeatedly playing back samples, given a start point and length.

A more interesting variant would be to move the start point progressively from the start to the end of the source file, which can produce unusual time compression or expansion effects. For example, a 2:1 expansion occurs when the cloud length is twice the source-file length. Usually, the result has more artifacts than you'd get from commercial time-warping products; you can get a variety of fashionably lo-fi sounds, especially if you use more percussive envelopes.

Of course, you can introduce randomness into the sample selection process. At one extreme, you can grab samples at random from an input file, utterly garbling the source signal. Another approach is *statistical evolution*, which means that, at the beginning of the cloud, there is a higher probability that samples will be taken from the beginning of the file; by the end of the cloud, there is a higher probability that samples will come from the end of the file. Statistical evolution still scrambles the source file, but it retains some of the original signal's shape, dynamics, and identity.

THE POOR PERSON'S GRAIN

If you don't have granular synthesis software, or you simply want to try another approach, you can approximate granular synthesis with your favorite sequencer and synthesizer or sampler. The idea is to generate "grains" using a simple oscillator-plus-envelope patch, or a short sample. (Claves, closed hi-hats, and other percussion work well.) You can trigger the grains by rapidly playing back MIDI sequences with short event durations. The amplitude, pitch, and other contours of the resultant "clouds" can be shaped by note and Velocity data and continuous controllers. A lot of editing is required to produce a single cloud! However, modern sequencers can help out quite a bit by generating or randomizing streams of controller events.

This method has its limitations. For instance, the MIDI transmission rate will limit the rate of grain generation, as will the characteristics of the receiving device. MIDI resolution on pitch, amplitude, and other parameters is not as good as that found in direct granular synthesis. Furthermore, some sequencers and synthesizers will behave erratically or even crash if the rate of events is too high. I'll leave debate over whether this technique qualifies as true granular synthesis to purists. I think of it as the "Poor Person's Granular Synthesis" (PPGS). Despite its drawbacks, PPGS can yield interesting sounds. I've used it extensively in one composition ("Projectile," 1987) and occasionally in others. I'm sure I'm not the only **EM** reader crazy enough to try something like this!

Try out some of the software listed in the sidebar, and you may discover an exciting new way to generate and manipulate sound. There are thousands of grains out there just waiting to be synthesized.

John Duesenberry'*s electronic music is available through the Electronic Music Foundation. Check the EMF catalog at www.emf.org.*

WORKING MUSICIAN

Get Started with E-Commerce

It's your turn to make some money on the Web.

By Jennifer Hruska

By now you certainly have heard enough hype about the Web. Supposedly, any musician with a computer can make a fortune off the millions of people surfing the Internet. So is it true? Can you get rich on the Web? Considering that only about 5 percent of all Internet-based companies are even profitable, perhaps we should ask a simpler question: can the Web at least help you make a living doing what you love to do? To that we can answer a resounding "yes."

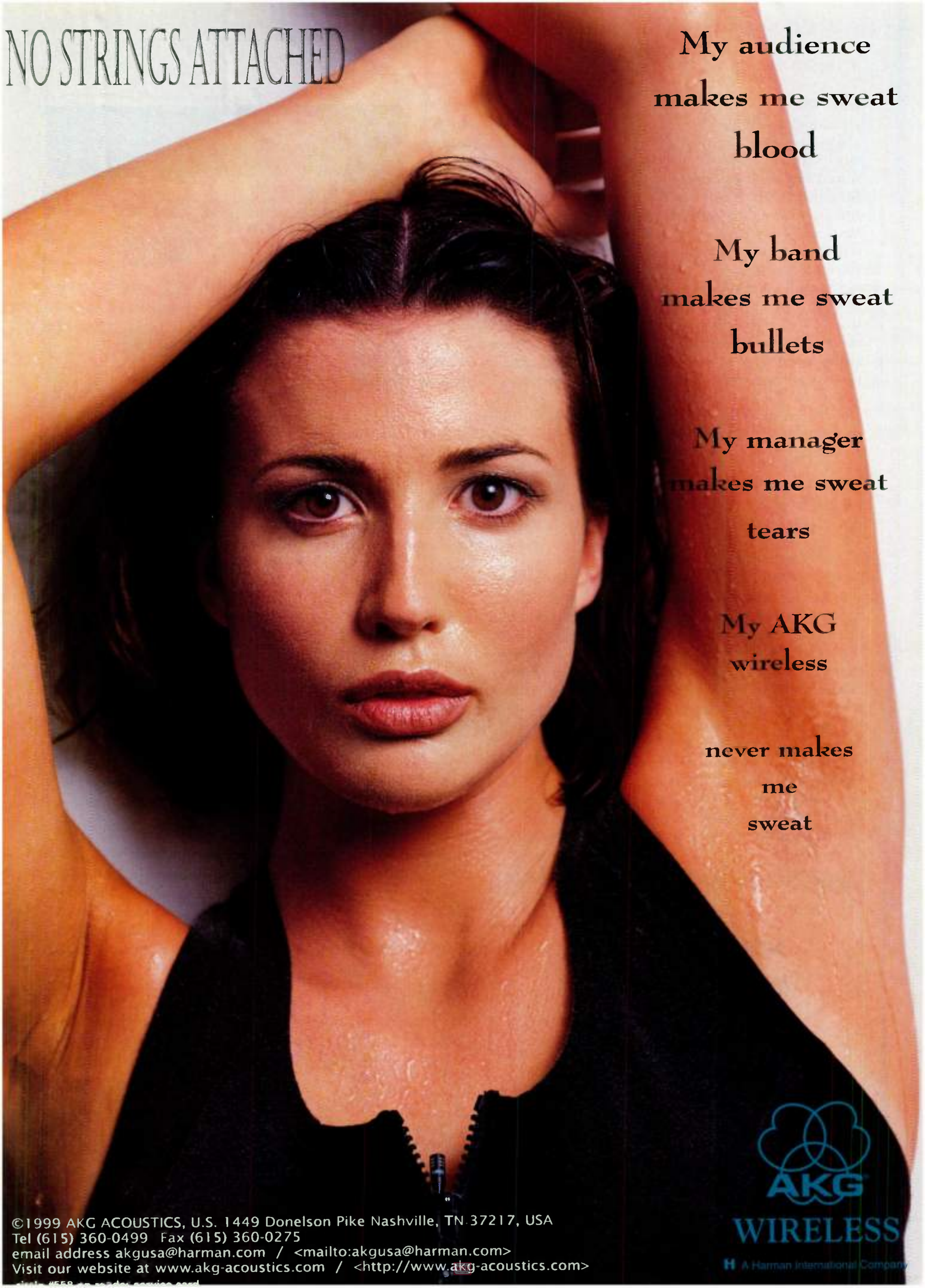
At Sonic Implants, all sound files are offered as downloads only, so the company doesn't have to deal with inventory and product shipping.

Most musicians who use the Web do so to market themselves, their bands, and their projects, but the Internet also holds great potential for selling products such as CDs, T-shirts, posters, and all the other things you offer at your gigs. Sound designers, in addition to doing work for hire, can sell their sounds over the Web.

There are a few ways to go about putting your merchandise up for sale on the Internet. One approach is to shop your wares to an existing Web site that markets and sells products. (Be sure to check out "Working Musician" in next month's **EM** for more information on using third-party Internet distributors.) If you'd like to have complete control over your product marketing and keep a higher percentage of sales dollars, there's another option: market and sell your products from your own Web site. For approximately $2,000 and a few months' time, you can set up electronic commerce on your site and start taking orders. Let's look at the basic workings of e-commerce and how you can get started with it on your own Web site.

SHIPPING AND HANDLING

Once you get your site set up and your merchandise together, you need to consider whether you want to actually ship products or just offer downloadable files. Obviously, if you're selling physical items such as CDs, posters,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and T-shirts, you must ship them. However, downloading is an option if you are selling songs or sounds. My Web site (www.sonicimplants.com) sells sound sets to musicians, and all the files are under 20 MB in size. If you run a download-only site, you avoid the hassles of shipping.

If you do intend to ship products, you must decide whether to do so yourself or hire a fulfillment house. A fulfillment house can take phone orders; store, package, and ship products; and carry out a host of other services. Many also take orders by e-mail, which can be sent to them automatically after a sale is processed on your site. A fulfillment house can perform any number of these services, and the fees will vary accordingly.

TAKE IT TO THE BANK

All businesses that accept credit cards must have a merchant account, so you'll need one too. Your merchant account must be set up through a bank. The bank, in turn, is associated with a credit card processing company, which handles the actual credit card orders (sales, returns, disputes, and so forth). When choosing a bank, you should be aware of several things. First of all, many banks don't really like Internet sales. Even though at the processing level there is no difference between an online sale and a phone sale, many banks view Internet sales as fraught with potential fraud and tend to shy away from those accounts. Banks are also wary of giving merchant accounts to folks who run small businesses out of their bedrooms or who haven't been in business long. Most banks will send a representative to your place of business, so it's best to present a professional look by putting on some nice clothes and cleaning up your place. You will also have to pay an application fee for your merchant account. These fees vary widely from about $200 to $600, so be sure to shop around.

Most e-commerce sites, such as ShopBuilder (above), offer a variety of services—from Web design to database management to order automation.

When you fill out an application for a merchant account, you will be asked how much sales revenue you intend to generate. The bank uses this figure to determine the percentage rate paid to Visa, MasterCard, and other credit cards you want to accept. (All businesses that accept cards pay a fee for each credit card sale, usually 2 to 3 percent of the sale.) Be careful, though, because the bank also uses your sales revenue estimate to determine whether to give you a merchant account at all. When talking to my first bank, I optimistically stated that my sales would be rather high, only to find out that the bank wanted a letter of credit (basically an account with a lot of money in it as collateral) because a company of such small size might not be able to pay such large credit percentage fees. So it's best to estimate your sales as rather small initially, even if it means that the fee percentage you pay is a bit higher. After six months, the bank will review your sales and can adjust your percentage if necessary.

The virtual shopping cart, such as the one on the Vaccination Records site, is the most common approach to selling items over the Web.

Also be forewarned that the process of applying for and receiving a merchant account can take several months and many phone calls. Daunting as this process may be, don't be tempted to believe the promises made by fly-by-night companies that claim they'll get you quickly approved for a merchant account. Most of these operations are scams.

SHOPPING FOR E-COMMERCE

So how does e-commerce work? When someone makes a purchase, a computer server processes the order and sends the sales and credit card information to your bank. Unless you're proficient at computer programming, you'll want to employ the services of an e-commerce provider, such as ShopBuilder (www.shopbuilder.com) or Eagle Software (www.eaglecorp.com), to take over this part of the process for you. Many e-commerce providers also handle credit card encryption, database management, and other important functions. After a buyer provides the relevant information (name, street address, e-mail address, credit card number, and so on), the server accesses a credit card verification service. Once the card is verified, an authorization number is generated and the sale is complete. All authorized sales from a given day then travel in a batch to a processing company that first pays the credit card companies their percentage of the sales. The balance is then deposited into your bank account, typically within three days.

Following are several things you should consider when shopping for an e-commerce provider.

Which encryption method does the provider use for credit cards? The bank at which you have a merchant account will require you to use an approved encryption method. Make sure that the

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Monitoring Perfection

Wedges give you an edge on stage.

By Rudy Trubitt

Although I'm not prone to stage fright, I do approach every gig with a sense of anticipation. Each performance comes with its own set of potential problems. However, few things can sabotage a performance quicker than the inability to hear one's self or fellow bandmates. An awkward stage layout, a lousy P.A., or a noisy audience all impact what you hear on stage. When musicians cannot hear themselves clearly, intonation, timing, blend, and musical interplay suffer to varying degrees.

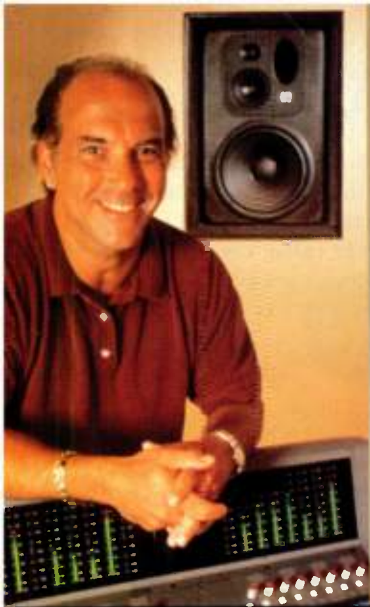
CHRISTOPHE VORLET

Fortunately, there is a solution for the musician who is willing to put some time, energy, and money into solving the problem: it's called a stage-monitor system. Stage monitors are additional loudspeakers that point directly toward the musicians and away from the audience. Usually these speaker cabinets sit on the floor and are tilted upward. Because of their distinctive shape, these cabinets are often called wedges, stage wedges, or wedge monitors. Wedges make it much easier for musicians to hear themselves play, which helps ensure a successful gig.

However, you must consider some important caveats. First, a monitor system adds to the expense, weight, and setup time of your P.A. system. Second, using monitors improperly is often worse than having no monitors at all. When microphones are being used on stage, a monitor system greatly increases the chance of feedback during the performance. And a show plagued by feedback is worse than a show with out-of-tune vocals.

TO EACH HIS OWN

The mix that you'll want to achieve in your monitors isn't necessarily the same mix you would create for the audience. You may need two independent mixes instead: one for the performer and one for the audience. For instance, if I'm doing a solo guitar/vocal gig, I'd want the audience P.A. to carry both the



Neil Karsh is the Vice President of Audio Services for New York Media Group. Recently, Karsh selected LSR monitoring systems for two of his Manhattan facilities, *Lower East Side* and *East Side Audio*.

"We've installed the first of our LSR 5.1 surround systems at East Side Audio and it's a great addition. The sound is extremely clear and is enjoyed by our mixers and our clients. Everyone is very pleased with the result."

New York

LSR. Profiles

The world's most noted recording professionals discuss the world's most advanced monitoring systems.

NO.1: New York / Los Angeles

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Since its introduction in 1997, the system-engineered JBL LSR Series has become a favorite choice of engineers, producers and performers, many of whom have also become its most loyal advocates. More important, this acceptance is found in every major geographic area of the recording industry; from Los Angeles and New York to Nashville and London.

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David Kershenbaum is a Grammy Award winner who has been on the cutting-edge of music production for decades. His discography is a remarkable *'who's who'* of popular recording.

"Speakers have always been important to me and I've had many systems that I have really loved. When Kevin Smith told me about LSRs, I tried them and was amazed at the accurate, flat response and how the mixes translated so well compared to other monitoring systems. Now we're using them to track our new records and we'll use them to mix, as well."

Los Angeles

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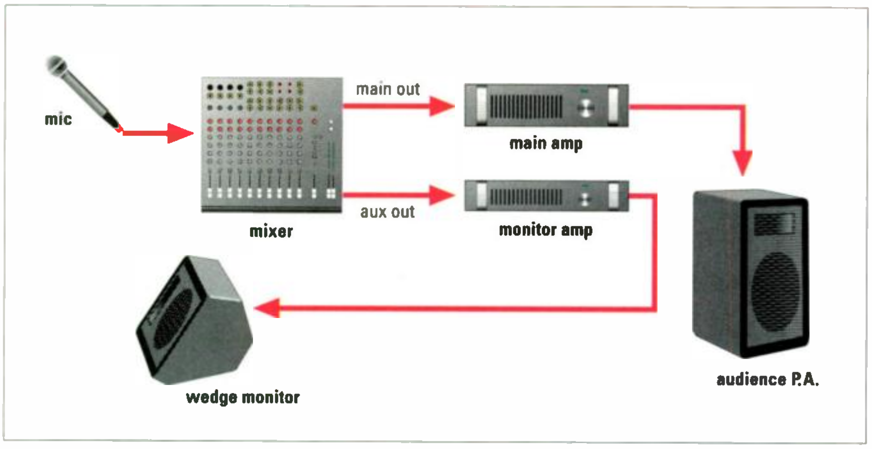

FIG. 1: Here's the signal flow of a minimal P.A. with monitors. A stage-monitor system requires an extra amp and speaker, as well as a mixer with an auxiliary output.

This means that turning down the fader of a vocal mic will automatically turn down the amount of that channel's signal that is sent to an effect, such as a reverb or echo unit. This type of aux send is also called a *postfader* send. It's the right choice when an effects unit is connected to an aux send.

The other type of aux, a *monitor send,* takes its signal from a point before the channel volume fader, hence it's called a *prefader* send (see **Fig. 3**). As a rule, prefader sends are preferred for monitor applications because they provide independence between the monitors and main P.A. This allows you to adjust the house mix without causing a change in the monitor mix.

HOOK IT UP

Enough theory; let's hook it up. Begin by figuring out which of your mixer's aux sends is prefader. Your mixer might include a switch that makes aux 1 pre- or postfader (or there might be such a switch on each and every channel). Find it and set it to prefader. Next, take that aux output jack and connect it to a graphic EQ, preferably one with 15 or more bands. From the output of the graphic EQ, run a cable to the input of a power amplifier channel. Finally, the output of the amplifier goes to your monitor cabinets. In most cases, three 8-ohm monitors is the maximum number of cabinets you should run from a single amplifier channel; using two 8-ohm cabinets is safer.

Now create your monitor mix. Start by turning up the monitor aux send on a channel with a vocal mic. Talk or sing into the mic. If all is well, you should hear your voice from the monitors. If the mixer has an aux-send master knob, make sure that it's turned up and that the output of the EQ and the input of the amp are also set to pass signal. Continue the process with each vocal mic until you can hear each singer in the monitors.

THAT HORRIBLE SQUEALING

Feedback is the most common (and frustrating) problem you'll encounter when doing live sound reinforcement. Feedback happens when a mic "hears" its own amplified sound coming out of a nearby speaker cabinet. As more microphones are added and the overall monitor level increases, the potential for feedback quickly increases as well.

Microphones designed for live sound usually have a tight pickup pattern, so they are more sensitive to sounds coming from the front and will reject sounds coming from the sides or rear. You can increase the mic's volume in the monitor without feedback through careful mic positioning in relation to the monitor.

To position the mics, start with the obvious: make sure the mic is pointed away from the wedge. Because cardioid mics have good rear rejection, you can tilt the mic up toward your mouth, as long as the angle of the mic is perpendicular to the wedge monitor (see **Fig. 4a**). Hypercardioid mics have a narrower on-axis pickup pattern and less rear rejection. Position the hypercardioid mic so that it's parallel to the stage in order to keep the wedge out of its polar pattern (see **Fig. 4b**).

USE AN EQ

Another crucial tool in the fight against feedback is the graphic equalizer. The graphic EQ should be patched between your mixer's monitor aux out and the power amp that feeds your wedge input. As you turn up the monitor level, you'll begin to hear feedback build. Find the graphic EQ slider that corresponds to the feedback frequency and pull it down. Then turn the monitor level up until feedback begins again. Repeat the process until two or three notes begin feeding back at once. This is the practical limit beyond which feedback cannot be suppressed.

Automatic feedback suppressors can be used in combination with, or as an alternative to, graphic EQs. These devices look for concentrated signal peaks at very narrow frequencies. This allows them to identify the feedback frequency and apply a notch filter to reduce it. When shopping for automatic feedback suppressors, pick a unit with as many notch filters as you can afford. You will also need to decide whether you want filters that must be set to specific frequencies before the performance or ones that are automatically adjusted during your set (some suppressors do both).

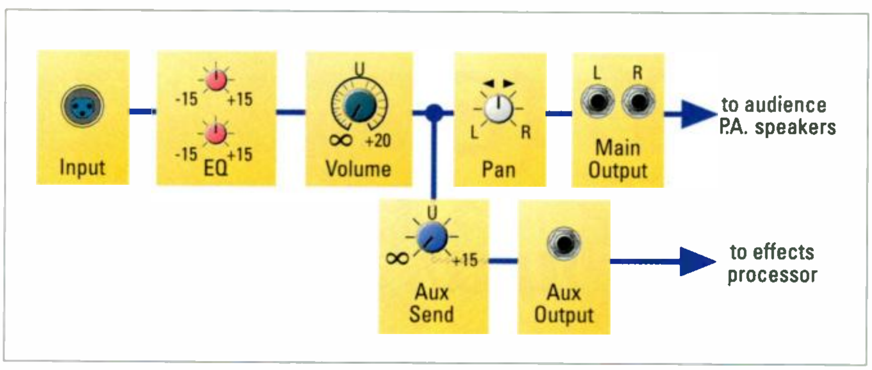

FIG. 2: Here's the signal flow of a mixer channel with a postfader send. When the aux-send level comes after the main volume control, a change in channel volume will also change the level being sent to the aux output. This arrangement generally works best for effects.

FINDING YOUR BALANCE

Once you have feedback under control, it's time to think about fine-tuning your monitor mix. Begin with vocals and quiet acoustic instruments; set aside miked amps for the moment. Don't put drums or bass guitar in the monitors—most small monitor speakers won't be able to reproduce these low-frequency sounds adequately. In most large-scale P.A. systems, the monitor engineer counts on spillover from the main P.A. to provide the lowest of the lows for the musicians on stage. Pumping excessive low end through the monitors muddies up both the stage and audience sound, and at the same time, it reduces the headroom in your monitor chain. Furthermore, putting too many things in the monitor mix pushes the onstage sonic clutter to new heights, quickly negating the advantages of a monitoring system.

Your particular situation will dictate the number and positioning of wedges you need on stage. For example, even nonsinging drummers will need to hear the vocal, both for reference and to gauge their own dynamics relative to the singer. On a small stage, you may be able to position a front-line wedge such that the drummer can hear it as well. Of course, a separate wedge for the drummer is preferable if space and spare equipment permit it.

Multiple wedges can all carry the same monitor mix; in this case you may be able to run as many as three or four stage monitors from the same power amp, as long as the combined load of the connected wedges doesn't exceed the amp's rated specs. On the other hand, if your mixer has more than one available aux send, you could have two or more monitor mixes (for example, one for the lead vocalist and another for the backup singers). This requires an additional graphic EQ and power amp channel for each independent monitor mix.

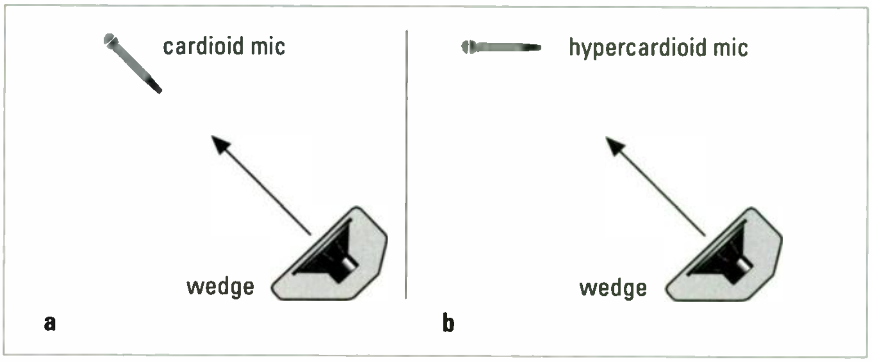

FIG. 4: (a) Cardioid mics often work best tipped up toward your mouth, perpendicular to the face of the wedge. (b) Hypercardioid mics, with their extremely narrow pickup pattern, are often slightly more sensitive to sounds from directly behind them. Try tilting them parallel to the stage. Experiment with both methods if you're not sure which type of mic you have.

DRIVING A WEDGE

It's fairly straightforward to get a monitor system working well enough to improve the intonation and timing of a band's musical performance. But that's only part of the story. Having the right monitor mix helps the players blend their performance with that of the rest of the band. This, in turn, will have a major effect on the total musical impact of your performance.

How does the monitor mix fit into the overall sound-reinforcement picture? Let's consider three background singers who are capable of a great vocal blend. What happens if all three of their mics are set to equal levels in the house P.A., but in the monitors one is set much louder than the other two? The singer who hears herself louder in the monitors will, naturally, pull back from her mic until she hears herself in the monitors at about the same level as the other two singers.

But what does this do to the house mix? Only two of the three vocal parts are heard clearly through the main P.A., because one singer has backed off her mic. If you have the luxury of a full-time sound person mixing your P.A., he or she would be able to compensate for this lower-level voice (assuming they were familiar enough with the material to notice that a harmony part was missing).

Unfortunately, many bands must mix their own house and monitor sound from the stage. In this case, it's really important that the relative levels of your monitor mix reflect the mix levels going to the P.A. This helps the entire band balance their own levels on stage, which is your best and only hope for good sound in the house: there is no way that you, as a performer, will be able to second-guess the house mix and make corrections to it from the stage, especially in the heat of the performance.

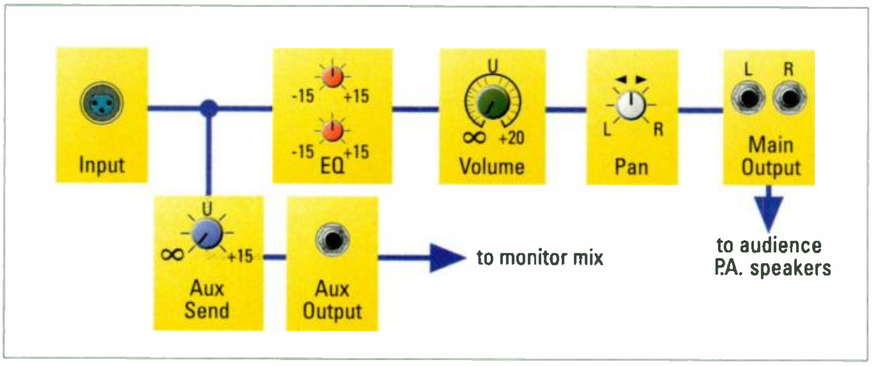

FIG. 3: Here's the signal flow of a prefader send. The aux send taps its signal from a point before the EQ or channel volume knob. This means you can adjust your stage-monitor level for a given channel without affecting that level in the house P.A. and vice versa.

WEDGE ISSUE

Stage monitors are not just for groups playing large clubs and arenas. Any time an ensemble is in a situation that requires a P.A. system, monitors will come in handy. With a small investment and a little practice, a monitoring setup will increase your chances of having a successful gig every time you go on stage.

Rudy Trubitt *is the author of* Live Sound for Musicians *and* Mackie Compact Mixers, *both published by Hal Leonard. Learn more about live sound by visiting www.trubitt.com.*

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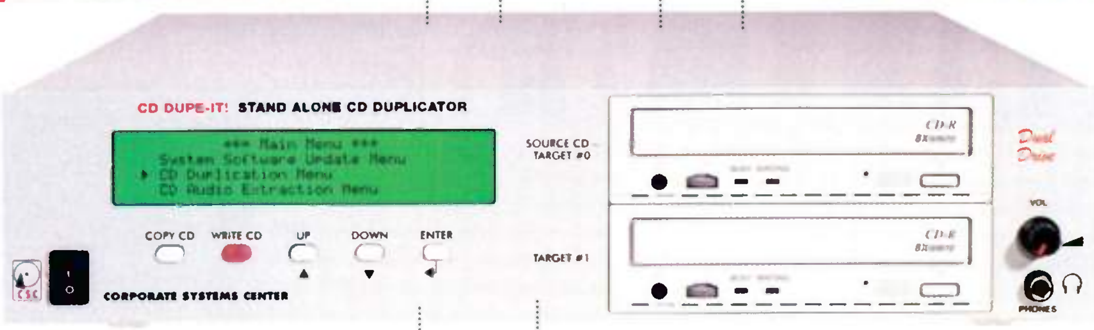

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REVIEWS

AKAI PROFESSIONAL

S5000 AND S6000

Akai messes with a good thing and comes up with a contender.

By Alex Artaud

It's been close to four years since Akai updated its S-series samplers to the S3200XL, the last in a series based on an operating system that debuted in 1988 with the S1000. Since then, Akai has maintained a loyal base of users and secured a strong reputation in the worldwide community of sound manglers and beatheads—all without any major product introductions.

So early last year, when the company announced the S5000 and S6000, with a new operating system (1.21) and a larger graphic interface, people stood up and took notice. The first units were delivered in fall 1998, and several months later some irritating early OS issues were resolved. Now the new samplers seem to be on a solid track. Was it worth the wait? Take a look.

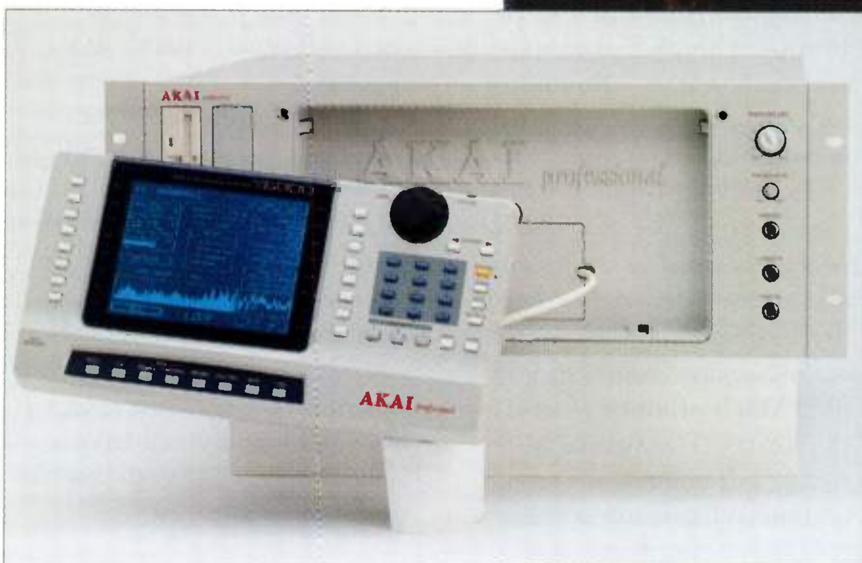


FIG. 1: The Akai S6000 and S5000 samplers sport large LCD screens that make it easy to see what you're doing. The S6000's front panel is detachable.

BASICS

Although the cosmetics of the units are slightly different, the S5000 and S6000 are essentially identical in function. I looked at the S6000, so I'll refer to that model from here on, noting

FIG. 2: The back panels provide plenty of connections, although the digital input/output uses ¼-inch TRS connectors instead of XLRs.

any significant differences between the two samplers as we go.

The S6000 fits in a 4U rackspace and sports a detachable front panel that includes all the main controls. (The S5000 occupies 3U of rackspace, and its front panel is not detachable.) The control panel's centerpiece is a 3.5 × 4.5-inch monochrome LCD screen—huge by any sampler's standards.

Eight function keys are located on either side of the screen. Along the bottom, eight backlit buttons correspond to screens for Multi, FX, Edit Sample, Edit Program, Record, Utilities, Save, and Load—features that should be familiar to S3000-series users. (On the S5000, these keys are located to the left of the display.)

The function keys correspond to labeled buttons on the screen and provide an excellent alternative to wading through various screen menus. You're generally only two or three keystrokes away from any parameter on the machine, so getting lost or distracted is not an issue. In addition, sounds are kept in folders that can include subfolders, which helps you organize them. Fortunately, the screen font is large enough to endear it to the squinting musicians of this world (see **Fig. 1**).

The front panel also includes a data wheel for scrolling values, a numeric keypad, and L/R cursor buttons, in addition to Window, Undo, Exit, Enter/Play, Mark, and Jump buttons. Three user-definable keys let you jump to frequently used screens. (The S5000 does not provide these user keys.) A very welcome PS2 keyboard input lets you name things with a QWERTY keyboard.

The remaining front-panel features (which are not on the detachable part) include main-volume and headphone-volume knobs, a ¼-inch stereo headphone out, and balanced ¼-inch left and right inputs for sampling. A 3.5-inch floppy drive comes with the unit, and there's space for another storage device, such as an Iomega Zip or Jaz drive or a fixed hard disk. (The S5000 has no space for a removable drive but can accommodate a fixed drive.) I experienced no playback glitches when my unit was accessing its Zip drive.

The back panel (see **Fig. 2**) features 16 unbalanced ¼-inch outputs that you can configure as 8 stereo pairs or 16 mono outs. Also, there are separate balanced stereo input and outputs that use XLR connectors. (The S5000 has balanced inputs on the front panel and unbalanced outputs on the back.)

The AES/EBU input and output (software switchable to S/PDIF) use ¼-inch TRS connectors instead of XLR; optical digital I/O is also available. A BNC word-clock connector allows digital audio synchronization. Two 50-pin SCSI-2 connectors are provided, making it easy to daisy-chain SCSI devices, and a handy termination switch lets you put the sampler in the middle or at the end of a SCSI chain.

Two complete sets of MIDI In/Out/Thru ports allow 32-channel operation. That means you can play the S6000

S5000 and S6000 Specifications

	S5000	S6000
Audio Outputs (analog)	(16) ¼" unbalanced	(2) XLR balanced; (16) ¼" unbalanced
Audio Outputs (digital)	(1) ¼" TRS stereo AES/EBU and S/PDIF (software switchable); (1) optical stereo S/PDIF	(1) ¼" TRS stereo AES/EBU and S/PDIF (software switchable); (1) optical stereo S/PDIF
Audio Inputs (analog)	(2) ¼" balanced	(2) XLR balanced; (2) ¼" balanced
Audio Inputs (digital)	(1) ¼" TRS stereo AES/EBU and S/PDIF (software switchable); (1) optical stereo S/PDIF	(1) ¼" TRS stereo AES/EBU and S/PDIF (software switchable); (1) optical stereo S/PDIF
Additional Ports	(2) MIDI In/Out/Thru; (2) SCSI (HD50); word clock; optional ADAT interface	(2) MIDI In/Out/Thru; (2) SCSI (HD50); word clock; optional ADAT interface
Polyphony	64 notes (expandable to 128)	64 notes (expandable to 128)
Multitimbral Parts	128 parts maximum per Multi; up to 32 channels	128 parts maximum per Multi; up to 32 channels
Sampling Rates	44.1/48 kHz	44.1/48 kHz
RAM	8 MB (expandable to 256 MB via SIMMs)	8 MB (expandable to 256 MB via SIMMs)
Removable Storage	none	bay for Iomega Zip or Jaz drive
Frequency Response	20 Hz–20 kHz	20 Hz–20 kHz
Dynamic Range	<90 dB	<90 dB
THD + Noise	0.006%	0.006%
Effects	optional EB20 effects board	optional EB20 effects board
Dimensions	3U x 17.3" (D)	4U x 18.2" (D)
Weight	19.6 lbs.	24.6 lbs.

For manufacturer information, please see Contact Sheet, p. 179.

from two different controllers at once. The internal MIDI file player eventually will be able to control external sound modules, although that feature is not yet implemented.

The S6000 provides three internal slots for optional circuit boards. Currently available are an ADAT board ($299), offering 16 digital outputs, and the EB20 effects board ($399). Other options are currently in development, but information was not available at press time.

Of course, one of the key features of the S6000 is that it doesn't force you to stand in front of your rack all the time. For this review, the control panel lived comfortably on a table top next to my mixer. By adding a 9-pin serial extension cord, I could have moved it to another work area, if necessary.

The S6000 comes standard with 64-note polyphony and can be expanded to 128 notes. Its stock 8 MB of RAM is expandable to an impressive 256 MB. The A/D converters are 18-bit, with a 20-bit D/A stage, and sampling frequencies are selectable between 44.1 and 48 kHz. The unit holds the OS in flash ROM, so it can be easily updated.

BEHIND THE VEIL

The S6000 provides three basic operational modes: Sample, Program, and Multi. I'll start with an overview and cover some of the important features in more detail shortly.

As you might expect, Sample mode lets you capture and edit samples using functions such as Normalize, Rescale, Trim, Chop, Loop, and Fade Up/Down.

Other digital signal processing features include BPM Match, Time-stretch, Pitch Shift, and EQ. Once you're satisfied with a sound, you can assign the sample to a Program.

Programs are composed of up to 99 Keygroups, each of which can contain up to four stereo samples. Keygroups let you map your samples across a keyboard with a lot of flexibility. Each sample is assigned to one of four zones, which you can layer or Velocity-switch or -crossfade.

Program mode is where you apply all detailed synthesis parameters. These parameters include 26 distinct resonant filters, two ADSR envelope generators and an auxiliary envelope, two LFOs, and assignable program modulation (APM) with up to 17 destinations. APM should be familiar to users of Akai samplers, and a similar feature is found on most high-end samplers. It provides modulation routing from MIDI controllers, LFOs, and envelopes to destinations such as pitch, filter, amplitude, and panning (see **Fig. 3**).

Once you've tweaked your Programs to perfection, you can assign them to Parts within a Multi. The S6000 can simultaneously play up to 32 Parts from a MIDI sequencer in Multi mode, and each Part's level, output assignment, effects send, fine-tuning, and MIDI channel can be adjusted. Incredibly, up to 128 Multis can be loaded into memory at a time.

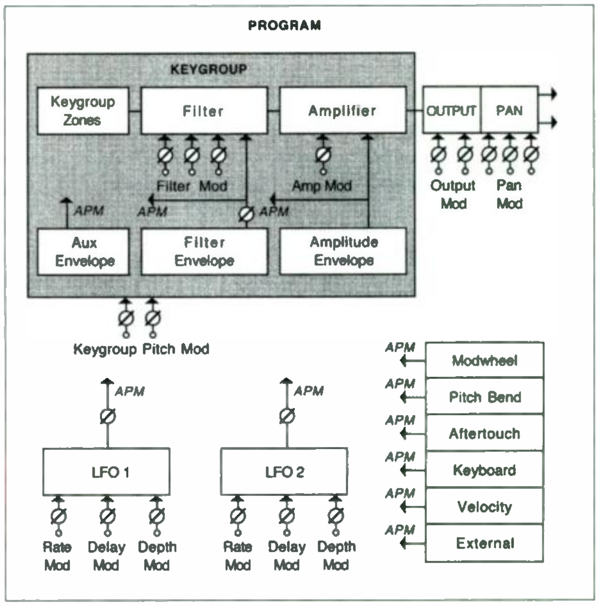


FIG. 3: You assign samples to Keygroups and establish all modulation routings within a Program.

SAMPLING MADE EASY

Recording a sample into the S6000 is a quick and painless process. The main Record screen displays all relevant items: manual start, threshold start, length of sample (including length in real time as well as number of samples), source selection (analog, digital, or ADAT), and mono/stereo setting. A progress bar scrolls during sampling, and you can abort or complete a process at any time. Once you're finished, you see a waveform, and a screen prompts you to either keep or trash what you've recorded. If the sample clipped, you are prompted to keep or discard. Sample names can be typed in from a computer keyboard or scrolled in with the sampler's data wheel.

One of the best features on the Record screen (and in the S6000 as a whole) is the Record To option, which lets you select whether the sample is recorded into RAM or directly to a hard disk. This opens up the possibility of triggering long samples from a hard disk without being dependent on sample RAM.

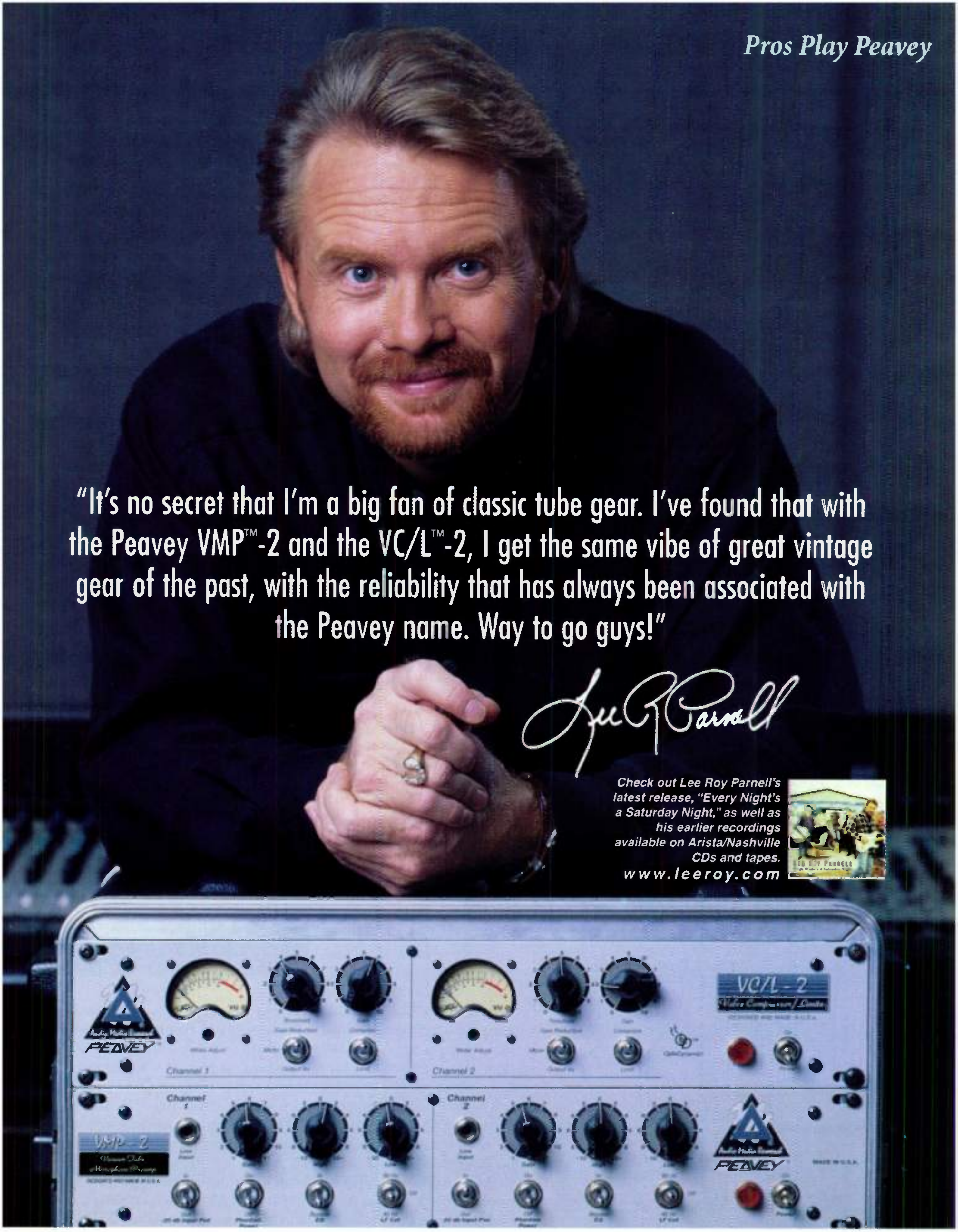
Even better, you can apply the S6000's powerful editing and modulation capabilities to such a sample. Imagine playing a six-minute sample

MAKING WAVs

Akai has chosen WAV files as its standard audio-file format. This is a blessing for PC users, although Mac fans will groan and wonder why it couldn't have been AIFF. However, this is the least of a computer user's concerns.

It is possible to connect a Mac or PC and the S6000 to the same external SCSI hard disk, but doing so could be courting disaster. If the computer and S6000 try to access the drive at the same time, nasty SCSI crashes could corrupt the disk. Akai discourages users from sharing drives with a computer, but it offers tips to those who care to brave these waters.

In addition, Akai provides no means to perform SCSI transfers to and from programs such as Steinberg *ReCycle* or BIAS *Peak*. The company will move in this direction with a future version that should be available when you read this. In the meantime, Akai suggests a work-around involving two Iomega Zip drives (or any other removable-media device). This method is safe but painfully slow, breaking up any working rhythm you might have established. Watch for future developments.



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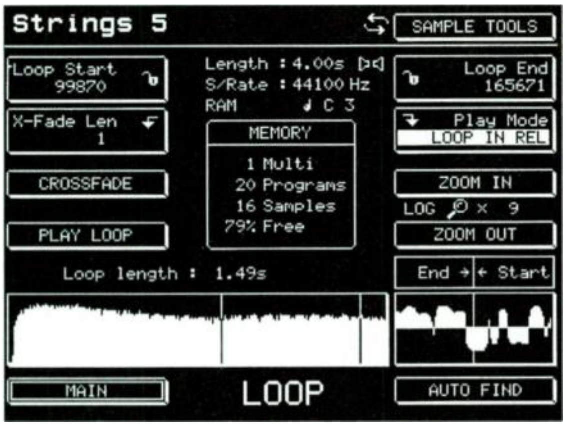

FIG. 4: The Loop subpage within Edit Sample lets you set the loop parameters for a particular sample.

while filtering and applying LFOs in real time, and you get the idea. There are some drawbacks, however. Samples that are not in RAM require longer edit times, and—brace yourself—you can't loop them. (Maybe in the next OS update?)

Once you have a sample to work with, the Edit Sample screen allows you to navigate to a number of subpages. The main screen indicates the length, sample rate, and sample type (RAM or disk); the loop status; and the mono/stereo status.

In the Edit Sample subpages, a large waveform display lets you zoom in for precision trimming, chopping, and looping. There are some particularly good features here. For example, you can extract smaller samples from larger ones and use the Play To and Play From soft buttons to fine-tune an edit. Both of these functions help you isolate materials for looping.

The Loop page itself is quite thorough (see **Fig. 4**), with scrollable start and end points, crossfade functions, and Auto Find for loop points (which is cool, if only to see what the box spits out). The Play mode determines whether the loop is one-shot, repeated until released, or repeated in release (the loop continues as long as the associated envelope is in its release phase). This feature is good for avoiding abrupt endings and creating seamless layers. Play Loop is great, although it occasionally sticks if you move either start or end points too quickly in sequence. This does not cause a crash, but Akai should fix this bug soon.

Other looping functions include Loop Lock and Loop Direction. With Loop Lock, the start and end points span a fixed length of time, and both points can be moved together through a sample to isolate the loop you want to use. Loop direction can be forward, reverse, or back and forth for truly weird results.

Samples can be faded up or down, with a choice of linear, sine, or logarithmic curves. You can also crossfade or mix two samples together. All changes can be auditioned before they become final, so you won't replace a great sound with a useless belch.

The Timestretch and Pitch Shift functions are well implemented and contain a curious group of presets that actually work fairly well, generating very few sonic artifacts. The BPM Match function does what its name implies, but you have to know the tempo of your source material to use it. A tempo finder is critical on a sampler of the S6000's caliber, and Akai should have included one. In addition, there is no way to assign start or end points for these functions.

On the plus side, resampling works very nicely to provide that low-bit crackle for those who need it. The three-band EQ is excellent, with high- and low-frequency shelves at 6 dB/octave and 12 dB/octave, as well as adjustable Q for the mids (500 Hz to 10 kHz).

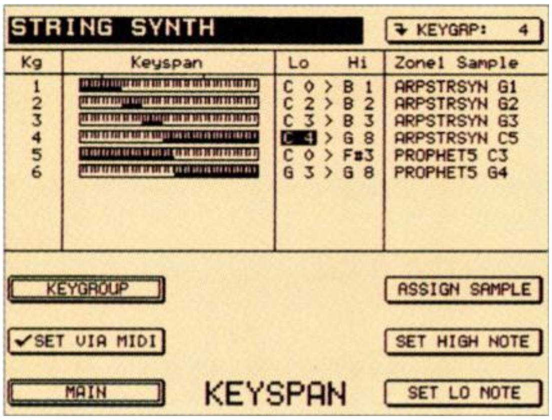

FIG. 5: The Keyspan subpage within Edit Program lets you assign different samples to specific keyboard ranges.

Over time, I found I could edit samples on the fly with increasing speed. The Edit Sample page is designed for smooth operation.

FROM PROGRAM TO MULTI

As I mentioned earlier, Edit Program lets you assign and modulate samples within a Program. There are far too many parameters to cover in detail here, but we'll look at a couple of key features that relate to performance and synthesis.

Within the Output subpage, you can adjust level and Velocity sensitivity. Various modulators can affect amplitude and panning, allowing auto-pan effects. The MIDI/Tune page enables you to select Programs remotely through MIDI commands, tune the Program, and select from eight different tuning templates (for example, just intonation). You can also create your own tuning templates and save them. Pitch-bend range is adjustable, and you can even select which notes will bend. In addition, the S6000 responds to both Channel and Polyphonic Aftertouch, and Aftertouch can bend notes up or down, as well as perform many other types of modulation.

Nine LFO waveforms are available,

AKAI PROFESSIONAL
S5000 and S6000 stereo samplers
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FEATURES ■■■■
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AUDIO QUALITY ■■■■
VALUE ■■■▌
1 2 3 4 5

PROS: Large graphic display with sensible layout and folder-based organization. Disk-based sampling. Fast Sample/Program/Multi loading. Extensive modulation routing and filter types. PS2 keyboard port. Dual SCSI ports. Optional ADAT interface. Good EB20 effects.

CONS: No SCSI transfers to external sample editors. No MIDI file player implemented. No tempo finder. Minimal support of non-native file formats.

CIRCLE #437 ON READER SERVICE CARD

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including sine, triangle, variations on square and sawtooth, and random. You can synchronize LFOs and modulate their rate, delay time, and depth. LFO2 mirrors LFO1, with two notable exceptions. You can specify whether LFO2 will retrigger with each Note On, and you can synchronize LFO2 to MIDI Clock from your sequencer. In that case, a Clock Division parameter sets the number of beats for each LFO cycle. The S6000 performed this function without a hitch when receiving from Emagic *Logic Audio*.

Within the Keygroup and Keyspan subpages (see **Fig. 5**), you can spread your samples across the keyboard and assign different samples to different Velocity ranges within a zone. Each Keygroup's pitch and amplitude can be modulated, and its overall level can be adjusted. A strange quirk in the nomenclature (going back to the S900) is that you select Copy to create a Keygroup. This may confuse Akai novices until they browse the manual, which is quite good.

The S6000's resonant filters are superb; they include an assortment of lowpass, highpass, bandpass, notch, and peak filters in addition to a phase shifter and a bizarre Vowelizer that simulates vowel formants. The filter envelope includes an assortment of templates for various acoustic instruments. Filter and envelope shapes are all displayed graphically within these screens. The amplitude envelope controls overall amplitude, and you also get an auxiliary envelope to use wherever you want.

As mentioned earlier, up to 128 Parts (and the Programs assigned to them) can play back within a Multi (see **Fig. 6**).

FIG. 6: Multi mode lets you combine up to 128 Programs into a multitimbral construct.

Parts and Programs have the same edit features, but you can't scroll through Programs in the Edit Part screen, as you can in the Edit Program screen; you assign Programs to Parts in the Multi screen. You can automatically load Multis on power-up—a helpful feature for live performance.

Fortunately, the S6000 can read all S1000 and S3000 files. You can convert older Akai stereo samples to interleaved WAV files and even convert older Multis.

FX TO BURN

The S6000 accepts an optional EB20 effects board that replaces the EB16 board. It has four effects sections, called FX1, FX2, RV3, and RV4. Algorithms within FX1 and FX2 include ring modulation/distortion, EQ, modulation (chorus/flange/pitch shift), delay, and reverb (see **Fig. 7**). Several of these include templates and graphics. RV3 and RV4 are dedicated reverbs. I'm quite familiar with Akai's effects, and these are the best I've heard so far—most likely because of the S6000's 20-bit D/A converters. In particular, the extensive EQ functions alone are worth the price of the effects card. However, there are some limitations: for example, you can't modulate parameters in real time through MIDI, nor can you save effects settings. But your effects settings *are* saved within a Multi file.

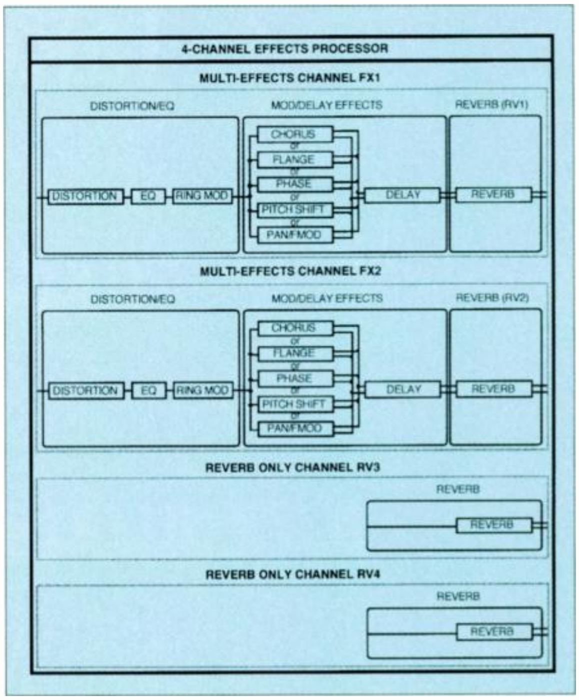

FIG. 7: FX1 and FX2 provide many different effects, while RV3 and RV4 offer dedicated reverbs.

GROWING PAINS

Akai leapt into the fire pit when it released version 1.0 of the S6000's software. Many users encountered unstable machines and had to wait for features to be implemented. The list of desired improvements grew, and to its credit Akai responded, overhauling the operating system with five updates in the past year. (The company is making an updated manual available on its Web site as a PDF file.)

Now, a year since its introduction, the S6000 is reasonably stable (I did have a mysterious crash that could have been related to my sequencer), and users are getting work done—although perhaps not entirely in the way they expected (see the sidebar "Making WAVs").

There is one unfortunate design element that isn't likely to change any time soon: if a sample is used in more than one Program or Multi, the sample itself is duplicated on the hard disk. This is a waste of storage space, but Akai reasons that having multiple copies fits well into its folder-based organizational model by keeping your compositions' sounds grouped together with little chance of loss. Nevertheless, I hope Akai implements pointers to overcome this.

The S6000 has made great strides in the venerable Akai lineage with an intelligent, intuitive interface and powerful features. It's basically a well-designed machine that's gone through some growing pains. With a great foundation and engineers who care enough to respond to users' needs, it should only improve with age.

Alex Artaud *is the editor of the Spanish-language edition of* Mix.

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MIXMAN TECHNOLOGIES

MIXMAN STUDIO PRO 3.0 (WIN)

Remix your brains out.

By Erik Hawkins

DJs have been remixing music live for years by blending records, sampled beats, and drum machines into cohesive, seamless grooves. Through their remixes, a unique school of music production has emerged—one that promotes looping, beat matching, and transforming. You can hear this school's influence in almost every form of popular music, from an alternative track with scratching to a country song with a loop. With the acceptance of DJs and their craft into mainstream culture, it seems as though everyone wants to try a hand at remixing.

Unfortunately, unless you're a practiced turntable wizard or a parameter-happy samplehead, remixing is no stroll in the park. At least it wasn't until Mixman Technologies came along. Thanks to its *Mixman Studio Pro*, anyone with a PC can cook up a hot remix. This groundbreaking program has an original user interface that lets you combine sounds and grooves on the fly without doing your own sampling or worrying about timings. At the same time, if sampling and beat matching are what you live for, you'll find enough parameters and options in *Mixman Studio Pro* to delve deeply into the world of professional remixing.

ANATOMY OF A MIXMAN

Mixman Studio Pro is a suite of four separate programs. Each represents a stage in the remix process, and together they make up a complete virtual remixing studio. The startup window displays an image of each: the Remixing Studio for remixing, the Recording Studio for sampling, the FX Studio for effects processing, and the Editing Studio for sequencing (see **Fig. 1**). Clicking an image takes you to that studio, and once inside, you can navigate to other studios by clicking buttons on the sides and bottom of the page. Getting around is simple and intuitive.

Mixman's main windows are modeled on an old-style television set, with each studio appearing as a picture on the tube. Switching studios changes the picture to a new user interface while keeping the controls on the TV frame pretty much the same (see **Fig. 2**).

Buttons at the bottom of the frame are action-oriented (such as, Play, Stop, Record, Save, and Load). Button sets change to match the current studio but always follow the same layout—a set of buttons on either side of an oval-shaped display that provides different information for each studio, ranging from bars and beats in the Remixing and Editing Studios to a VU meter in the Recording Studio. Toolbars for each studio are found on the left and right sides of the picture frame. The tool symbols and button hieroglyphics take a while to master, but holding the cursor over a button usually yields a pop-up label to let you know what you're looking at.

Stereo LED-type meters appear at the top of the TV frame straddling a text window that displays the current file's name and size. The 10-segment multicolor meters offer no peak hold option or dB markings, but they get the job done.

MANY A WAVE TO MAP

Mixman works with 44.1 kHz mono or stereo samples in WAV or Mixman's proprietary Track (TRK) file format. TRK files are used to address "beat-mapped" samples, which are samples that have been cut into several smaller samples to define and capture a recording's component beats. These discrete samples are then mapped to an event list (which is essentially a tempo map of the original sample's groove) for playback in sequential order. Steinberg's *ReCycle* software performs beat mapping, but slicing up a sample in *ReCycle* and sending the keymapped beats to a sampler—and an associated MIDI file to a sequencer—requires external gear, a solid grasp of MIDI, and additional software. With beat-mapped TRK files, *Mixman* acts like a sampler and a sequencer rolled into one; no extra equipment, software, or MIDI components are required.

TRK files are much more flexible than standard WAV files. With TRK files you can tune a sample without changing its number of beats per minute, and you can change a remix's bpm without tuning the sample. A WAV file has no tempo map, so changing its pitch (tuning it) speeds up or slows down its tempo, causing it to become longer or shorter than it needs to be. Furthermore, changing the remix's bpm causes the WAV file to be out of time, because no tempo map tells its beats where they

FIG. 1: *Mixman Studio Pro* consists of four separate "studios," which are represented in the startup window. Clicking one of the images takes you directly to that studio.


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FIG. 2: *Mixman*'s Remixing Studio is where you perform the actual remixing using two virtual turntables with 16 slots for live triggering of samples.

should land. Of course, a WAV file can be forced to adhere to a specified tuning and tempo setting without beat mapping by applying a time-stretching algorithm, but this is not usually a real-time operation, so it's not appropriate for live remixing. Beat-mapped TRK files, on the other hand, let you make tuning and bpm changes on the fly.

If sampling and beat mapping are not your cup of tea, Mixman Technologies offers a variety of ready-made, beat-mapped TRK files. *Mixman Studio Pro* includes a collection of 450 royalty-free TRK files and 100 WAV files; others are offered on CD-ROMs that cost $39.95 each. The styles range from acid jazz to jungle, house, and funk. Everything I heard sounded great. (I especially liked the India TRK files; they were exotic and funky with some wonderful percussion and a cool vocal sample.) The CD-ROMs include TRK files from such musical luminaries as George Clinton, K-Klass, and Skinny Puppy. According to Mixman Technologies, the conversion of more name-act tracks to *Mixman*'s file format is on the way. Watch the company Web site for regular updates on what's available. (Incidentally, Mixman's site is fantastic. It's informative and interesting, with lots of free remixes. A radio show dedicated to *Mixman* remixes should be up by the time you read this.)

REMIXING STUDIO

Mixman's Remixing Studio is where you do most of your work, from previewing samples to performing the actual remix. The process centers on two virtual turntables in the middle of the screen (see **Fig. 2**). Each platter has eight slots for holding the samples that you want to use in your remix. Double-clicking a slot opens a window from which you can browse your hard drive for sounds. Once you've decided on a sample, click the Load button and it's assigned to that slot. Slots have an "X" in them when they're empty, so it's easy to see which slots have samples. To see the name of a sample, simply move the cursor over the slot to trigger a pop-up label. Unfortunately, the pop-up labels don't work in either Play or Record mode, and that makes it difficult to see which slots hold which samples while you're remixing.

With the slots loaded, it's time to start remixing. Click the Play button (or the Record button if you don't need to rehearse) and trigger a sample. The tone arm swings over the platter and a red strobe light turns on, just like on a real turntable. Triggering a sample is done by clicking a slot with your mouse or by pressing a key on your computer keyboard. The mouse just turns samples on and off, while the keyboard provides a variety of triggering options.

The keyboard is the program's primary real-time controller. Multiple samples can be triggered simultaneously by holding down several keys at once. A sample's playback can be locked (looped indefinitely) by holding down a sample's assigned key and a Lock key at the same time. You can execute breaks and solos, turn groups of samples on and off using macros, and start or stop playback from the keyboard. The manual fails to mention the fact that the left/right arrow keys allow you to scroll through a sample's parameters: pan, volume, pitch, and tempo. (Displays for these parameters are just beneath the turntables.) The up/down arrow keys let you adjust the parameters while remixing without having to touch your mouse.

Though *Mixman Studio Pro*'s QWERTY controller system works reasonably well, it's anything but perfect. Computer keys are not designed for live remixing; they are small and densely packed, which makes them difficult to differentiate. Try using this controller in a dimly lit nightclub, and you're apt to have problems seeing what you're doing. Mixman Technologies has been developing a dedicated controller that will plug into your PC's keyboard port; it should be available by the time this review hits the newsstands.

Nonetheless, what *Mixman* really needs is MIDI compatibility so that it can take advantage of the great MIDI controllers that are already out there, such as the Keyfax PhatBoy and the Peavey PC 1600x. Live remixing with *Mixman* will then be the cat's meow. Mixman Technologies claims that MIDI for the Windows platform will be available before year's end and that *Mixman* for the Mac will have MIDI as soon as the software is released.

The Remixing Studio will not let you load samples while it's playing. In other words, you have to stop playback to access samples other than the 16 that are currently loaded. That's a disappointment—it means that you can't flow seamlessly from one set of remix samples to the next, and you can't change the remix smoothly over time by loading in new samples. In a professional setting, that can cause problems: at some point, the song will end, people will leave the dance floor, and you'll be twiddling your thumbs while the next sample set loads. If you plan to use *Mixman* at a gig, I suggest having another sound source handy (such as a drum machine) to fill in the down times.

FIG. 3: *Mixman*'s Editing Studio has a specialized sequencer with a graphic display and color-coded bars.

For manufacturer information, please see Contact Sheet, p. 179.

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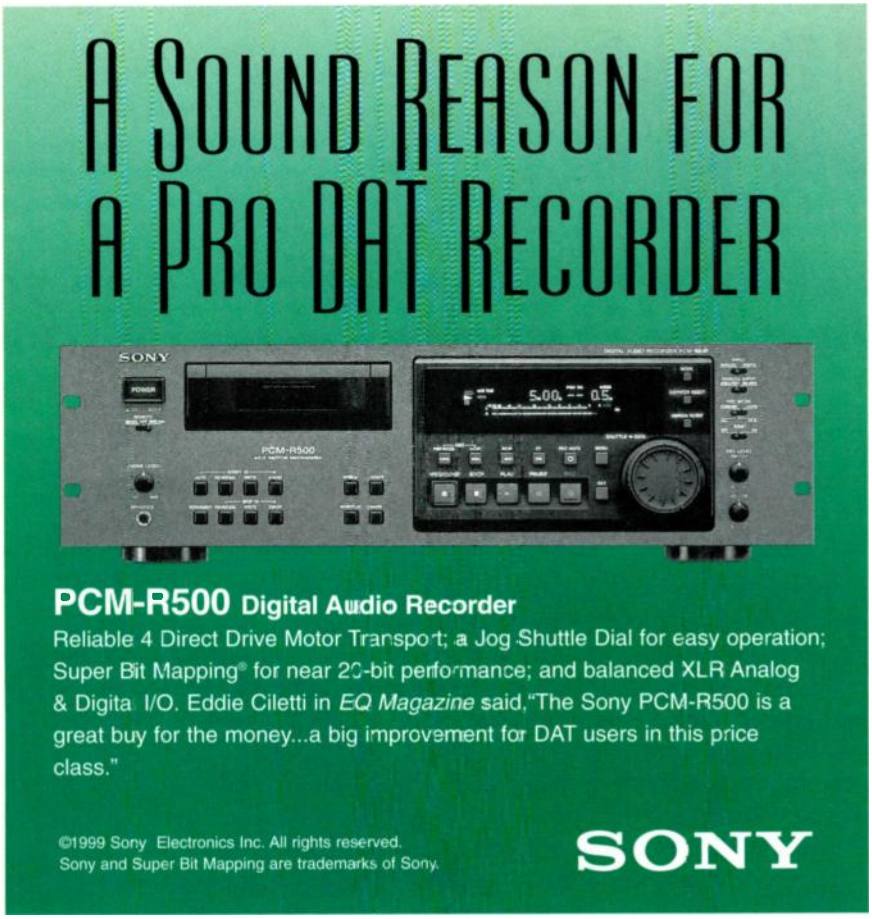

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EXTRA SETTINGS

When you first choose a sample for loading into the Remixing Studio, you can make adjustments to how it plays back. You don't have to fiddle with these settings every time you load a sample, because samples work fine with their default settings. However, as you get deeper into the program and become more experimental, the extra settings become invaluable.

Samples can be time-stretched or compressed. The algorithm produces good results and is useful for fine adjustments. As with most time-stretch algorithms, however, big changes cause artifacts. Samples can be fine-tuned to a hundredth of a semitone. This is different from tuning a sample after it's been loaded into a turntable slot, because there you can make changes only in semitone increments.

A function called Time Shift provides complete control over where a sample starts within a bar. A sample can trigger on any beat or subdivision of a beat, with up to 480 ppqn resolution. A sample's loop length may also be adjusted. For example, if you have a 4-bar loop but like only the first two bars, and you want the loop to start on the upbeat of the second beat, set the loop length to two bars (instead of its default of four) and set the time shift to 720 ticks. That's it. Adjust these settings before loading the sample, and it will always trigger on the beat you specified, regardless of when you hit its sample key.

Samples can be told to always play for their full duration (that is, a full release) whenever triggered, or just for the amount of time a sample key is held down. Minimum Spacing lets you adjust the amount of time that must pass before a sample can be retriggered. This is handy for controlling double triggers (if you accidentally hit a sample key twice). You can set the Minimum Spacing to a specific duration (a 16th note, for example) and intentionally double-trigger to

Exporting in MP3 format makes your remix Internet ready.

Mixman Studio Pro
Minimum System Requirements
Pentium 200; 32 MB RAM (64 MB is recommended); 16-bit sound card; Windows 95/98; DirectX 5.0 or later; 35 MB hard-disk space (210 MB for a full installation)

create some cool in-time echo effects.

In addition, a new feature called WARP (Wide-Band Audio Real-Time Processing) offers real-time DSP audio effects that are triggered by mouse movements within a special display. WARP allows you to have as many as 12 presets loaded at one time (although there are several more to choose from), and you can switch between presets at any time during a performance and change parameters with mouse gestures as the music is playing. You can then store the real-time data as part of the performance.

JUST FOR WAVES

Mixman provides special settings for WAV files that are not applicable to TRK files. WAV files need to be perfectly looped or they won't play back properly in *Mixman*. If you need a waveform editor, Steinberg *WaveLab Lite* is included with *Mixman Studio Pro*. However, because WAV files are not beat-mapped, *Mixman* must know a WAV file's tempo and number of beats. *Mixman* can calculate a tempo automatically after you enter the number of beats. I discovered that the automatic tempo calculation worked great in the preview menu but didn't always result in a perfect loop when the sample was actually loaded into a slot. Manually entering the tempo yielded rock-steady results. (Hint: *ReCycle* can calculate the exact bpm of a sample based on its length and the bars/beats that you enter. You can then give that figure to *Mixman* as the tempo value.)

My favorite WAV parameter is Synchronized Start. With this function activated, a sample can be turned on and off (repeatedly) at any time during its playback while always remaining in time. In other words, hitting a sample key doesn't really trigger the sample but instead unmutes it. The sample keys then act as mute/unmute buttons rather than triggers. This opens up myriad performance choices, from transforming (rapidly clicking the sound on and off in time with the music) to isolating specific beats with a well-placed unmute (for example, unmuting only the snare in a loop). The only glitch I ran into with Synchronized Start was a strange amplitude ramp (a very rapid fade-in) at the beginning of certain WAV files. This caused trouble on some percussion samples, because it removed the transients of the first downbeat. Sounds that had slow attacks, however, presented no problem.

RECORDING STUDIO

The Recording Studio is quite rudimentary. You can record WAV files from your sound card's mic inputs, line inputs, or internal CD-ROM drive. Auto Normalize lets you automatically normalize your sample after recording. An Auto Trim feature allows you to strip silence at the beginning and end of your sample. You can adjust its threshold and pre- and post-strip times in decibels and milliseconds, respectively.

Unfortunately, there's no way to listen to your remix while recording, so rapping or singing along with the beats can be difficult. A metronome is available, but I couldn't get it to work. According to Mixman, the company has had problems getting the metronome to address certain sound cards, so you can't always use it. The only other

MIXMAN TECHNOLOGIES
Mixman Studio Pro 3.0
remixing software
$89.95

FEATURES ■■■■
EASE OF USE ■■■■▌
DOCUMENTATION ■■■▌
VALUE ■■■■■
1 2 3 4 5

PROS: Inexpensive. Easy to use. Lots of exporting options. WAV-file compatible. Addresses beat-mapped samples. Includes cool effects.
CONS: No MIDI support. Can't load samples during play. Manual lacks an index.
CIRCLE #438 ON READER SERVICE CARD

options are to move your remix to an external medium (such as cassette, DAT, or multitrack) and work with it there, or transfer the tracks internally (as WAV files) to a multitrack program for recording. Of course, for just recording samples and sound effects, listening to your remix while recording isn't crucial. The Recording Studio is fine for this.

EDITING STUDIO

The Editing Studio (see **Fig. 3**) is a simple and straightforward sequencer. Because I'm used to high-end sequencers (such as Emagic *Logic* or Mark of the Unicorn *Performer*), I thought *Mixman*'s sequencer was going to be a joke. I was wrong. It's certainly no bells-and-whistles sequencer, but it's well suited to editing your remix. It's also fast and intuitive, and it doesn't get in the way of the creative flow.

Triggered samples are represented by color-coded bars: green represents a locked sample, orange a manually triggered sample (played with a keystroke), blue a pitch-adjusted sample, red a soloed sample, and gray a muted sample. (Turntable slots use this same color scheme.) With the different colors as visual cues, you're able to quickly determine what's happening at any given moment. The only thing that would improve the user interface is if the names of the samples appeared on the tracks. (Pop-up labels give you each sample's name but only one at a time and not during playback.)

Navigation is extremely simple. A slider lets you zoom in and out. Clicking on a measure number at the top of the window moves the playbar to that location. There are also markers, but they seemed unfinished, as if they were an afterthought. I got some markers into my project, but they appeared misplaced; they were on the window frame instead of neatly placed inside the actual editing window. Action keys on the frame skipped the playbar between the markers, but clicking on the markers themselves had no affect. Nevertheless, I didn't really miss the markers, because clicking on a measure number is easy enough.

Clicking and dragging allows you to select an area to edit. Various tools allow you to draw in or erase events, cut, copy, paste, and quantize.

Great automation features abound. Simply select the bars you want to automate, click on the attribute you want to work with (volume, pan, tempo, or pitch), and enter a beginning value and an ending value. There are displays for each parameter just beneath the edit window. Each display has two fields to show the beginning and ending values of selected bars. Unfortunately, you can't select bars that are playing to see their changes; playback must be stopped to view a selection's values.

FX STUDIO

Although *Mixman Studio Pro*'s effects aren't the best I've ever heard, they are great for processing samples. With very little tweaking, it's easy to create dirty, underground, dance, and ambient sounds. Up to five effects can be cascaded (or chained) together in the FX Studio (see **Fig. 4**). A total of 23 effects include auto pan, flange, filter, delay, pitch shift, time stretch, wah, and reverb algorithms, and all of the effects have an ample number of presets to choose from (more than 150 altogether). Most sound good right off the shelf, but if you don't like what you hear, there are plenty of parameters to play with. I particularly liked the filter effects; they were quite dramatic—perfect for adding that hard-core edge. *Mixman*'s effects processing is not done in real time. (You can't audition effects changes while the sample is playing.) If you have a slow computer, count on taking a coffee break while some algorithms, especially long reverbs, process.

These are the kind of effects that would sound really sweet in stereo; unfortunately, they are all monophonic. Even Auto Pan is essentially a mono effect. It works by assigning a pan value to the sample whenever it's triggered. The result is not a smooth panning effect, like when you twist the pan knob on a mixing board, but instead a choppy effect as the sample gets stuck, for its duration, at a pan position each time it's triggered.

You can rename and save samples after processing for loading into any remix. Your own effects presets can also be saved. User presets can contain a single effect or multiple effects. Recalling an effect is as simple as clicking on an effect slot and loading the preset. Just don't try to load a multi-effect preset if you've already got an effect loaded—you'll crash the program.

A new feature called Time Adapter allows you to import files with widely differing tempos. It adjusts the beats to match without destroying the original feel of the tracks by slowing down or speeding up the samples too much. In other words, a loop with a fast tempo can stay fast while still being metrically matched to a slower tempo.

SAVE OUR REMIX

Once you've created a remix, you'll want to save it. *Mixman*'s proprietary remix files are called Mix files. They contain performance data only (event lists, a sample directory, an effects directory, and so on). Because of this, they take up very little hard drive space. If you prefer to keep all of a remix's samples together instead of in different folders, there's an option for storing the Mix file with its samples in a single folder. However, doing this creates duplicates all of the samples from the other folders and eats up significantly more hard drive space.

Because Mix files can be read only by *Mixman*, playing your remix outside of the program requires that you save it to a different format. A variety of options are available: stereo WAV, TRK, RealPlayer G2, MP3, Windows Media Audio, and SoundFont (SF2). Saving as a WAV file yields a 44.1 kHz, 2-track master that's ready for burning to CD-R. Exporting as a WAV file is amazingly fast; a six-minute piece took only a couple of minutes to process.

There is little point in exporting as a

FIG. 4: In the FX Studio, you can cascade up to five effects and have them applied to your mix nondestructively.

TRK file, because only *Mixman* can read this format, unless you plan to reintegrate the 2-track master back into your remix. Exporting in RealPlayer G2 or MP3 format makes your remix Internet ready. You can also save your remix in Creative Labs' Sound Blaster format (as an SF2 file); it includes all of a remix's samples saved as separate SoundFont instruments with an associated MIDI file. This is great for throwing the whole remix into a multitrack audio program that is able to read SF2 files, so you can do more complex work on the track—for example, mix down with individual outs or apply true stereo processing.

I thought *Mixman's* sequencer was going to be a joke. I was wrong.

HOW PRO CAN YOU GO?

Although its low price belies the fact, *Mixman Studio Pro* is no toy. Live remixing and the production aesthetics that go along with it, however, are not for everybody, so *Mixman* will probably be considered a "pro" program only by DJs, sampleheads, and other people working in alternative dance genres like jungle, deep house, and electronica.

Mixman Studio Pro may not be perfect, but it's hard to quibble when you consider how novel and inexpensive the program is. It doesn't support MIDI and can't load samples on the fly, and those are certainly shortcomings. But in the end, how serious you get with *Mixman* is entirely up to you and your creative inclinations. Because the program uses 44.1 kHz samples and has a variety of importing and exporting options, sound quality and cross-platform compatibility aren't issues. This program is on the cutting edge, and it's a real bargain to boot. So keep an eye on *Mixman*, because it's the future of live remixing.

Erik Hawkins *is a musician/producer working in Los Angeles County and the San Francisco Bay Area. Check out his fledgling indie label at www.muzicali.com.*

CAKEWALK/PEAVEY

STUDIOMIX

Two industry heavyweights deliver an impressive hardware and software package.

By Allan Metts

Over the years, I've watched *Cakewalk* grow from a basic MIDI sequencer to a full-fledged MIDI and audio tool. With all that flexibility and power, however, comes a multitude of onscreen controls: countless faders, buttons, and knobs vie for limited display space. And using a mouse to control all those parameters can become tedious rather quickly. Ever try mixing a 32-track production with just a mouse? It's like trying to work a real mixer with just one finger. Forget it.

Enter Peavey Electronics. This venerable manufacturer has teamed up with Cakewalk to create an excellent hardware addition to the *Cakewalk* experience. The StudioMix package combines a hardware surface controller—sporting real knobs, buttons, and motorized faders—with the powerful *Cakewalk Professional* digital audio sequencing software.

Already a *Cakewalk Professional* user? Well, unfortunately, you can't purchase just the hardware controller. (For existing *Cakewalk Professional* owners, the box does contain a free offer for *Audio-FX* effects plug-ins. A free upgrade to *Cakewalk Pro Audio* 9 is also available to anyone who has recently purchased StudioMix.) However, if you use *Cakewalk Pro Audio,* you don't have to downgrade to *Cakewalk Professional*—the StudioMix controller is supported as of *Pro Audio* v. 8.04.

PLUG IT IN

EM has covered *Cakewalk* many times, so for this review I'll just focus on the features that are specific to StudioMix. Starting from the top: installation was a breeze. I installed the software, wired up the hardware, and voilà—it worked. Without even cracking the manual, I successfully made adjustments to the *Cakewalk* transport and onscreen faders using the StudioMix controller.

The controller itself is reasonably compact, occupying less than a foot of space from front to back. If it were a keyboard, it would be about two and a half octaves wide. The surface slopes up gently, and there is a nice wrist rest in front.

It comes with nine fader groups; typically, you'd use eight to control individual tracks and one as a master. Each fader group contains one 60 mm motorized fader, two rotary encoders (infinitely rotating knobs), and one button. These controls have no labels to indicate their function, but there are spaces provided to scribble down your own notes. Your use of *Cakewalk* will determine what adjustment should occur when a control is tweaked.

StudioMix
Minimum System Requirements
Pentium 120; 16 MB RAM (32 MB RAM for Windows NT); Windows 95/98/NT 4.0; 16-bit sound card or MIDI interface

To the right of the faders are five additional buttons that trigger various software events or commands, six transport control buttons, and a jog/shuttle wheel. One of the transport control buttons switches *Cakewalk's* MIDI Machine Control messages on and off, allowing you to control an external tape deck with the same set of buttons.

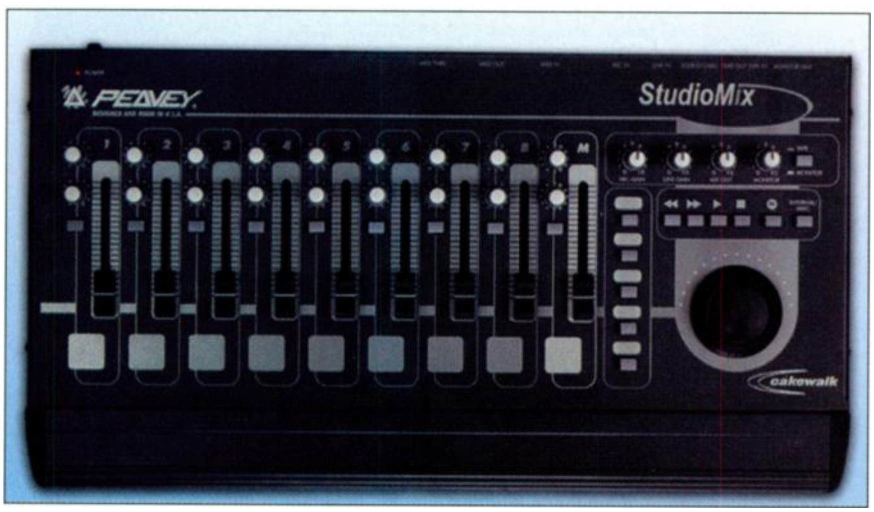
SPLIT PERSONALITY

The StudioMix controller isn't just a MIDI device; it has audio capabilities as well. On the back of the unit is an XLR microphone input that is supplied with nondefeatable phantom power. There are stereo sets of RCA connectors for a sound card output; a second stereo output that can be used for a DAT, CD, or tape recorder; a stereo line input; and a stereo feed for your monitoring system. Also included is an ¼-inch headphone jack.

There are separate gain controls for the mic and line inputs, and separate level controls for the monitor and mixdown outputs. What's more, the monitor signal can be switched between the sound-card and tape-deck inputs. But unlike the other StudioMix controls, these knobs don't send or respond to MIDI messages.

I'm always wary of inserting new devices into the audio chain, so I decided to test the noise levels of the StudioMix audio connections. I noticed a bit of hiss and a whining noise when I turned the levels way up. I tried to isolate it by removing my sound card from the chain, eliminating it as a possible culprit. The noise was still there, but it was hardly noticeable at normal monitoring levels. As for the mic preamp, it sounded fine to my ears. I wouldn't put it up against a high-end professional preamp, but it's perfectly decent for casual use.

Those with complex studios probably won't have much use for the StudioMix audio section. Instead, you're likely to have your sound card, microphones, and sound modules wired into a larger



Cakewalk and Peavey have teamed up to deliver an impressive hardware surface controller for Cakewalk's software. The StudioMix package is easy to set up and a breeze to work with.

RELIVE THE PAST REGULARLY.

With the Yamaha 01V, you can create the exact same mix every time, right down to the EQ settings on input 12. That's because the 01V digital mixer lets you save every detail of its 32 parametric EQs, 22 limiter/compressor/gates, two 32-bit effects processors and 15 motorized faders.Then, when the time is right, you hit one button and recall the mix precisely and instantly. And, with an external sequencer, you can even let the 01V perform the entire mix. So last night's performance sounds exactly like tomorrow's.

The 01V comes with 24 inputs, 6 busses, 6 aux sends and 12 mic preamps. If that's not enough for you, link 01Vs together to create a much larger digital console without paying the price. In fact, the 01V comes in at a paltry $1,999, far less than the cost of all the "external" digital gear it includes free.

With its advanced components and its ability to save past mixes, the Yamaha 01V should make your mixing future quite clear. Get one today at an authorized Yamaha Pro Audio dealer.

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mixing board. But if you create your music with little more than a keyboard and a microphone, the StudioMix mixer could be all you need. It's ideal for recording to and mixing from a single 2-channel sound card. In fact, I did just that.

To test the unit, I moved StudioMix, a small keyboard, and a microphone to my office computer (which has a basic 2-channel sound card). I found myself with a nice little composing station.

This exercise made me wish for one more StudioMix feature. Most sound cards come with only one set of external MIDI ports; StudioMix eats that up, leaving you without a place to plug in an external controller directly. StudioMix does have a MIDI Thru connection, so you could connect the computer MIDI Out to StudioMix MIDI In, StudioMix Thru to your external controller In, and finally the external controller Out back to Computer In. You could also purchase a third-party MIDI merger device, but an extra MIDI input with a built-in MIDI merger would be a welcome addition to the StudioMix console.

HEAR AND OBEY

The StudioMix controller itself is not programmable. Each knob, button, and fader puts out a specific MIDI message on MIDI channel 16; this cannot be changed by the user. It's up to you to configure *Cakewalk* to respond to the messages the way you want.

To do this, go to the new StudioMix Configuration window in *Cakewalk* (see **Fig. 1**). This window provides a drop-down list for each type of StudioMix control (except for the transport controls, which can't be changed). Each drop-down list contains the possible behaviors that the control is capable of. Changing *Cakewalk*'s response to a button, fader, or knob is as simple as picking a behavior from the list.

The StudioMix controller addresses eight consecutive tracks at a time. The behaviors for each of the track controllers are set globally. In other words, you can't configure StudioMix so that one track's upper knob controls pan position while another track's upper knob controls reverb. But you can specify differing behaviors for audio tracks and MIDI tracks.

The fader and two rotary controls can be used to send MIDI Control

For manufacturer information, please see Contact Sheet, p. 179.

StudioMix is designed to connect directly to a sound card, and it offers analog inputs and outputs for connection to a mixdown deck. However, it does not provide digital I/O.

Change messages, including CC 93 (Chorus Depth), CC 91 (Reverb), CC 10 (Pan), or CC 7 (Volume). For audio tracks your choices are pan, level, and any of up to 16 aux buses that you have configured. I set up the signal level with the faders, pan with the lower rotary control, and effects amount—either through an aux send on audio tracks or a controller level on MIDI tracks—with the upper rotary controller.

The track buttons can control mute, solo, track arming (for recording), aux bus enabling (on audio tracks only), and Write Fader. Write Fader is an option for recording additional automation data on a track that has existing automation data. Basically, it disables the motorized fader movements while recording the new data, so you and *Cakewalk* aren't fighting each other for control of the fader.

All of the choices for the track buttons are common operations, and here I found myself wanting more buttons on the StudioMix controller. For example, I will often mute some tracks and arm others during the tracking process. I finally decided to set the buttons to Arm for tracking and then to Write Fader during mixdown. I muted and soloed the old-fashioned way. (In this case, "old-fashioned" means using your mouse to make changes onscreen.)

MASTER MIXER

Your choices for the master fader group are a little different. The button can control the pre- or postfader status for the aux buses and can also enable the Write Fader mode for the master fader. The fader and rotary knobs can control left or right master signal level, as well as send, return, and pan levels for the aux buses.

If you want to control the left and right master level with one controller (for example, the master fader), you simply link the two onscreen faders together in *Cakewalk*'s Console View window. The master fader group can control any of *Cakewalk*'s available audio outputs; you decide which one by choosing from a drop-down list in the StudioMix Configuration window.

The five buttons to the right of the faders are capable of performing a number of functions. In fact, they can be configured to use any of *Cakewalk*'s Key Bindings (which let you assign about 99 percent of *Cakewalk*'s functions to computer keyboard or MIDI shortcut keys). For example, they could execute common responses to prompts like Yes, No, OK, or Cancel, or they could launch anything found on a *Cakewalk* menu. (Save, Print, Cut, Paste, Reverse Audio, and Insert Marker are all fair game.) In addition, you can use the buttons to open a StudioWare panel or run a CAL program. I set one up to undo the last operation, giving me a quick way to abandon a botched recording.

One of the more notable Button Bindings (that's what *Cakewalk* calls them) is the ability to shift the track focus of the StudioMix controller in steps of one or eight tracks. So if the controller is linked to *Cakewalk* tracks 1 through 8, you can press a button to have it control tracks 9 through 16 instead (or tracks 2 through 9, if you're shifting tracks one at a time). You can travel in both directions through the tracks. You can also shift the master controls to different audio output ports in a similar manner.

FIG. 1: The StudioMix Configuration window provides everything you need to map StudioMix controller messages to *Cakewalk* events and actions.

Rounding out *Cakewalk*'s StudioMix Configuration window are resolution settings for the rotary controls and for the jog/shuttle wheel. You can choose between coarse, medium, and fine resolution, although the wheel can also be set to Frame, Beat, or Measure. Incidentally, the jog/shuttle wheel can be used only to move *Cakewalk* to different parts of your song. You can't program it to do other things. (Not that I'd want to anyway—using the wheel is an incredibly quick way to scoot around in a song or locate hit points in a video file.)

PUTTING IT TO USE

Now that StudioMix is set up just right, how do you actually go about using it? The closest tie-in between StudioMix and *Cakewalk* exists in *Cakewalk*'s Console View (see **Fig. 2**). You can consider the StudioMix controller the physical manifestation of what you see on the Console View screen (or at least a portion of it). Move a Console View fader, and the corresponding StudioMix fader moves along with it. Twiddle a StudioMix knob, and you should see something move in the Console View.

When I'm tracking, I set the StudioMix

CAKEWALK/PEAVEY
StudioMix
MIDI control surface
$899.99

FEATURES
EASE OF USE
AUDIO QUALITY
VALUE
1 2 3 4 5

PROS: No-hassle setup and configuration. Intuitive integration with the Cakewalk software. Flexible controller assignment.
CONS: Needs more track buttons. Minimal printed documentation. Audio section is geared to consumer-grade sound cards.
CIRCLE #439 ON READER SERVICE CARD

circle #584 on reader service card

TC WORKS

SPARK 1.01 (MAC)

A fresh approach to real-time audio editing and processing.

By David Rubin

Now that Mac users are well ensconced in the post–*Sound Designer* era, the once-moribund marketplace for stereo audio-editing software is finally showing new signs of life. MicroMat, for instance, has staked out the low-end, under-$50 territory with its surprisingly capable *SoundMaker*, and BIAS's *Peak* 2.0 has matured into a well-designed powerhouse for pro-level users. (For a detailed survey of 2-track audio-editing programs, see "Shaping Better Waveforms" in the March 1999 issue of **EM**.) Furthermore, several Mac-based digital audio sequencers now offer in-depth audio-editing capabilities, heaping more pressure on developers of stand-alone editing software.

Into this highly competitive environment, TC Works has introduced its new *Spark* digital audio editor. With a price tag of just under $500, this program is clearly aimed at serious professional-level users. According to TC Works' promotional material, *Spark* is the "ultimate integrated audio-editing and mastering solution for Mac OS" and is intended for sound design, editing, and mastering.

As we'll see, the program is not the all-encompassing editing solution one would expect from the marketing claims. Its greatest strengths lie in its powerful and versatile processing and mastering tools rather than in detailed waveform-editing and post-production features. Indeed, when judged as a mastering and processing program, this newcomer is off to a pretty good start, offering numerous real-time features; ASIO and Digidesign Direct I/O drivers; CD burning; and support for VST plug-ins, a variety of sampler formats, and 24-bit, 96 kHz audio. In short, *Spark*'s collection of high-quality plug-ins, combined with its unique real-time processing environment, lets you do some very cool things.

HERE'S LOOKING AT YOU

Spark is a great-looking program, with nicely drawn 3-D controls and tastefully designed graphics. For the most part, the program works smoothly and intuitively and seems admirably resistant to crashing. That's pretty impressive for a first release.

When you first launch *Spark*, you see the large Browser View main window and a floating Transport palette that contains lifelike transport controls and an elapsed time display (see **Fig. 1**). If the Transport palette gets in your way, you can always hide it and use menu commands or key combinations to operate the program; *Spark* provides a substantial number of key combinations for navigating, performing tasks, and editing files.

The Browser View is divided into three components: the Wave Editor in the lower half of the window provides a standard waveform display; the File View in the upper left shows a familiar hierarchical file tree; and the Playlist in the upper right contains a list of selected regions. This all-in-one-window interface design lets the user see, in a single display, the elements that TC Works feels are most often used during editing. Indeed, sound designers and sound-effects editors will appreciate the File View section when it comes to locating, adding, and organizing files. And having the Playlist directly above the Wave Editor is handy if you spend most of your time preparing files for CD mastering.

Whenever you select and save a region in the waveform (by choosing Create Region), it instantly appears in the File View, where the relationship between the region and the parent audio file is graphically delineated. You can then drag any region's icon from the File View into the Playlist.

Although the Browser View appears to be a single window, its three components are not all fully active simultaneously. When you click in one of the displays, a thin blue line appears around its border to indicate that it is the currently active section. The remaining sections are then at least partially disabled. For example, if you listen to a recording with the Wave Editor active and then click in the File View, playback continues unabated. But if you click in an empty Playlist instead, playback stops immediately. That takes some getting used to.

You can drag the horizontal border between the upper and lower halves of the window to add real estate to the lists or to the waveform, but you can't completely eliminate any of the three components. In a like manner, you can

FIG. 1: *Spark*'s Browser View main window is divided into three sections with movable borders: the File View is in the upper left, the Playlist in the upper right, and the Wave Editor at the bottom. The floating Transport palette includes a multifunction jog/shuttle wheel.

For manufacturer information, please see Contact Sheet, p. 179.

FIG. 2: *Spark*'s Recording dialog box allows you to keep adding takes into the same file by automatically creating new regions.

drag the vertical divider between the File View and Playlist to reveal more information in one section at the expense of the other.

This emphasis on border dragging makes the window rather awkward to reconfigure. For example, if you want to maximize your view of a long Playlist, you may have to intrude on the File View section and shrink much of the Waveform Editor. That involves dragging two separate borders. If you later want to do some close-up editing on the waveform, you'll probably have to drag the horizontal border again. If you lose track of a file or region and want a large view of the file tree, you may have to drag the borders yet again. I'd rather have separate windows that I could bring to the front or minimize with a single mouse-click.

Also, if you want to view all of the columns in the Playlist, you'll probably have to do some scrolling; I couldn't open the section wide enough on my 19-inch monitor to see all of the columns at once—the presence of the File View prevents maximizing the Playlist width—and *Spark* doesn't let you rearrange columns or delete unnecessary ones to conserve space.

CAPTURING SOUND

Recording in *Spark* is relatively simple and straightforward: clicking the Record button in the Transport palette opens a Recording dialog box with an excellent set of dedicated peak-hold meters and a second set of transport controls (see **Fig. 2**). You can set the meters to display the input level, the level after a plug-in, or the level that is recorded to the file. What's more, the Recording dialog box allows you to keep adding takes into the same file by automatically creating a new region for each take. You can even apply any two plug-ins during recording. To top it off, the program can detect the incoming audio's sampling rate and convert to the target rate on the fly.

Spark can read and write several file formats, including AIFF, WAV, and Sound Designer II. It can import and export *Sound Designer* playlists and can export its own playlists to Adaptec *Toast* (included with *Spark*) or *Jam* for CD burning. All settings—track ID, crossfades, emphasis, and so on—are saved with the image file, so you can simply click *Jam*'s Write Disc button and let it burn.

Spark supports QuickTime; you can import a movie and edit its soundtrack as you view the movie in a dedicated QuickTime window. The Transport

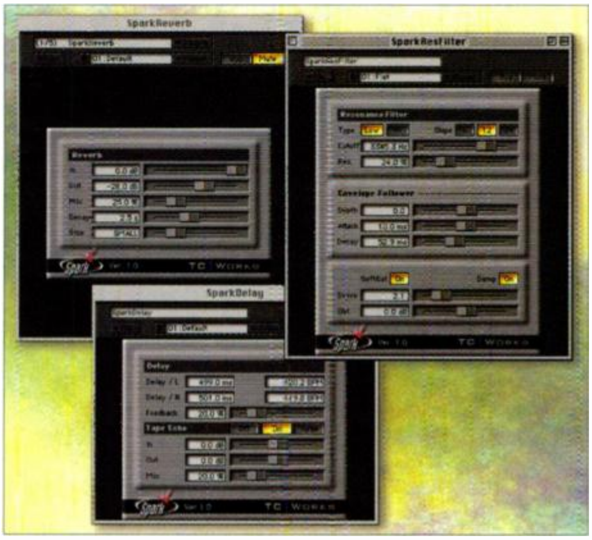


FIG. 3: *Spark* comes with a varied collection of VST plug-ins that share a similar interface design.

palette's jog/shuttle wheel effectively scrubs both audio and video together. The program also imports any file type supported by QuickTime, including MP3 (decoding only).

The program's support for a wide range of samplers is noteworthy. It can transfer samples to and from the Akai S1000, S2000, and S3000; E-mu ESI and E4 series; Kurzweil K2000 and K2500; Roland S760; Yamaha A3000; and any sampler that supports the MIDI Sample Dump Standard or the SMDI protocol. In addition, *Spark* reads Akai samples directly from CD-ROM and automatically splits interleaved stereo files during transmission.

For converting large numbers of files into various formats, *Spark* provides a Batch Converter window that is adequate for most tasks but is not especially feature laden. Batch-processing options let you save files in AIFF, WAV, or Sound Designer II format with resolutions of 8, 16, or 24 bits and sampling rates from 8 to 96 kHz. You can also dither, normalize, and correct DC offset during conversion. Unfortunately, you can't apply any plug-ins or other effects—so you can't, for example, prepare a bunch of sound effects for a Web page by converting them to 8-bit audio and running them all through a bandpass filter in one operation.

MAKING WAVES

Most editing activities revolve around the Wave Editor, which provides a typical waveform display along with an overview. Selecting an area in the overview instantly fills the lower display with the selected area. That offers a great way to quickly zoom in on part of an audio file; simply select a narrow slice of the waveform in the overview, and that section is spread out across the main display.

Spark has separate vertical and horizontal zoom controls in the lower right corner of the Wave Editor. A small arrowhead, wedged between the two sets of controls, provides a drop-down menu of zoom options, so you can skip directly to another level rather than step through the levels one at a time. You can also change the zoom levels in the Zoom Factor field at the bottom of the window, where neighboring fields supply information about the currently selected region. During playback, you can watch the cursor scroll through the waveform, or you can have the cursor remain centered as the waveform scrolls along behind it—a very cool feature.

The File View section is nicely integrated with the Wave Editor: clicking on a region in the File View highlights the corresponding area in the waveform display. That makes it easy to locate and audition regions—even from different audio files—before editing, processing, or dragging them into the Playlist.

Spark's drag-and-drop capabilities are quite helpful when you're handling files or regions in the Browser View window. In addition to dragging regions from the File View into the Playlist, you can drag files directly from the Desktop or from other windows and drop them into the File View to add them to a project. If you drag a region from the File View onto the Desktop, it becomes a separate file with its own icon.

MISSING IN ACTION

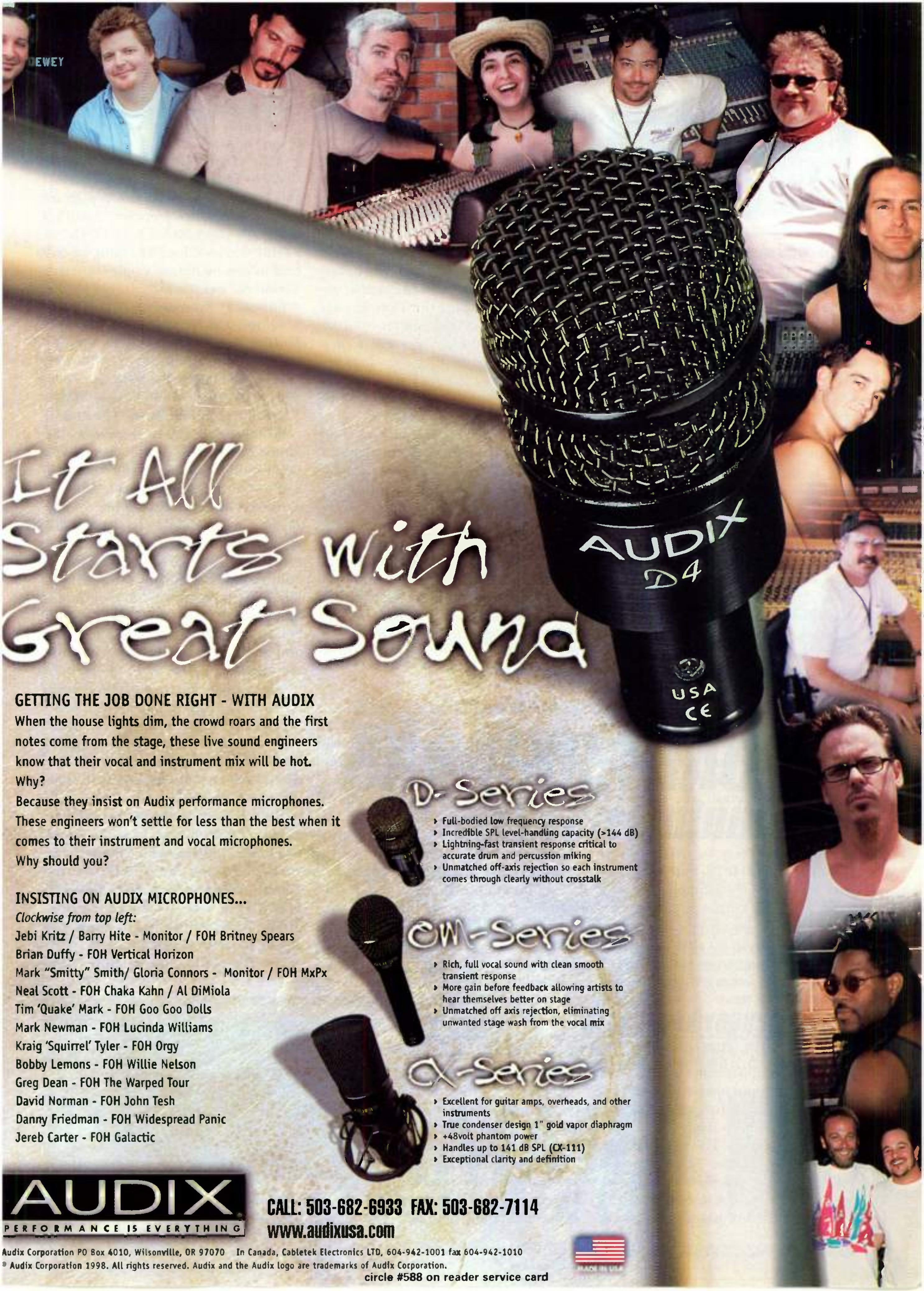
In studying the program, it quickly becomes apparent that TC Works has lavished its attention primarily on *Spark*'s impressive real-time audio-processing and stereo-mastering abilities (more on this shortly). Missing from the current version are several waveform-editing tools that many users expect in a high-end audio editor.

Aside from the usual Cut, Copy, and Paste commands, for example, *Spark*'s Edit menu offers only a Silence and Trim command for modifying waveforms. In fact, the Edit menu isn't even mentioned in the documentation. The Process menu adds several more powerful commands, but most are surprisingly minimalist. The Fade command, for example, provides three fade-in and three fade-out shapes, but no user-definable envelopes (with a graph and grab handles) for custom fades. Other commands include Reverse, Invert, DC Removal, and Change Gain. The Normalize command is an all-or-nothing proposition; you can't, for instance, normalize to 90 percent of maximum. That's a disadvantage if you plan to do more processing after normalizing.

The Pitch Shifting command lets you adjust the pitch in cents over a ±5-semitone range. But the best command in the group is Time Stretching, which allows you to specify a destination length in samples, time, or tempo in a range from 125 percent to 75 percent of the original. Time Stretching produces excellent results, especially with dialog, which it handles smoothly and usually without noticeable artifacts. The results with music are also quite good.

The Wave Editor lacks a Crossfade command, making it difficult to smoothly splice together pieces of audio material. *Spark*'s only Crossfade function resides in the Playlist section, which performs nondestructive real-time crossfades between Playlist entries. Double-clicking an entry in the Playlist opens a dialog box where you can specify a hard cut, overlapping cut, or one of five crossfade types. As with the Fade command in the Wave Editor, however, the Playlist's Crossfade function offers no user-definable envelopes.

The Wave Editor's Marker implementation is quite limited. You can add a Marker to the waveform during playback, during recording, or when playback is stopped, and each Marker appears as a vertical line with a small red diamond on top. That's about it. You can't label the Markers; you can't even number them. There is no Marker window to display a list of the Markers and their positions. You can reposition a Marker by dragging its diamond, but Markers serve little functional purpose in locating, editing, identifying, or looping material.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It All Starts with Great Sound
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CE
GETTING THE JOB DONE RIGHT - WITH AUDIX
When the house lights dim, the crowd roars and the first notes come from the stage, these live sound engineers know that their vocal and instrument mix will be hot.
Why?
Because they insist on Audix performance microphones. These engineers won't settle for less than the best when it comes to their instrument and vocal microphones.
Why should you?
INSISTING ON AUDIX MICROPHONES...
Clockwise from top left:
Jebi Kritz / Barry Hite - Monitor / FOH Britney Spears
Brian Duffy - FOH Vertical Horizon
Mark "Smitty" Smith/ Gloria Connors - Monitor / FOH MxPx
Neal Scott - FOH Chaka Kahn / Al DiMiola
Tim 'Quake' Mark - FOH Goo Goo Dolls
Mark Newman - FOH Lucinda Williams
Kraig 'Squirrel' Tyler - FOH Orgy
Bobby Lemons - FOH Willie Nelson
Greg Dean - FOH The Warped Tour
David Norman - FOH John Tesh
Danny Friedman - FOH Widespread Panic
Jereb Carter - FOH Galactic
D-Series
Full-bodied low frequency response
Incredible SPL level-handling capacity (>144 dB)
Lightning-fast transient response critical to accurate drum and percussion miking
Unmatched off-axis rejection so each instrument comes through clearly without crosstalk
OM-Series
Rich, full vocal sound with clean smooth transient response
More gain before feedback allowing artists to hear themselves better on stage
Unmatched off axis rejection, eliminating unwanted stage wash from the vocal mix
CX-Series
Excellent for guitar amps, overheads, and other instruments
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FLEXIBLE EFFECTS

Much of *Spark*'s powerful processing and mastering functions are controlled from the Master View window (see **Fig. 5**). This window is composed of two parts: the Master section for monitoring and adjusting output levels, and the FX Machine for processing with plug-ins.

The Master section provides a set of long-throw faders with accompanying tricolor peak-hold meters and clip indicators. You can disable the peak-hold function entirely, or you can set it to infinite hold or 6-second hold. The same options are available for the clip indicators. A Dry button lets you bypass the FX Machine section to monitor the unprocessed sound, and a second button allows you to activate the real-time dithering function. Dithering options include 24, 20, 16, and 8 bit.

The right side of the Master View window is dedicated to *Spark*'s FX Machine, a powerful yet intuitive real-time effects-routing matrix. The FX Machine can accommodate as many as four parallel audio streams (derived from a mono or stereo file), with up to five plug-ins per stream. Furthermore, you can split or merge the streams and copy any plug-in to multiple locations. The processing possibilities are truly mind-boggling.

For example, you can use a filter plug-in to divide the signal into four frequency bands, then process the bands separately with a compressor to accomplish frequency-based compression. But that's just one application; you can also process an audio file with two stereo streams, each starting with the *Grainalizer*. The top stream could pass through the *Reverb*, while the bottom stream bypasses it. The audio with reverb could then pass through *ResFilter* while the dry audio enters the *Delay* plug-in. Finally, you could process both streams with EQ before summing them at the output. This is a fairly simple example with only two streams; things can easily get much more complex. The key is that you can do all of this in real time, experimenting with different settings as you go along.

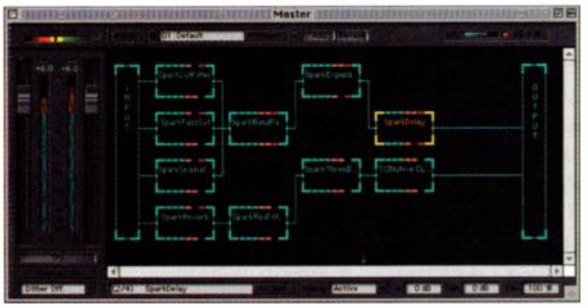


FIG. 5: The Master View window consists of two parts: the Master section for monitoring and adjusting levels and the FX Machine for processing with plug-ins.

Even if you don't use all 20 plug-in slots (when was the last time you needed 20 plug-ins cranking at the same time?), the FX Machine's flexible matrix design makes it easy to experiment with multiple effects. After you add a plug-in to one of the slots, you can drag it to another slot. Double-clicking a slot opens its plug-in so you can change settings at any time. The program's Automatic Wiring feature connects the plug-ins' inputs and outputs to the FX Machine so you don't have to draw the connections. You can also save FX Machine configurations as presets or in banks of eight.

With multiple streams passing through different plug-ins, the phase relations between the left and right channels can quickly get out of whack. (Relatively small phase discrepancies induced by some effects begin to add up when using large numbers of plug-ins.) To address this situation, the Master section includes an unusual multicolor Correlation meter that indicates the phase relationship between the channels. The Correlation meter can provide you with valuable information on such issues as mono compatibility and seriously out-of-phase signals. Furthermore, by turning individual plug-ins on and off, you can use the Correlation meter to isolate a particularly bad offender.

In addition, the colorful segmented border around each slot is more than just decoration. It's actually a small set of input meters for each plug-in, so you can keep track of the plug-in levels as you accumulate effects. A CPU meter in the upper right corner lets you know how much processing power is being used, and an Overload indicator warns you if you have loaded more plug-ins than your computer can handle. Keep in mind that few if any CPUs can handle a full load of plug-ins without triggering the Overload warning. On my 300 MHz Mac G3, for instance, I can load "only" about ten plug-ins without triggering a warning. (Some plug-ins require more processing than others, so your limits may vary.)

GIVE AND TAKE

Whether or not *Spark* is the program for you depends mainly on the nature of your audio-editing work. This program is not a one-stop solution for sound design or post-production, and if you spend most of your time fixing and tweaking waveforms at the molecular level, *Spark*'s modest editing toolkit will probably disappoint you. However, the program's strengths could nevertheless endear it to sound designers, who often are accustomed to using more than one program to accomplish their goals.

If your work focuses on CD mastering or effects processing, *Spark* is an excellent choice. Especially noteworthy are its FX Machine section, high-quality plug-ins, and other real-time nooks and crannies. So if combining real-time effects is your cup of tea, *Spark* might truly ignite your creativity.

Associate Editor **David Rubin** *lives and works in the Los Angeles area.*

TC WORKS
Spark 1.01 stereo audio editor
$499

FEATURES ■■■
EASE OF USE ■■■■
DOCUMENTATION ■■
VALUE ■■■
1 2 3 4 5

PROS: Supports 24-bit, 96 kHz audio; ASIO-compatible hardware; and VST plug-ins. Real-time resampling, dithering, and time compression. Includes 11 real-time VST plug-ins. Flexible handling of multiple effects. Good sampler support. CD-burning capability.

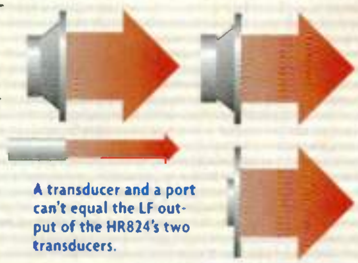
CONS: Limited number of editing tools and commands. No SMPTE support. Single-level Undo. Inadequate marker implementation. Browser View window is awkward to reconfigure. Poor documentation.

CIRCLE #440 ON READER SERVICE CARD

NO WONDER THE HR824 HAS THE BEST BASS RESPONSE OF ANY 8" MONITOR. IT'S REALLY A 12" MONITOR IN DISGUISE.

TIGHT, RESPONSIVE BASS FLAT DOWN TO 39HZ. Reviewers and owner's warranty card responses are unanimous: The HR824 has the most accurate bass they've ever heard from an 8-inch monitor. And bass **quality** is as astonishing as the quantity. Fast low frequency transients like kick drum slaps and electric bass notes have a crisp, articulation that makes other monitors sound like mush.

ANOTHER TRANSDUCER INSTEAD OF A PORT. The more LF transducer cone area a speaker has, the more bass it can produce. But a huge low frequency transducer isn't an option on a compact near field monitor.

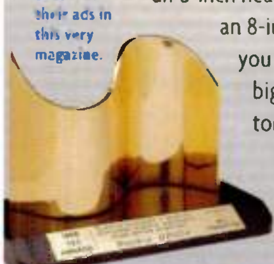


A transducer and a port can't equal the LF output of the HR824's two transducers.

To augment primary bass output, other monitors resort to using ducted ports that can convert cone movement into extra low frequency air movement. But for optimal output, a ducted port needs to have **the same area** as the low frequency transducer — an 8-inch near field monitor would need an 8-inch vent. Needless to say, you haven't seen any vents this big on other near field monitors. When vent size is reduced, bass output is compromised. And, forcing a lot of energy out of small ports can create audible wheezing and whooshing.

Instead, the HR824 adds a large passive transducer with the cone area of another 8-inch woofer. This ultra-rigid, honeycomb laminate piston tightly couples with the HR824's active bass transducer. With a combined cone area greater than a single 12-inch woofer, you get exceptionally extended bass without port noise complaint.

Last fall we won the pro audio industry's coveted TEC Award for best near field monitor. Modesty prevents us from listing the impressive field of competitors but you'll probably encounter their ads in this very magazine.



MASSIVE POWER THAT WOULD PROBABLY POP A PASSIVE MONITOR. Punching out crisp bass requires a lotta watts. The FR Series high-current bass amplifier module inside the HR824 delivers a solid 150 watts of power with peak output in excess of 250 watts (plus another 100 watts for mid and treble). That's significantly more than any other 8-inch active monitor. Moreover, the HR824's servo coupling and ultra-short signal path put that power to work far more effectively than a passive monitor and a 250-watt stereo amp could.

PART OF A TIGHTLY-INTEGRATED SYSTEM. Our servo bass system is only one contributing factor to the HR824's amazing accuracy.

Internal power amplifiers are "fed" by phase-accurate, low distortion electronic circuitry instead of a crude coil-and-capacitor passive crossover. The HR824's proprietary logarithmic wave guide not only widens treble dispersion but also smooths the midrange transition between high and low-frequency transducers. Thanks to the wave guide's flaring design, the HF transducer's output is **acoustically the same diameter** as the LF transducer's at the critical 3500Hz crossover point.

Rear view: The HR824's electronics conceal an ultra-rigid, honeycomb composite passive transducer.

The HR824's LF transducer even contributes to midrange accuracy. In many monitors, woofer cone harmonic vibrations bounce around inside the enclosure and then exit through the thin woofer cone. The result: smeared imaging and muddled details. Instead of a chintzy chunk of fluff, the HR824's enclosure is utterly packed with high- density absorbent foam. Cone vibrations go in, but they don't come back out.

DON'T SKIMP. It's amazing how many studio owners will mortgage the farm for money-is-no-object, esoteric microphones... and then monitor on cheap, passive loudspeakers. If you aren't using *some* brand of ACTIVE near field monitors you're seriously compromising your creative product.

HEARING IS BELIEVING. We urge you to visit your nearest Mackie Designs Dealer and carefully audition all of their active monitors with some demanding, bass-rich program material. Judge our claims (and those of our competitors) for yourself. We think you'll agree that the HR824 is truly the best of the best.

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KORG

ELECTRIBE ER-1 and EA-1

A traditional synth maker starts hanging out at dance clubs.

By Julian Colbeck

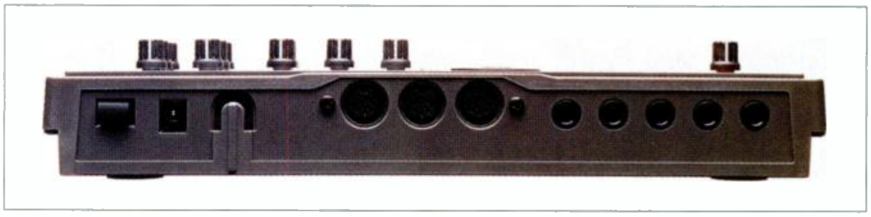

FIG. 2: The ER-1 is a stereo device with MIDI ports, as you would expect with a drum machine. But don't overlook its audio inputs, which are unusual for a beatbox. A front-panel button opens and closes the audio-input circuit, which allows you to use two channels of incoming audio as parts.

So-called groove machines are the hot-ticket item of the late 1990s. For the moment, at least, it is simply not enough to offer a synthesizer, or a drum machine, or even a sampler; dance-oriented electronica is the name of the game. Korg, best known for its classy synths and workstations, has been late in recognizing the revolution. But at the 1999 Winter NAMM show, the company enthusiastically donned a baseball cap, pierced its eyebrows, and launched the beknobbed Electribe ER-1 rhythm synthesizer and the EA-1 analog-modeling synthesizer.

Although they are quite separate in their range and capabilities, the physical-modeling-based Electribes clearly work as a pair: the red ER-1 provides the beats, and the blue EA-1 generates all the other squeaky bits.

FAMILY CHARACTERISTICS

The two units are each the size and shape of a small phone book, and their panels are smothered with a dozen or so knobs and 20 or 30 rubbery, semiopaque buttons that glow a rich Odeonic pink when active—very cool.

The look is superb, and the construction quality seems adequate for groove on the move—provided you're reasonably careful. (The anodized aluminum casing appears prone to scuff marks.) Each Electribe has a 3-digit LED screen and a free-flowing data wheel, and both come with wall-wart power supplies.

The MIDI implementation of the two units is good, with two noteworthy exceptions: the Electribes recognize neither MIDI Velocity nor Aftertouch. Less important, they do not transmit and receive the Delay Type and Motion Sequence parameters. Otherwise, they transmit and can receive every parameter and knob adjustment using MIDI Non-Registered Parameter Number (NRPN) messages. As a result, real-time manipulation of internal sounds can be recorded using an external sequencer. Pitch Bend, Bank Select (CC 0 and 32), Data Entry, and Reset All controllers are all transmitted and recognized, as are Program Changes and SysEx dumps and loads.

From here on, though, the ER-1 and EA-1 operate differently from each other and so must be looked at as different animals.

ER-1: OLD IS NEW

The name of Steinberg's *ReBirth* software is an accurate, albeit curious, analogy for understanding the ER-1. The irony, of course, is that *ReBirth* is a software approximation of early 1980s Roland hardware synth/beatboxes. In some ways, then, we have come full circle.

What you get with the ER-1 is a wealth of squeaky, squelchy drum and percussion sounds that are employed in 256 4-bar (maximum) patterns. Both sounds and patterns are freely tweakable on the fly using the knobs for sound changes and the buttons for pattern changes.

The ER-1 is not laid out like a conventional drum machine. The unit has five basic sections (see **Fig. 1**). The Common section includes the Master Volume knob, Audio In/Thru key (which lets you use external audio inputs as parts), LED screen, data wheel, cursor keys, Write button (for saving settings), peak LED indicator, and beat LED (which flashes on each quarter note to indicate the tempo). The mode buttons—Pattern, Song, Global, and MIDI—are also included in this section. Above the mode buttons is a matrix that

FIG. 1: Although the Electribe ER-1 certainly qualifies as a drum machine, it's more than just an old-fashioned beatbox. Designed specifically for live performance of contemporary electronica/dance music, it is more like a real-time percussion synthesizer integrated with a loop-oriented pattern sequencer.

FIG. 3: The EA-1 uses physical modeling to emulate some of the simpler classic analog synthesizers. Like the ER-1, it is aimed at the live dance/electronica market and integrates a real-time synth with a loop-oriented pattern sequencer.

beat of the bar. Activate buttons 1, 5, 9, and 13, and the currently selected sound will trigger on beats 1 and 3 of a 2-bar pattern, or on beats 1, 2, 3, and 4 of a higher-geared, 1-bar pattern.

This tried-and-tested, grid-style input method will be instantly familiar to anyone who has used a Boss rhythm machine or Roland TR-series beatbox in the past decade or so. The Select keys are associated with two rows of LEDs—four green and four red. The green LEDs indicate the location within the currently playing rhythm pattern; the red LEDs indicate the location within the rhythm pattern selected with the Step keys.

That's all fine, but the Select keys also have another, even better use: when the Pattern Set function is on, these keys let you access four different groups of 16 patterns. So you can set up your favorite 64 patterns to be activated using a combination of 16 Step keys and four Select-key positions. This handy live-performance feature is sure to be popular with DJs.

Unless you're happy flipping back and forth between patterns, though, you'll want to string patterns together to form a Song. Song mode gives you one final chance to do real-time tweaking, which Korg calls "event recording." You can mute and solo individual parts, you can record knob movements—including recording more than one parameter, unlike in Pattern mode—and you can call up (and thereafter tweak) different parts within the patterns as your song plays back.

The ER-1's audio inputs are what set it apart from a regular drum machine (see **Fig. 2**). This handy feature lets you patch in a CD player, a loop of scratchy old vinyl from your sampler, or whatever else suits you. The ER-1 is not a sampler, of course; it simply allows two external audio signals (or one stereo signal) to be used as parts in the same way you use drum parts. Pressing an Audio instrument key opens and closes a gate, so you can turn the external signal on and off in rhythm manually, and you can automate this ability using the sequencer. You also can pan the signals and route them through the effects.

Note that this feature gates the audio, rather than retriggering the sound; other than the amplifier and effects sections, the signal is not passed through the synthesis features. The Delay Depth and Time knobs work especially well on audio signals; rotating the Time knob produces a convincing scratch effect.

The Electribe units are primarily intended for live performance, and the ER-1 is at its best in this venue. Remixers looking for instant gratification should love the ER-1's dramatic range of tweakable rhythmic sounds, controls, and patterns. Electronic musicians who are more studio-oriented can also put the unit to good use, but if you're just looking for modern sounds in a modern context, be sure you understand the ER-1's limitations before taking the plunge.

EA-1: ROUGHLY RETRO

The EA-1 is essentially a simple, duophonic synthesizer with fancy sequencing facilities. Because of its simplicity, its similarities to the ER-1, and the familiarity of its components, I will discuss it in less depth than I did the ER-1.

The EA-1 is set out much like its drum-oriented partner, with a small LED screen, a free-flowing data wheel, and buttons and knobs for modifying sounds and for programming patterns and songs (see **Fig. 3**). And like the ER-1, it looks superb and is great fun to operate.

Based on Korg's Z-1 and Prophecy analog-modeling technology, the EA-1 provides independent access to two synthesizer parts, each of which can use dual oscillators, with filtering, envelope shaping, effects, and such. Two parts? It's not a lot, and you can best judge the instrument as a purveyor of bass, arpeggio, effects, and lead lines—all of which have a deliberately synthetic hue. Obviously, polyphonic pads are out of the question.

BASIC USE

Like the ER-1, the EA-1 is full of patterns, each of which contains 2-part, 4-bar (maximum) grooves with sounds twisted and tweaked anew. Spin through the presets and you'll hear classic techno and house stuff by the yard. At this point you can simply toggle between Part 1 and Part 2 and tweak the current sounds in real time using the oscillator, filter knobs, and so on.

You can control tempo using the

Electribe ER-1 Specifications

Synthesis Type	physical modeling; (4) PCM samples: crash cymbal, open and closed hi-hat, handclap
Memory Capacity (RAM)	256 patterns, 16 songs
Number of Parts	(4) synthesizer, (4) PCM, (2) Audio In, (1) Accent
Sequencer (pattern)	max. 64 steps/part; 1 parameter/part, 64 events; Motion Sequence
Sequencer (song)	max. 256 patterns/song; max. 35,700 events for event recording
Effects	delay (normal, Motion Sequence, tempo delay)
Audio Inputs	(2) ¼" unbalanced mono (–10 dBu)
Audio Outputs	(2) ¼" unbalanced mono (–10 dBu); (1) ¼" stereo headphone (32Ω)
Other Ports	MIDI In, Out, Thru
Power Supply	9 VDC adapter
Dimensions	11.8" (W) x 8.9" (D) x 2.1" (H)
Weight	2.75 lbs.

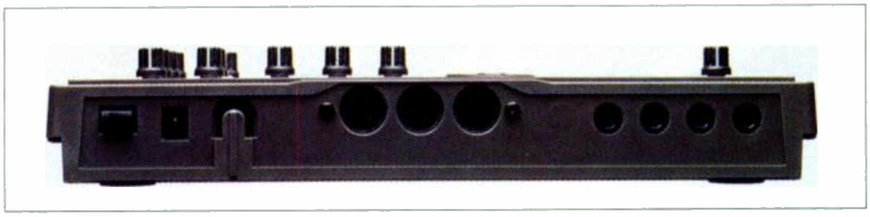

FIG. 4: The EA-1's rear panel holds no big surprises, just the usual audio and MIDI ports. As with old-time analog synths, the signal appearing at its audio input can be processed through the synthesis architecture.

wheel or the tap-tempo button, or you can sync the unit to an external device. You can add or subtract "notes" to or from the current pattern by activating or deactivating any of the rubber pads. The Motion Sequence feature found in the ER-1 is here, too.

SOUND CONTROL

The EA-1's synthesizer engine is presented in a style similar to a classic analog synth's. The oscillator panel gives you a choice of three waveforms (triangle, sawtooth, or square) per oscillator, plus ring mod, sync, and oscillator balance. Portamento control is shared between the oscillators. The pitch of oscillator 2 can be set independently of oscillator 1 so that you can produce 2-part harmonies on each part.

The Decimator is an interesting feature that permits you to alter the sampling rate of oscillator 1 as a function of the frequency of oscillator 2, providing a source of lo-fi sounds. To use the Decimator, you select a fixed frequency for oscillator 2; the lower the frequency of oscillator 2, the lower the sampling rate of oscillator 1—and the result can be some nasty sounds. You hear only the Decimator's output, not the unprocessed outputs of the individual oscillators. The feature is effective when used with a triangle wave or the audio input, but not with saw or square waves.

As with the ER-1, you can route an external audio signal into the synth (see **Fig. 4**). However, unlike with the ER-1's audio inputs, the signal appearing at the EA-1's single audio input can be used as an oscillator waveform and processed through the whole synthesis architecture.

The EA-1's sounds are bright and dynamic, but I didn't find walls crumbling when it came to bass lines. The lowpass filter section has controls for cutoff frequency and resonance, as well as for envelope amount and decay—hardly comprehensive. Although the EA-1 offers more in the way of recordable real-time control over sounds, sonically you can achieve ten times more on a 20-year-old Korg MS-20. Then again, the EA-1 costs only $499; even a used

Electribe EA-1 Specifications

Synthesis Type	physical modeling
Memory Capacity (RAM)	256 patterns, 16 songs
Polyphonic Voices	2
Multitimbral Parts	1
Sequencer (pattern)	max. 64 steps/part; 1 parameter/part, 64 events; Motion Sequence
Sequencer (song)	max. 256 patterns/song; max. 65,500 events for event recording
Effects	distortion; tempo delay; chorus/flanger
Audio Inputs	(1) ¼" unbalanced mono (−10 dBu)
Audio Outputs	(2) ¼" unbalanced mono (−10 dBu); (1) ¼" stereo headphone (32Ω)
Other Ports	MIDI In, Out, Thru
Power Supply	9 VDC adapter
Dimensions	11.8" (W) x 8.9" (D) x 2.1" (H)
Weight	2.75 lbs.


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Marshall

MS-20 will set you back more than that.

The amplifier section offers level control and distortion (triggered by a button), and the effects section offers depth, time, and effects type (essentially delay or chorus). I have no problem with this clutch of controls: it provides all you'll need for many types of dance-music applications. They're fine if you just want to come up with a catchy little line, sample it, and move on.

The instant-fun factor is there, without question, no matter what sort of music you're into. But if you have much prior knowledge of synthesis and want to use it for in-depth sound manipulation, or if you need specific levels of control, the EA-1 will probably only frustrate you. That statement is by no means intended as a criticism of the EA-1 or as a slur on the dance scene for which it has been primarily designed. I think that Korg is absolutely right to make the EA-1 as wild and dynamic as can be. My point is that just because the unit makes excellent squeaky, squelchy sounds, you shouldn't think that you are buying the level of control of, say, a Synthi 100 for only a few hundred bucks.

What you're getting with the EA-1 is a well-tuned performance machine. Therefore, long-term satisfaction with the unit is most likely for die-hard dance-music producers. Electronic musicians in need of more in-depth synthesis should probably look elsewhere.

KORG
Electribe ER-1 rhythm synth
Electribe EA-1 analog-modeling synth
$499 each

FEATURES ■■■

EASE OF USE ■■■■

QUALITY OF SOUNDS ■■■▌

VALUE ■■■▌

1 2 3 4 5

PROS: Instant, hands-on access to the latest cutting-edge grooves. Textures sound authentically complex without much fuss. Great fun. Eye-catching design. Almost complete MIDI control using NRPNs.

CONS: No disk drive. No battery-power option. Limited recordability of real-time sound changes in ER-1's Pattern mode. Low polyphony and overly simple synth architecture on EA-1.

CIRCLE #441 ON READER SERVICE CARD

THE ELECTRIBE VIBE

With the ER-1 and EA-1, Korg has gone for full-on flash and live-performance immediacy over technical completeness. There's nothing wrong with that approach; in fact, it's absolutely right if traditional musical-instrument companies are going to capitalize on the burgeoning DJ and remix market.

Given Korg's approach, I think that not offering a battery-power option is an unfortunate oversight. (I'd love to annoy fellow airplane passengers with either of these units.) Otherwise, however, both Electribes appear to be well targeted for the DJ or remixer who does not want to do in-depth tweaking.

Julian Colbeck *is retired from active duty on the road as a keyboardist, deeming running the U.S. branch of Keyfax Software/Hardware a more dignified midlife occupation.*

DIGITAL AUDIO INNOVATIONS

SPACE STATION PRO 1.41 (WIN/DOS)

Turn your aging PC into a sampling workhorse.

By Allan Metts

Remember DOS? In this era of window-based, multitasking operating systems, it's hard to imagine that anyone would write a DOS-based application. But someone has. *Space Station Pro,* from Digital Audio Innovations, transforms your PC into a highly useful sampler, complete with audio recording, editing, and multitimbral playback capabilities.

Space Station Pro will run on your Windows 95/98–based system, if you don't mind that the program grabs exclusive use of the audio and MIDI devices it uses. *Space Station Pro* will not run under Windows NT, however. The program can eke out better performance in a native DOS environment because it doesn't have to deal with Windows' extra overhead. In fact, it can even run on an old 386 (although you'll need a Pentium PC for CD-quality output). So if you have an aging computer lying around, *Space Station Pro* could easily transform it into a useful studio tool.

Installing *Space Station Pro* is a bit tricky, but I managed to get through it. (Admittedly, I'm a little out of practice when it comes to properly configuring a DOS machine.) The program won't launch at all unless it can find a supported mouse and sound card, and it works only with Creative Labs Sound Blaster 16 or AWE audio cards. (Support for Aardvark's 20/20 is under development.) Once my drivers were installed and configured correctly, I was fine.

Space Station Pro is copy protected, which means you first have to type in a license key, obtain a serial number from the program, and then contact Digital Audio Innovations with both the key and serial number. The company will give you a security code, which you type in to unlock the program. Even then you're not home free: *Space Station Pro* prompts you for the installation CD-ROM every time you launch it. Failure to produce the disc causes the program to boot in demo mode (which means you can't load or save anything).

I understand Digital Audio Innovations' desire to protect its investment from software pirates, but I found this copy-protection scheme a royal pain. What's more, the correct security code can change between installations, so you'll have to contact the company with a new number if you install it on a different hard drive or reformat your existing one. I feel that the program should allow full use for at least a few days. With the present scheme, a hard-disk crash outside of U.K. business hours leaves you dead in the water. (Digital Audio Innovations will, however, sell you a second license at a nominal fee for use on a backup drive.)

FIG. 1: One-third of *Space Station Pro*'s user interface stays constant as you move about the program. The keyboard control, transport buttons, and navigation controls are always there when you need them.

FLYING THROUGH SPACE

Once you have it installed and running, you'll see that *Space Station Pro* is easy to navigate. Roughly one-third of the screen—the area where controls and status information are displayed—stays the same as you move about the program. The rest of the screen changes to reflect the tasks at hand.

Moving from the bottom of the screen to the top, you'll first find a keyboard display. Indicators for selected keys and keyboard ranges help with sample mapping (see **Fig. 1**); you can audition notes and select keyboard ranges by clicking the keyboard. Also present are a display of the keyboard range and controls to set the current MIDI channel. (*Space Station Pro* is 16-part multitimbral.)

Just above the keyboard (to the left) is a set of transport, time-signature, and tempo controls, with status indicators that show SMPTE time and the current bar:beat:tick position. Wondering what these controls are doing in a sampler? As it turns out, *Space Station Pro* has MIDI sequencing features "under construction," and the time display will be a vital part of that interface when it becomes available. For the time being, these controls set things in motion for Loop Mode 2 sounds (described later). LFOs and envelopes also require a running transport when you sync them to MIDI.

Moving to the center of the lower screen area, you'll find a set of four buttons that can be enabled with the mouse or with dedicated function keys. These buttons do different things, depending on where you are in the program. In one screen, the buttons are replaced by a nifty MIDI analyzer, with activity meters for each MIDI channel and for several types of MIDI messages.

To the right of these "soft" buttons

For manufacturer information, please see Contact Sheet, p. 179.

are dedicated navigation controls that take you to specific parts of the program. This is, after all, a space station, so the program displays its screens according to Zones and Levels. Zones represent major categories of functions (Wave recording and editing, for example), and Levels represent the various screens within each category. Clicking the Ship button calls up a pictorial view of all Zones and Levels (see **Fig. 2**); clicking any of them takes you straight there. Each Zone has its own color that appears in the windows and buttons, making it easy to see where you are in the program.

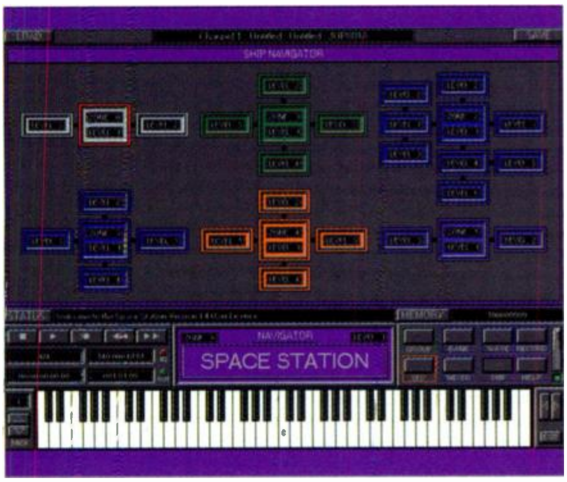

FIG. 2: This screen comes up when you click the Ship button. From here, you can get anywhere in the program with one click of the mouse.

Rounding out the set of common controls are Load and Save buttons as well as indicators for MIDI activity, memory usage, and system status. Also present are controls for synching, clicking, punching, and looping. *Space Station Pro* can sync to MIDI Clock and MIDI Time Code messages. You can start and stop audio recording automatically at any bar:beat:tick boundary.

Space Station Pro's sound structure is remarkably simple. Waves (samples) are combined into Banks, and Banks are combined into Groups. A Group is a collection of up to 16 Banks (one for each MIDI channel). I question *Space Station Pro*'s use of the "Bank" moniker, however, because most musicians think of a bank as a collection of synth programs. Instead, a *Space Station Pro* Bank is closer to what I would call a "patch" or an "instrument."

There are Zones for Waves, Banks, and Groups, and plenty of functions for each of them. We'll start with Waves and work our way up to a complete multitimbral setup.

SURF'S UP

Space Station Pro has a capable set of Wave editing and recording features. When recording, you can assign Waves to keys without ever leaving the recording screen. Just select a key, click Record, and move to the next one. I found this to be a very efficient way to work.

The program places several recording options at your disposal. You can choose any of the input sources on your sound card, a sample rate of up to 44.1 kHz, mono or stereo, and a triggering method. The triggering method determines how recording starts and stops; choices include Normal (you press a button), Mouse (you click the mouse), an audio level threshold, or a bar:beat:tick range (that is, punch in and out). A nice touch is the ability to record *Space Station Pro*'s output as an input source. Resampling, anyone?

Once you've recorded or loaded your Waves, you'll probably want to edit them in the Wave Editor (see **Fig. 3**). You won't find any esoteric processing features here—the Effects Rack and Processor Rack are nonfunctional in this version of the program—but you will find the tools you need to do basic editing. You can cut, copy, and paste Wave sections; insert silence; mix audio material; and create fade-ins and fade-outs.

The zooming, viewing, and selection tools allow precise placement of your editing selection's start and end points. What's more, the auditioning tools let you hear the entire Wave, a selected portion, or the sections just before or after a selection. The program can display time in seconds, samples, bytes, SMPTE time, or bars: beats:ticks, with grid markings placed on the Wave display at user-selectable intervals.

The Bank Zone is where you'll head when it's time to apply keyboard zones, envelopes, LFOs, and looping to your collection of loaded Waves. Each Wave in a Bank also has independent settings for level, pan, tuning, and sustain-pedal response.

Space Station Pro's loop editor is a good tool, with one display showing the entire Wave and another showing the loop's end right up against its start. You adjust loop points by dragging the start and end points or by incrementing and decrementing the numeric displays. (The latter option is best for fine-tuning.) You can save eight sets of loop points as you work and easily switch among them; only one set gets saved when you leave the window. You can also choose whether a loop moves from start to end or bidirectionally.

Finally, a handy Auto-Loop window restricts your loop to those points where a crossover or zero value occurs (you decide which one). All in all, I had no trouble creating respectable-sounding sample loops with this set of tools.

If you're into sampled dance grooves, Loop Mode 2 is for you. It lets you assemble a 1-, 2-, or 4-bar loop by loading Waves into locations within the loop. Start the transport, press a key, and *Space Station Pro* plays the samples in perfect time (provided the tempo of your sampled groove matches the one specified on the transport bar).

SPACE MAPPING

If you didn't assign Waves to every root note as you recorded or loaded your samples, you'll need to use *Space Station*

FIG. 3: *Space Station Pro*'s Wave Editor is a capable one, providing the tools necessary for basic audio editing.

Pro's keyboard-mapping feature. Mapping Waves to notes is relatively painless: just choose a Wave and open the mapping screen, then specify the root note of the Wave and indicate the high and low key ranges. *Space Station Pro* does the rest.

Though *Space Station Pro* supports Velocity switching, you can switch only between two Waves. Alternatively, you can set up a crossfade, which lets you fade gradually from one Wave to the next as you move up and down the keyboard. Unfortunately, you can never have more than two Waves assigned to one key (within a given MIDI channel). Also missing is the ability to set up mutually exclusive groups, so your open hi-hat always gets cut off when the closed hi-hat plays, for example.

Space Station Pro provides a very capable screen for creating amplitude, pitch, and panning envelopes (see **Fig. 4**). You can draw each of these envelopes with the mouse, and plenty of tools help you get just the shape you need. Each envelope appears in its own color directly on top of the Wave display, so it's easy to visualize the sound you're creating. You can hide any envelope—or the Wave display itself—if things get too crowded onscreen.

Although only the attack and release points of an envelope are MIDI controllable, you can use a nearly unlimited number of breakpoints to create as exotic a shape as you want. If you don't want your envelope to remain at a fixed sustain point, you can add a loop starting point; the envelope will cycle back to the loop start when it reaches the sustain point.

Envelopes can take a constant amount of time to execute, but they can also speed up or slow down with the Wave's pitch. You can also link the envelopes to *Space Station Pro*'s tempo setting, which causes the envelopes to move faster as the tempo increases. An envelope's attack and release times can be set to increase or decrease according to Velocity. You can also set an overall amplitude-sensitivity value for each channel. And, finally, envelopes can be saved to disk and reused as needed.

I don't think I've ever seen a more capable tool for creating envelopes, but the astute reader will notice one thing missing from this discussion: the program has no filter envelopes. With up to 768 envelopes per Bank, it would be very difficult to know which envelope was controlling the filter, but perhaps a solution could be found. *Space Station Pro* offers two 24 dB resonant filters per MIDI channel. They are controllable both internally and externally through MIDI continuous controllers. While you can't trigger a filter sweep with every keypress, you could route an LFO (described shortly) to one of these MIDI controllers.

The filters themselves offer lowpass, highpass, bandpass, band-reject, and "additive" modes, with controls for cutoff frequency and resonance; additive mode combines the original and filtered

Space Station Pro
Minimum System Requirements
'386/40 MHz; 8 MB RAM; DOS or Win 95/98; Creative Labs Sound Blaster 16, AWE32, or AWE64 sound card

signals. You can also run the filters in series or in parallel. They sound fine to me, although I could hear a fair amount of "stepping" with resonant filter sweeps. Because a single 7-bit MIDI continuous controller has only 128 possible values, the stepping doesn't surprise me.

LFOs IN SPACE

In addition to its extensive envelope control, *Space Station Pro* really delivers on its LFO capabilities. The program lets you specify up to six LFOs in a Bank, and any of them can modulate pitch, amplitude, panning, or other LFOs. The LFOs' output can even be sent to external MIDI devices as continuous controller data. A mapping screen allows you to draw modulation routings and also lets you establish connections with external MIDI controllers for real-time control.

LFO shapes and LFO connection maps can be saved separately as disk files. *Space Station Pro* comes with several examples of each. If you don't like the LFO shapes provided, you can draw your own in the LFO designer window. Now *that's* a feature I haven't seen before!

As you create or load Banks in *Space Station Pro*, the program begins to fill up your Group list. There are 16 Banks in a Group—one for each MIDI channel. Each Bank gets its own pitch-offset and tuning control. To try out *Space Station Pro*'s multitimbral abilities, I

DIGITAL AUDIO INNOVATIONS
Space Station Pro 1.41 software sampler
$100

FEATURES
EASE OF USE
AUDIO QUALITY
VALUE
1 2 3 4 5

PROS: DOS provides maximum polyphony on a minimal machine. Advanced envelope- and LFO-editing capabilities. Excellent MIDI support.
CONS: No filters. Few built-in effects. Weak hardware support. Included sounds are not up to professional standards. Cumbersome copy protection.
CIRCLE #442 ON READER SERVICE CARD

loaded up a bunch of Banks and played some rather dense sequences into the program from an external sequencer. Everything sounded quite acceptable.

So just how much polyphony do you get with *Space Station Pro?* The program comes with a built-in test for this. Under DOS, I was able to get 117 voices on my 400 MHz Pentium II. According to the manual, you could expect about half that many voices on a 166 MHz Pentium. These numbers are quite respectable for an all-software sampler.

FIG. 4: The system's advanced envelope-editing capabilities allow you to create complex envelopes for pitch, panning, and amplitude.

MIXING MAYHEM

Space Station Pro includes several mixers. There's the poorly named Output Mixer, which manages all the inputs and outputs of the sound card itself. Output Mixer is similar to the controls that a Windows wave driver gives you, with volume controls for the line and mic inputs, CD Audio, and so forth. Because *Space Station Pro* is a DOS application, controls like these have to be written into the program.

The Main Mixer (see **Fig. 5**) provides volume, pan, attack, and release faders for each of the 16 MIDI channels. Each fader can be locked to one or both of the Special faders, allowing you to move multiple controls simultaneously. There are mute buttons for each channel and a solo control for the currently selected channel. The faders all respond to MIDI control. Channel settings can be saved one channel at a time and restored individually or all at once.

If you need to generate additional MIDI messages, you can do so by creating User Mixers. Each User Mixer consists of 16 faders and 16 switches and is capable of transmitting almost any MIDI message on any MIDI channel. User Mixer 3 comes preconfigured to control the two filters on each MIDI channel. It also provides controls for a MIDI-based delay, which enables you to equip your sounds with some basic echo effects.

As in the Main Mixer, faders in a User Mixer can be locked to Special faders for group movement. (These faders also respond to MIDI control.) You can use four User Mixers simultaneously, and you can save them to disk.

FIG. 5: *Space Station Pro*'s Main Mixer gives you volume, pan, attack, and release controls for each MIDI channel. The faders can be grouped to prevent cumbersome mouse movements.

SPACE: THE FINAL FRONTIER...

Space Station Pro's documentation can be a bit cryptic at times, and it lacks a decent description of the MIDI-controlled filters and delay. The only printed documentation included is a 15-page installation guide; everything else appears in the programs context-sensitive help. HTML and PDF versions of the help files can be found on the installation CD, which is nice if you want everything in one place. This reference information was hard to follow, however. A well-written getting-started guide would be very useful.

Although the program is stable for the most part, I did keep the folks at Digital Audio Innovations busy by uncovering several bugs. A couple of these were serious; for example, I couldn't save Waves when running under DOS without my PC locking up, and I had to reboot each time I exited the program. The manufacturer claims that it has not been able to duplicate these problems and that they have not been reported by other users.

The program comes with numerous Groups, Banks, Waves, envelopes, and LFOs on the installation CD-ROM. A wide variety of sounds are thrown in, including acoustic instruments, synth waves, special effects, and dance grooves. Most of the supplied instruments are only 22 kHz samples, however, and many of the Banks were loaded with only one sample per octave. This surprised me because I had no trouble loading lots of 44.1 kHz samples. Unfortunately, *Space Station Pro* loads only WAV and *Space Station Pro*–format files, although converters for Akai and Roland formats are provided.

Overall, I wasn't thrilled with the sound quality of this program. Without filter envelopes and a good set of effects, I had a hard time creating sounds that knocked my socks off. Throw in the 22 kHz samples included on the CD and output them to a $99 Sound Blaster, and you end up with sounds that say, well, "ho-hum." Perhaps if I had used the digital output on a Sound Blaster AWE64 Gold card, I would have felt differently. (Of course, including support for other sound cards would also be useful to many musicians.)

Space Station Pro does many things very well, but even with its low price tag, I can't really recommend it for serious professional use at this point. Digital Audio Innovations is heading in the right direction, though, by planning to support additional professional gear and more common sample formats. With a few more features, this could be a powerful music-making platform. (I would also love to see an alternative copy-protection scheme.)

In the meantime, you may be in luck if you have some cash and a dusty old 90 MHz Pentium PC sitting in the closet. *Space Station Pro* can help you put them both to good use.

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FIRST WE LISTEN

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Down and FS Up. We'll cover that feature later.

As you can see in **Figure 3**, the FREQue stacks two identical modules—ColOscil modules, to be precise—and includes some FREQue-specific features. At the far left are input-level meters labeled Mod and Music. The lower LED registers green for an input above –40 dB, and the upper shines yellow if you're around +2 dB. For better performance, DACS recommends staying between +2 dB to +12 dB at the input. However, there is no input-gain control. Depending on your source, you may need a decent preamp for your material. I had no complaints with introducing material slightly below +2 dB, but below 0 dB, the noise floor became an issue.

The cryptically labeled Weight and Edge knobs vary the spectral content at the output of the ring modulator. The two controls act on the music input, providing bass and treble filtering at 80 Hz and 8 kHz. A red button activates and deactivates the effect, but treatment is also bypassed when the knobs are centered. The effect is not as pronounced as you might imagine because it's entirely dependent on material at the Mod input. I tended to get a harsher quality with Edge boosted and more support on low end with Weight cranked.

The FREQue features two built-in sine/cosine wave oscillators as modulation sources. Activating them bypasses signals at the Modulator inputs. Each oscillator has broad and fine-tune

BIG BRIAR
Moogerfooger MF-102 ring modulator
$299

FEATURES
EASE OF USE
AUDIO QUALITY
VALUE
1 2 3 4 5

PROS: Warm sound. Built-in carrier oscillator. Selectable LFO. Footswitch. Mix pot. Four expression-pedal inputs. Low price.
CONS: Wood side panels may get scuffed when used on floor. LFO can't modulate external carriers.

CIRCLE #443 ON READER SERVICE CARD

controls as well as a four-range switch that selects among 0.1 Hz to 28.5 kHz, 5 Hz to 153 Hz, 30 Hz to 1.3 kHz, and 111 Hz to 16.5 kHz. You get desirable LFO frequencies for gating and tremolo effects, as well as a warm low end, with clean mid and high frequencies in the audible range. In the box that I used, the output of Osc 2 tended to be hotter than Osc 1, but I believe this was an anomaly in my unit.

WHICH ROUTE?

FREQue includes a front-panel routing switch that disconnects Osc 1 and sends Osc 2 to the Mod inputs of the bottom module. This lovely feature allows Osc 2 to modulate both channels, which is useful for controlling stereo effects from one knob.

Next to the Osc 1 switch is a frequency modulation (FM) switch. When engaged, Osc 1 modulates the frequency of Osc 2. A knob to its right functions as a mix knob, increasing modulation as you turn it clockwise. In general, this setting tends to impart a grittier sound due to the increase in sidebands. A low setting can yield a vibrato or phasing effect.

The last router is the FREQue button, which engages Frequency Shift mode and deactivates the Osc 1 switch. This feature requires both Osc 1 and 2 to be activated and employs both outputs. The treatment shifts frequencies upward at FS Up and downward at FS Down by a fixed number of cycles per second. The degree of modulation is determined by Osc 2, which can be controlled internally or via CV. This makes a great subtle stereo treatment. The only downside is that my unit made an unpleasant click in the monitors when I pressed this button. (The manufacturer is currently addressing this problem.)

SPECTRAL LIFTOFF

As with the MF-102, most of my test source sounds came from a Minimoog, radio, guitar, bass, an Oberheim 2-Voice, and lots of sampled oddities. I punished the FREQue with sound, and it responded well. Generally its quality was less warm and more metallic than the MF-102's, but more radical and unpredictable given the doubling up of modulators and increased routings. Also, having CV inputs to control both modulating oscillators was a smart design choice. Those of you who own MIDI-to-CV converters will be able to finely control both oscillators from your sequencer. This is useful for phrasing (and for freeing up your hands, too).

Items on my wish list include a Mix control to vary the intensity of modulation at the outputs, peak-level meters, and a footswitch. Also, the unit has no power switch, though turning it on from a power strip didn't bother me.

DACS includes a concise manual that reviews all the functions of the FREQue. The company's Web site mentions downloadable application examples, setups, and sound files, but as of this writing none were there. The company has published *DACSData*, a useful newsletter that includes tips on drum, delay, vocal, and multiple modulation. Perhaps they'll move that information onto the Web.

FREQue Specifications

Audio Inputs	(2) ¼" unbalanced modulator; (2) ¼" unbalanced audio
Audio Outputs	(2) ¼" unbalanced main L/R; (2) ¼" unbalanced oscillator
Other Connections	(2) ¼" CV inputs
Internal Oscillators	(2) switchable 0.1 Hz to 28.5 kHz, 5 Hz to 153 Hz, 30 Hz to 1.3 kHz, or 111 Hz to 16.5 kHz
Filters	highpass (80 Hz cutoff); lowpass (8 kHz cutoff)
Indicators	(4) pairs of input-level LEDs
Frequency Response	20 Hz to 52 kHz (± 3 dB)
Signal-to-Noise Ratio	–82 dB
Dimensions	2U x 12" (D)
Weight	7 lbs.

BACK TO EARTH

Both the MF-102 and the FREQue excel as signal processors, and I would be hard-pressed to choose one box over the other. More than just a throwback to earlier days, both units include enhancements that make them quieter and more flexible instruments than their predecessors. I've also listened to lots of ring-modulator algorithms on synthesizers and computer workstations, and these boxes blow the digital versions out of the studio.

Although the $299 MF-102 costs a mere fraction of the $1,099 FREQue, the latter features two ring modulators with a larger set of modulation features and possibilities for stereo manipulation. Besides the cost factor, the main issues really are the interface and the character of the sound you get. The MF-102 has well-marked, clearly readable controls on a portable chassis. It's a high-class stompbox for guitarists and makes a great addition to any electronic-music studio. The FREQue is just that: a psychedelically clad anomaly that makes sounds which defy description. Markings are scant, however, so you might get a little irritated with the free-form approach that the designer has imposed on you. But you'll probably be the only kid on your block to own one.

If what you want is a warmer sound, you might lean in the direction of the MF-102. If you're looking for more edge, try out the FREQue. My vote is to let your budget point the way. You won't be disappointed with either choice.

DACS
FREQue dual ring modulator
$1,099 for standard unit
$1,295 for balanced unit
(upgrade kits available)

FEATURES	■■■■
EASE OF USE	■■■
SOUND QUALITY	■■■■
VALUE	■■■
	1 2 3 4 5

PROS: Wild sounds. Two channels for stereo effects. CV inputs. Built-in carrier oscillators with outputs. Frequency modulation. Frequency shifting.
CONS: Odd nomenclature. Controls are difficult to read. No peak-level meters. No footswitch. Clicking sound from FREQue switch. No power switch. Pricey.
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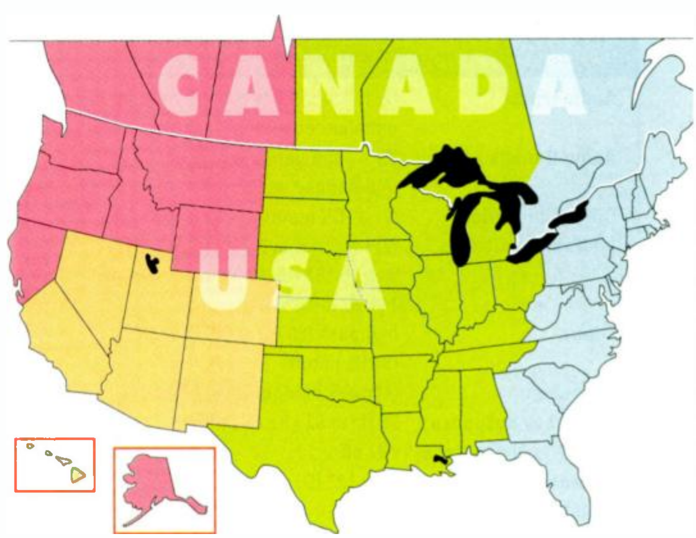

CONTACT SHEET

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MCDSP

FILTERBANK 1.04 (TDM/AUDIOSUITE)

A flexible, great-sounding EQ plug-in for Digidesign systems.

By Peter Freeman

The conventional wisdom seems to be that plug-ins, although good for many applications, just don't deliver great-sounding equalization and compression. It's not uncommon to see producers and engineers routing their Pro Tools tracks to vintage outboard EQs and compressors and then back into Pro Tools, rather than relying solely on plug-ins for these important tasks.

It's true that no plug-in is a perfect replacement for vintage outboard EQ and dynamics processors, but McDowell Signal Processing (McDSP) has made a significant step in that direction. Founded by one-time Digidesign employee Colin McDowell, McDSP is making its debut with the *FilterBank* equalizer, easily the most comprehensive EQ plug-in I've seen.

The program is available on a CD-ROM that contains both TDM and AudioSuite versions, though as of this writing it operates in real time only on the TDM platform. I tested the TDM version, which requires *Pro Tools* 4.0 or later software and Pro Tools PCI, 24, 24/Mix, or 24/PowerMix hardware. The software is compatible with Mackie's HUI as well as Digidesign's ProControl hardware controllers.

The informative, comprehensive, and well-organized electronic manual (which comes in both PDF and HTML formats) is packed with JPEG screen shots. Although I prefer paging through traditional paper documentation to reading a manual in a Web browser, I had no trouble learning to use *FilterBank* with these electronic documents.

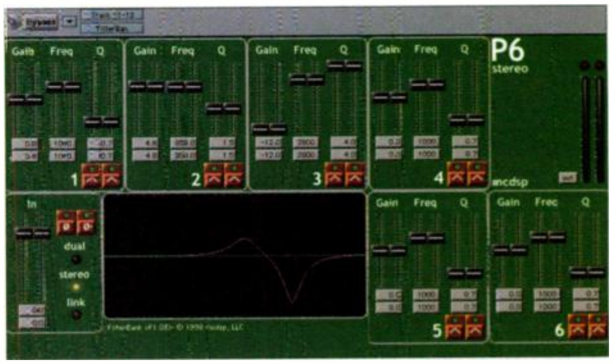


FIG. 2: ***FilterBank*'s P6 6-band parametric EQ configuration, shown here in stereo, offers more precision than you will need for most applications.**

SOFT SLIDERS

Looking at *FilterBank*'s design, it's immediately clear that this plug-in is unusually flexible. The software's high and low shelf EQ, parametric EQ, highpass and lowpass filters, and bandpass and band-reject filters are available in both stereo and mono versions and in 2-, 4-, and 6-band configurations. You can control frequency, gain, peak, slope, and dip (explained shortly), over a frequency range of 20 Hz to 21 kHz.

You get a minimum of 22 bands of EQ per DSP chip on a Pro Tools PCI system and between 42 and 48 bands on Mix and Mix-Plus systems, which is generous. McDSP says that the TDM version of *FilterBank* keeps the noise floor at –144 dB by calculating the values with 48-bit precision.

Sliders are the controller of choice in the *FilterBank* interface, along with bypass buttons that have been amusingly styled after '70s Sequential/Korg LED switches. When you apply equalization, a white line displaying the selected cut or boost curve appears in a Display Graph, which is a black rectangular window with a fixed blue horizontal "zero" line.

GETTING STARTED

After a fairly painless installation process (the program is copy protected, offering a choice of challenge-response or key-disk authorization), *FilterBank* is ready to go. As with any TDM EQ, using *FilterBank* in its most basic form is simply a matter of instantiating it and adjusting its frequency and gain controls to boost or cut the desired frequencies. *FilterBank* includes mono and stereo versions; instantiating it on a mono or stereo track in the host program (such as *Pro Tools* or a TDM-compatible digital audio sequencer) will bring up the appropriate version.

Thanks to the program's wide range of possible configurations, things get much more interesting from here on. The next order of business is choosing the right configuration for the material that you're equalizing. Nine EQ configurations are available (see the table "FilterBank EQ Configurations"), but I'll focus on the four that best show the plug-in's flexibility: E6, P6, F2, and B1.

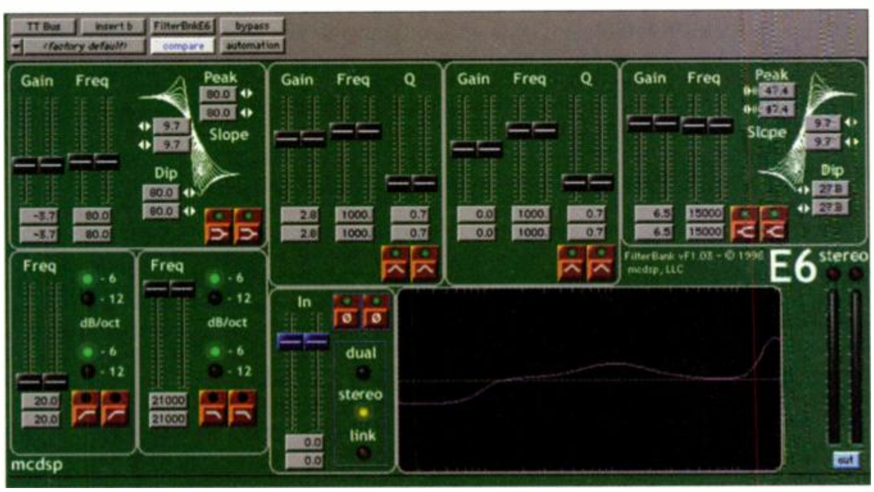
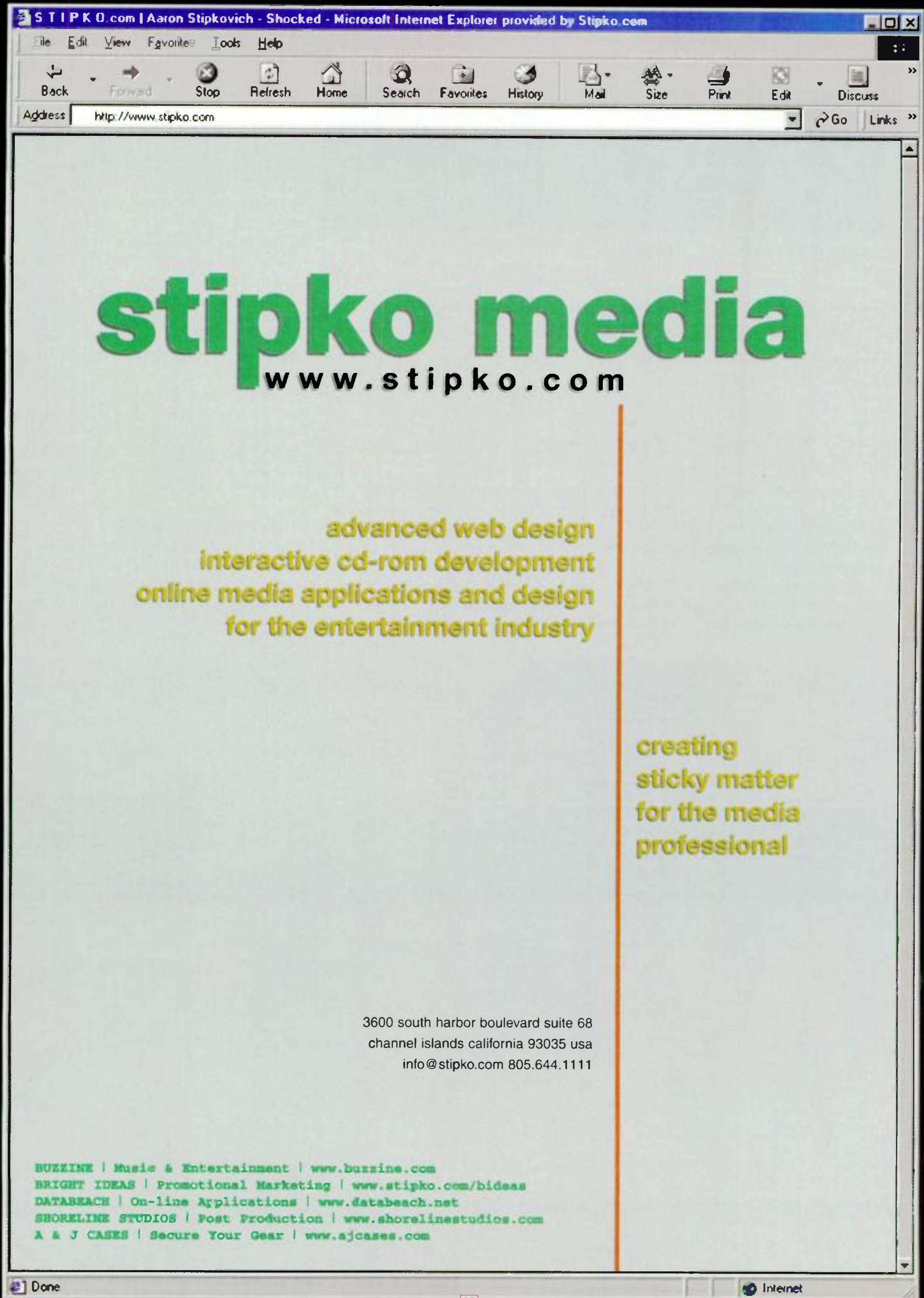


FIG. 1: You'll probably find the most uses for *FilterBank*'s E6 configuration, shown here in its stereo version.

COMPLEX CONFIGURATIONS

FilterBank's E6 configuration (see **Fig. 1**) is probably the most versatile and has the widest range of musical applications.

For manufacturer information, please see Contact Sheet, p. 179.



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FilterBank EQ Configurations

McDSP *FilterBank* includes nine combinations of EQ types. Mono and stereo versions are provided for all except B1, which is mono only. Detailed specs are available in chapter 3 of the online user manual (www.mcdsp.com/FilterBank/UserManual/FilterBankConfigurations.html#anchor105021).

CONFIGURATION	BANDS
E2	1 low shelf, 1 high shelf
E4	1 low shelf, 1 high shelf, 1 parametric, 1 highpass
E6	1 low shelf, 1 high shelf, 2 parametric, 1 highpass, 1 lowpass
F1	1 highpass
F2	1 highpass, 1 lowpass
P2	2 parametric
P4	4 parametric
P6	6 parametric
B1	bandpass/band-reject

It contains highpass and lowpass filters (20 Hz to 21 kHz), high-shelving EQ (5 to 21 kHz), low-shelving EQ (20 Hz to 5 kHz), and two parametric EQs (20 Hz to 21 kHz).

Except for the highpass and lowpass filters, which have no gain controls, each section of every *FilterBank* configuration has its own gain slider, frequency slider, and bypass switch. Additional controls are provided for specific filter types.

For example, the shelving EQs' Peak, Dip, and Slope parameters provide extra shaping power that helps to distinguish *FilterBank* from other EQ plug-ins. The Peak control determines the amount of added punch or emphasis near the shelving frequency. The Dip control does just the opposite, reducing the amplitude of frequencies near the shelving frequency. Naturally, increasing one of these controls decreases the other automatically. The Slope parameter controls the gradient of the shelved response (that is, the transition between the shelved and nonshelved bands). Using these controls, you can simulate some of the characteristics of classic analog equalizers (more on this subject in a moment).

The highpass and lowpass filters in the E6 configuration offer you a choice of either 6- or 12 dB/octave attenuation. They also provide an input-level control with a phase-reversal switch.

PARAMETRIC POWER

The P6 configuration consists of six identical parametric equalizers whose frequencies range from 20 Hz to 21 kHz, with possible Q settings ranging from 0.2 to 4.0 (see **Fig. 2**).

I used this configuration often, especially when dealing with tricky, heavily processed sounds containing unwanted peaks and dips at multiple frequencies. As you might expect, having a 6-band parametric EQ allows you to focus on specific frequencies with even more precision than is possible with conventional 4-band parametrics. In fact, with individual Q controls for each band, in addition to frequency and gain controls, you often have more control than you need. (I'm not complaining, of course!)

BOTH HIGH AND LOW

The F2 configuration (see **Fig. 3**) comprises highpass and lowpass filters, each ranging from 20 Hz to 21 kHz and offering 6, 12, 18, or 24 dB/octave attenuation. Both filters have Peak and Frequency controls. Peak lets you adjust the amount of resonance at the cutoff frequency by up to 24 dB, depending on the attenuation curve selected.

High- and lowpass filters with this level of flexibility and control are invaluable, allowing for drastic frequency changes. This makes them very good for corrective applications (eliminating stray low-end rumble from a live recording, for instance) as well as for special effects. You can create quasi-bandpass effects by setting the highpass and lowpass cutoff frequencies close to each other, or you can deconstruct sounds completely.

PASSING THOUGHTS

The B1 configuration allows either bandpass or band-reject (notch, if a narrow bandwidth is selected) filtering with very precise control. It's intended as a corrective tool for removing hum, buzz, or other narrowband noises from a signal without heavily compromising the signal itself. As with F2, though, you can create interesting effects with this tool, such as the

FilterBank
Minimum System Requirements
Power Macintosh; Pro Tools PCI, D24, Mix, Mix-Plus, or PowerMix (AudioSuite only) system with v. 4.0 or later software; limited support for NuBus-based Pro Tools

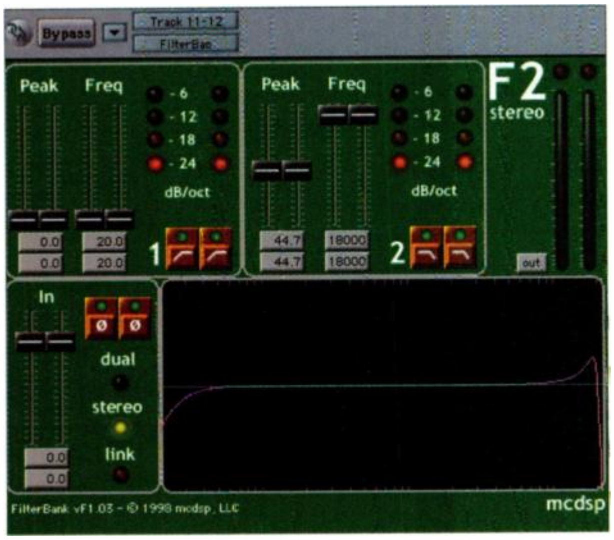

FIG. 3: The F2 configuration, shown in stereo, includes highpass and lowpass filters with selectable slopes. The Peak control allows you to adjust the amount of resonance at the cutoff frequency.

ever-popular "telephone EQ" vocal effect, to choose an obvious example.

SPECIAL FEATURES

Several other aspects of *FilterBank* are noteworthy. For example, when the plug-in is instantiated across a stereo track, separate left and right controls appear for each slider. You can stereo-link them absolutely using the Stereo button, or link them relatively using the Link button. When you use an absolute stereo link, changing the value of one channel changes the other to maintain the same value. When you use a relative link, changing one channel alters the other by the same relative amount, but the two can have different values. The left and right channels can be unlinked using Dual mode.

Another interesting feature is *FilterBank*'s Analog Saturation Modeling. This algorithm ensures that *FilterBank* will not produce digital clipping when overloaded. Instead, it exhibits a distortion similar to that of an overloaded analog equalizer. In practice, this feature works pretty well: it's far less obnoxious than digital clipping. This is indeed a welcome feature in the digital audio universe.

FROM SUBTLE TO EXTREME

Throughout the review process, *FilterBank*'s sound proved quite smooth, even at extreme cut or boost settings. The plug-in can make very subtle tonal changes, as well as totally drastic sound alterations, and it generally had a decidedly musical character. I now use it for virtually all applications that call for an EQ plug-in.

McDSP includes a large number of presets, some of which are designed to emulate the sounds of various well-known analog equalizers by companies such as Neve, GML, and Avalon. Although I didn't have these devices and could make no direct comparisons, the presets sounded musical and pleasant and were, in fact, reminiscent of high-end analog equalizers. I think it's safe to say that this is the best-sounding TDM equalizer yet.

MY TWO CENTS

My gripes are minor and few, and mostly pertain to graphic design. First of all, the screens for the more elaborate configurations, such as P6 and E6, look a bit cluttered. One such example is the lack of clear graphic distinctions between the parts of a configuration; perhaps color coding them would be helpful.

I'd also like to see a more intuitive layout. In the E6 configuration, for instance, the highpass and lowpass filters are in the lower-left portion of the *FilterBank* window. I expected to find the lowpass filter at the extreme left and the highpass filter at the extreme right.

It would be helpful if the currently selected EQ curve didn't disappear from the Display Graph window when Bypass is engaged. It's annoying not to be able to see the current EQ setting constantly while toggling Bypass to gauge the effect of EQ on the current track. But as I said, these are minor points.

Fortunately, there is a partial solution for the E2, E4, E6, P2, P4, and P6 configurations. Holding down the Control key while instantiating one of these configurations will bring up a simplified version of the user interface, which still includes dual phase-reverse switches and input sliders. Although this alternate user interface doesn't solve the layout problems, it does reduce clutter.

Finally, although the $495 price tag is not unusual for a TDM plug-in, it is over the top for an AudioSuite plug-in.

A FINE FILTER

Judging by its excellent sound and flexibility, a lot of thought and effort clearly went into designing *FilterBank*, and I was very impressed by its performance. I did not test the AudioSuite version, which offers the same basic features but does not operate in real time.

Fortunately, McDSP plans to release a version of *FilterBank* and the newly announced *CompressorBank* for Digidesign's new Real-Time AudioSuite (RTAS) plug-in architecture sometime during the fourth quarter of this year. (For more information on RTAS, see "Stop the Presses" on p. 40 of the October 1999 issue of **EM**.) The company is also working on support for Pro Tools for Windows NT but has not yet announced a release date.

FilterBank is subtle and musical enough to use on acoustic instruments and vocals, yet it allows you to create drastic sound-design effects. I wholeheartedly recommend it to any TDM or AudioSuite user who needs a serious equalizer.

McDSP
FilterBank 1.04 EQ plug-in (TDM/AudioSuite)
$495

FEATURES	■■■■
EASE OF USE	■■■▮
AUDIO QUALITY	■■■■
VALUE	■■■
	1 2 3 4 5

PROS: Extremely flexible, with lots of parameters. Very good sound quality.
CONS: Interface could be better organized. EQ graph disappears during Bypass. Expensive, especially for AudioSuite.
CIRCLE #445 ON READER SERVICE CARD

Peter Freeman *is a freelance bassist, synthesist, and composer living in New York City. He has worked with such artists as John Cale, Jon Hassell, Chris Spedding, Nile Rodgers, Shawn Colvin, L. Shankar, Sussan Deihim, Richard Horowitz, and Seal.*

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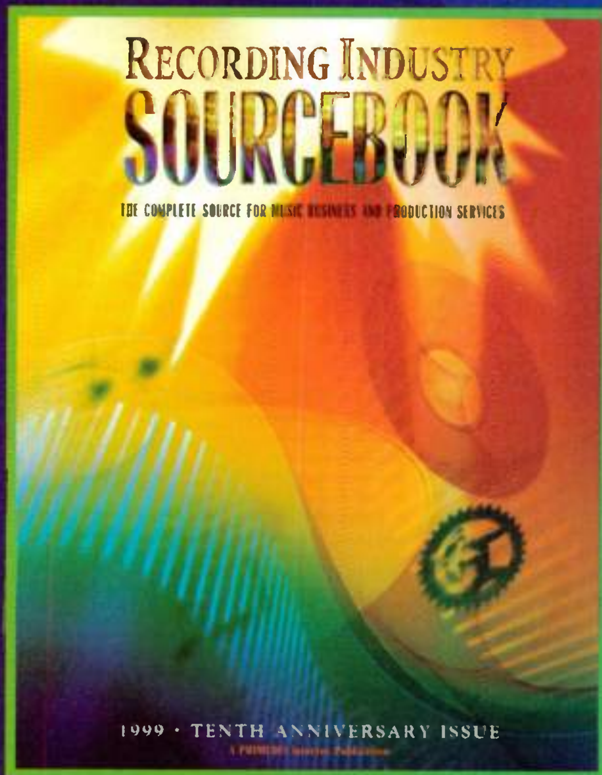

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QUICK PICKS

KURZWEIL

SynthScapes

By Jeff Obee

Those of you who use samplers in your music are probably already familiar with keyboardist, synthesist, and producer Jason Miles. In making *SynthScapes* ($299), Kurzweil invited the sampling maverick into the studio and recorded from his

With Kurzweil's *SynthScapes*, you spend hours happily adrift in new sonic realms.

extensive array of synth sounds to create this sound library for Kurzweil K2000 and K2500 synthesizers.

Synth Scope

The *SynthScapes* CD-ROM contains 516 MB of samples, collected from a gamut of classic and contemporary analog and digital synths. You get the Kurzweil K2000; Korg Wavestation A/D; Yamaha TX802 and TX816; Minimoog; Waldorf Microwave, Microwave 2, and Pulse; the Generalmusic Pro 2; E-mu E-III XS, E-IV, and Vintage Keys; Roland MKS-80, MKS-20, JD-990, and JD-800; and Ensoniq ASR-10, TS-10, and MR Series synthesizers.

You need a minimum of 16 MB of sample RAM and OS version 3.54 to use *SynthScapes.* The sounds are filed in directories according to timbre: Analog, Digital, Claviers, Hybrids, and Leads. There are macros for each category as well. A variety of mono and stereo programs is available within the macros.

Macros of the samples and programs are also included. A Macroprg file allows you to select individual programs, but you will need to load the corresponding sample macro along with it. The Fullprg category enables you to take advantage of the samples that you already have loaded and load in the programs based on those samples.

Synth Escape

There's nothing like feeding fat, enticing sounds to fabulous axes such as the K2500 and K2000. Both the samples and programming are excellent, and whether you use the sounds as they are or import the samples and keymaps into your own programs, you'll spend some happy hours adrift in new sonic realms. There are breathy textural sounds, punchy comping numbers, biting techno patches (many with a bite to them), subdued pads, and distinct digital sounds—in short, a very usable collection.

"Shiner," from the Analog category, is a deep, expanding pad, and "EtherealHybrid" is a beautiful, subtle composite patch that descends in pitch. "Choir Space" is a gorgeous, breathy hybrid patch, the kind of sound that brings a lot of emotion with it. "SlowMatrix" and "MatrixHybrStrg" are both lush Matrix-12 emulations.

To say that "TouchCompSynth" is fat is an understatement. It kicks butt—thick and punchy to the max. The Electric Pianos are superbly detailed. The digital directory is full of goodies. "Textural Lows" is a bassy, growling patch that'll eat your speakers alive. "Low Sweep Waldo," played on a Waldorf Microwave, darts from left to right in a sizzling metallic resonant sweep.

K-series internal effects are used amply, which enhances the programs, and continuous controllers are programmed with savvy. The data controller is programmed to create some wild ring-mod effects on some leads, which is a creative touch.

Synth States

These are rich, meaty sounds. Be aware that Kurzweil has optimized *SynthScapes* for the K2500, and you'll find some differences in how an occasional effect or keymap may load into the older K2000 (although I experienced only one missing keymap on my K2000RS). The documentation is thorough, but I would like to see the differences in the K2500 and K2000 documented. Nevertheless, I strongly recommend this disc—it will be an indispensable addition to your collection.

Overall EM Rating (1 through 5): 4.5

CIRCLE #446 ON READER SERVICE CARD

STEINBERG

Spectralizer (Mac/Win)

By Phil Darg

Steinberg and Spectral Designs have introduced *Spectralizer* ($199), an affordable software plug-in that can substantially enrich your digital audio. It runs as a VST or TDM plug-in on the Mac and as a DirectX plug-in on the PC. It also appears as a native plug-in when used with Steinberg's Windows audio editor, *WaveLab.*

Spectralizer's role in life is to enhance the spectral content of your material. Its settings let you control signal input and gain, as well as the amount of second and third harmonics. You can also adjust the percentage of effects used in a mix. *Spectralizer* includes a Density button for adjusting the amount of an effect, a Solo button that enables you to hear only the added enhancement, and a Kick button for adding greater presence. The Kick feature generates harmonics at the *transient* (onset) of your file, adding a desirable "snap" or "pop" to the sound. Finally, a highpass frequency (HPF) control is adjustable in increments of 1 kHz, from as low as 1 kHz up to 7 kHz.

During my tests, the sound was altered most noticeably by the mix slider, the harmonics sliders, and the HPF cutoff setting. I used these controls to produce enhancements ranging from crunching high

Steinberg's *Spectralizer* provides a number of tools for enhancing digital audio. The clean, uncluttered work area makes the software easy to use.

For manufacturer information, please see Contact Sheet, p. 179.

mids to gentle brightness. The Solo button helped me determine precisely which enhancement was being added.

Silver Bullet

But is *Spectralizer* a silver bullet against the werewolf of muddy sound? In a word: yes. *Spectralizer* added a crisp presence without adding the harshness of treble gain—a significant achievement. (Adjusting treble with equalization often yields undesirable anomalies and changes overall instrument levels, especially in a mix.)

More than that, *Spectralizer* seemed to add presence and highs to the areas that stand to benefit the most, such as vocals and cymbals, without raising the treble levels of midrange instruments. And unlike traditional EQ, which can boost noise along with your signal, *Spectralizer* introduced no additional noise. Truly astounding.

One of the more noteworthy features of *Spectralizer* is its ability to independently adjust second and third harmonics. Experimenting with these settings created some interesting results. However, in most cases, the smoothest sounds resulted from using the second and third harmonics in equal proportions. Moreover, you can adjust the HPF setting as low as 1 kHz and achieve some interesting options when sweetening individual tracks, especially electric guitar sounds.

Warm, No Fuzzies

Steinberg claims that the *Spectralizer* retains the "warmth" of a mix when adding higher harmonics. This is indeed the case: the HPF setting does not affect bass or low-mid frequencies, so the warmth of low pads or the pounding of bass notes is not diminished.

Warning: watch the input and gain controls. These extra features are included to boost the level of tracks as well as enhance them. Overloading the volume level is easy to do, however, and this can introduce some harsh, fuzzy distortion. I found that if the track/mix level is optimal, leaving the input and gain controls at "0" made for the best results.

Another warning: because this process does not introduce additional noise, it's easy to go overboard adding highs. But even too much high-quality enhancement can give you a tinny sound. *Spectralizer* works best when used at moderate settings. It should be used sparingly on individual tracks in an overall mix.

The only significant drawback to *Spectralizer* is that the highpass frequency setting can be adjusted in 1 kHz increments. It would be nice if you could adjust the frequency setting using more precise values, such as 2.4 kHz for vocals or 1.5 kHz for a snare-drum sound.

Spectralizer delivers its best real-time performance on the PC with Steinberg's *WaveLab,* where it appears as a native plug-in. It will work fine under other DirectX hosts, but be sure to check the manufacturer's recommendations about how to get the best performance from your plug-ins. Just about any Macintosh application that supports VST should also serve you well.

Overall, Steinberg's *Spectralizer* gives you a very cost-effective way of adding clear enhancements to digital tracks and mixes, and gives the desktop musician one more tool in the quest for crisp, clear sound.

Overall EM Rating (1 through 5): 4.5

CIRCLE #447 ON READER SERVICE CARD

KEYFAX

Modular Madness (Mac/Win)

By Jeff Obee

Keyfax, proprietor of the well-known Twiddly Bits line of Standard MIDI Files, is covering some new terrain with *Modular Madness* ($39.95), which is billed as the "ultimate analog synth grooves as MIDI." The above statement may seem a contradiction in terms. How can you have the ultimate analog synth grooves as SMFs? It helps to suspend your disbelief as you explore this disc; if you can do that, you may be surprised at what you find.

Modular Madness comes with 30 files (called Mods), each containing a chord pad, a drum groove, and an average of 13 tracks of differing lead lines, arpeggios, and so forth that fit with the pad and drum part. Each track is two or four bars long, and every Mod has a basic key center around which the runs and lines are based. The drum and chord parts are mapped for use with General MIDI, with Program Change messages that call up drum patches on MIDI channel 10, and an appropriate patch for each chord part.

Achieve the effect of a classic arpeggiated/sequenced synthesizer using MIDI files with Keyfax's *Modular Madness*.

Modular Testing

I put Keyfax's claim to the test using the Oberheim Matrix-12 (one of the greatest analog synths ever produced) and a Kurzweil K2000RS. Real-time manipulation is recommended; to accomplish this, I used a Keyfax Phat.Boy controller.

I was taken aback by the authenticity of these tracks. When used with an analog synth and real-time control, you really do get the effect of a classic sequenced/arpeggiated synth. Bubbling lines that peak and dip, many with 32nd-note triplets or other syncopations, provide inspiring and useful enhancements or foundations for dance/techno/electronica music. The fun cranks into high gear when you combine two or more tracks, and the juxtaposed rhythmic and tonal interplay becomes substantially more than the sum of its parts.

Every track is musically useful in some way. I experienced some delightful, serendipitous results using the techno files in Mod 7, scrolling through different sounds on my synths while muting/unmuting and transposing the tracks as the drum part played.

Mods 28 through 30, modeled on the Roland TR-303, are great fun as well. Mod 9, made up of "oblique lines and arpeggios," contains classic analog-style bleeps and bloops, with some sharp interval leaps. Mod 17 is based on a G sus4 to Em7 (♭13) chord progression, and I chopped up its riffs into one-bar segments and arbitrarily mixed and matched them until I achieved a happening eight-bar line. Though the documentation provides the key and length of the files and suggests helpful ways to use them, I preferred to use my instincts while manipulating the tracks, as a simple arpeggiated line or sequence can stoke the creative fires and give way to an entire piece.

The GM Program Change data can be distracting. Be aware that it's included in the drum/chord tracks as well as in some of the modular lines. You can delete it if you aren't using a GM synth.

Give Me Modular

This product works best when used with contemporary synthesizers that have real-time control on the front panel, or with classic instruments that will perform with an external controller. The introduction in the documentation states that there are parts here that you wouldn't play on a modern keyboard in a million years. You're bound to have loads of joyous synth fun with this disc.

Overall EM Rating (1 through 5): 4

CIRCLE #448 ON READER SERVICE CARD

KURZWEIL

Bass Gallery

By Jeff Obee

As a K2000RS user, I'm bothered by the shortage of Kurzweil-specific sample discs. That's why I'm heartened to see several new releases from Kurzweil, especially *Bass Gallery* ($399), which is right up my alley.

As its title implies, this CD is composed entirely of basses. These individual bass sounds (no loops or phrases) are played by such luminaries as Marcus Miller and Jeff Andrews, but any competent bassist could have provided the elements on this disc. I'd love to see a floppy of SMFs bundled with this CD featuring lines, licks, and phrases from these illustrious players.

The sounds are presented mostly as Kurzweil macro files, which conserve memory by avoiding redundant objects. Most of the samples were recorded at both 32 and 48 kHz, and they are presented in subdirectories, such as Finger, Slap, Pick, and Multi, that combine all the playing styles. You also get directories with the raw stereo and mono samples, programs, and text for both Mac and PC.

Covering the Basses

Bass Gallery covers most of the necessary tonal bases well. Featured instruments include the Alembic Series I, Jerry Jones Longhorn, Steinberger XL-2D, and Rickenbacker 4001, as well as 1963 and custom '77 Fender Jazzes, and '65 and '79 Fender Precisions (the '79 is amped and direct). Guild Pilot, M.V. Pedulla, and Ken Smith 5-string basses are also included. The disc is rounded out with Washburn AB-20 acoustic, Tyrolean 1890s upright, Guild Ashbory "rubber band," and BC Rich and custom fretless basses.

The audio quality is top-notch, just as you'd expect from Kurzweil. Some key-mapped transitions in the samples are quite obvious—this is most noticeable in the Guild Ashbory and Pilot. The inclusion of the quirky Ashbory, with its silicone strings and piezo pickup, was a nice surprise.

The Tyrolean upright is an outstanding and beautifully recorded bass that contains many subtle nuances. The Alembic is very effective for creating meaty, distinct slap patterns, whereas the metallic-sounding Steinberger and Rickenbacker are more suitable for rock. The round tones of the amped Fender Precisions make them a good choice for rock and R&B.

My favorites were the fretless, handmade Ken Smith—the fingered samples have an appealing, pronounced midrange and clear highs—and the Jerry Jones Longhorn, which was refreshingly different and fit anything from country to rock to experimental jazz. The Fender Jazz Basses weren't especially to my taste; the producers favored the neck pickup or neck-bridge combination and eschewed sampling the punchier bridge pickup on its own.

Program Notes

The samples here are presented in an excellent set of programs. Each program name has its parameters included in somewhat cryptic, abbreviated form. Each bank starts with the elemental program and is followed by some combination of variations, which include Finger, Slap/Pull, Velocity, Hammered, Distorted, Wah, House, Flange, Detune, Mid Cut, Layered, 12-String, and Sync and Syn (sawtooth-style synth basses). Many of the articulations are included, such as ghost notes, slides, and double stops.

You won't find anything too fancy in the continuous controller area, as these are bass sounds. The CC assignments are optimized for the extra 2500 sliders. Sliders A though D send Control Change messages 6, 22, 23, and 24, respectively, which control tone, effects depth, Alt. Start, and attack envelope.

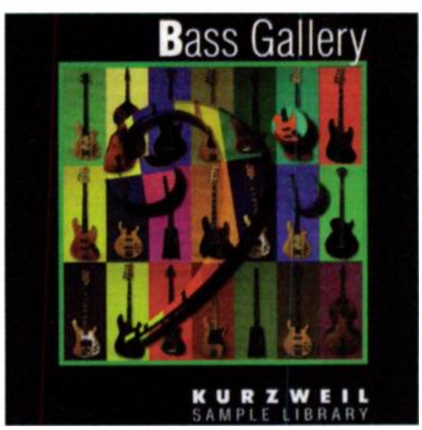
Bass Expectations

The documentation is solid, with a brief description of each bass, explanations of the program name abbreviations, and the whys and hows of macro files.

The numerous articulations on this disc enable you to emulate bass playing quite well. The sounds here are very good, and you're sure to find many that work for you amid the abundant and diverse programs. Of course, $399 is a fair piece of change, but if you use sampled basses a lot in your music, you'll get your money's worth out of *Bass Gallery*. ⊕

Overall EM Rating (1 through 5): 4

CIRCLE #449 ON READER SERVICE CARD



Kurzweil's *Bass Gallery* covers most of the necessary tonal basses very well.

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FEATURES–
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- 24 bit MT Pro Recording Mode for massive headroom and dynamic range
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- 20-bit A/D D/A converters
- 2 optional 24-bit stereo effects processors (VS8F-2) provide up to 8 channels of independent effects processing.
- 12 audio outs: 8x RCA, 2x stereo digital & phones
- New EZ routing function allows users to create and save various recording, mixing, track bouncing, and other comprehensive mixer templates for instant recall.
- 10 audio inputs: 2 balanced XLR-type inputs w/ phantom power, 6 balanced 1/4" inputs, and 1 stereo digital input (optical/coaxial)
- Direct audio CD recording and data backup using optional VS-CDR CD recorder.

KORG
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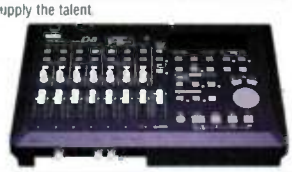
The new D8 Digital Recording Studio features an 8-track recorder, a 12-channel mixer, onboard effects, and basically everything else you'll need to record and mix your music, you supply the talent.



FEATURES–
- 8-track recorder, 12-channel mixer.
- 1.4GB hard disk for up to 4.5 hours of recording on a single track.
- High and low EQ on each channel.
- 130 high-quality stereo digital effects for complete recording in the digital domain.
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FOSTEX
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DIGITAL SAMPLERS

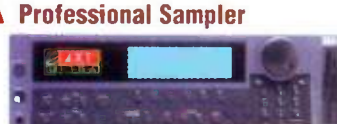
AKAI
S5000 & S6000 Studio Samplers

Akai is proud to announce its next generation of samplers with the introduction of the S6000 and the S5000. Building upon Akai's legendary strengths, both machines feature up-to 128-voice polyphony and up-to 256 MB of RAM. They use the DOS disk format and .WAV files as the native sample format allowing standard PC .WAV files to be loaded directly for instant playback - even samples downloaded from the Internet into your PC may be used And of course, both the S6000 and S5000 will read sounds from the S3000 library.

FEATURES–
- OS runs on easily upgradeable flash ROM.
- 2x MIDI In/Out/Thru ports for 32 MIDI channels
- Stereo digital I/O and up to 16 analog outputs.
- 2x SCSI ports standard • Wordclock connection
- Optional ADAT interface provides 16 digital outs
- .WAV files as native sample format

S6000 ONLY FEATURES–
- Removable front panel display
- User Keys
- Audio inputs on both the front and rear panel allow you to wire the S6000 directly into a patchbay from the back and override this connection simply by plugging into the front.

E-MU E-mu Systems, Inc.
E4XT ULTRA Professional Sampler



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FEATURES–
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CD RECORDERS & BURNERS

HHB
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The new HHB CDR850 is one of the most comprehensive CD-R, CD-RW recorders available today. It delivers the outstanding sound quality that HHB is known at a lower price than previous models. Equipped with a complete range of analog and digital I/O and easy to use one touch recording modes make the CDR850 suitable for any audio environment no matter how sophisticated or demanding.

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- 4 one touch recording modes; 2 manual, 2 automatic
- Sample rate converter accepts any digital signal from 32kHz to 48kHz including varispeed
- Copies all CD, DAT, MD, DVD and DCC tarck starts
- Complete user control over SCMS
- Balanced XLR analog I/O, Unbalanced (RCA) phono analog I/O, AES/EBU digital input, coaxial & optical S/PDIF digital I/O

MICROBOARDS
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Features–
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- Supported Formats: CD DA, CD ROM mode 1 & 2, XA, CD Bridge, Photo CD, CD Extra, Multi Session, Mixed Mode, Karaoke, (optional)
- Duplication Speed: 8X Read/ 4X Write
- Windows 95, NT, 3.1, Mac OS and Unix compatible
- Headphone output with level control

Jam Session PlayWrite 4080

The Jam Session PlayWrite 4080 from Microboards is an all-in-one SCSI CD recorder specifically packaged for audio CD pre-mastering on the Mac OS platform. Built around a 4X write 8X read Matsushita CD-R mechanism the Jam Session comes bundled with all of the pro level software nessecary to edit, master and sequence audio files for CD burning.

Features–
- 4X write/8X read Matsushita CD-R mechanism
- Includes Red Book compliant Adaptec Jam software for editing PQ codes and crossfades for audio CD pre-mastering, will also break down Sound Designer II regions into seperate audio files for editing
- Includes Bias Peak Le for recording/editing and optimizing audio files before burning
- Also includes Adaptec Toast for Data backup, CD Rom CD I, and CD Video authoring
- Includes 2 Microboards CD-R's and SCSI cable Requires Power PC running Mac OS

STUDIO DAT-RECORDERS

Panasonic. SV-3800

The SV-3800 features a highly accurate and reliable transport mechanism with search speeds of up to 400X normal. It use 20-bit D/A converters to satisfy even the highest professional expectations. Panasonic DATs are found in studios throughout the world and are widely recognized as the most reliable DAT machines available on the market today.

FEATURES–
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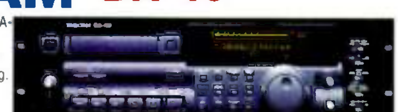
- Single program play, handy for post.
- Adjustable analog input attenuation, +4/-10dBu
- L/R independent record levels
- Front panel hour meter display
- 8-pin parallel remote terminal
- 250x normal speed search

TASCAM DA-40



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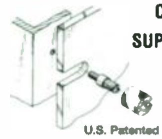

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FINAL MIX

By Larry the O

It's the Law

Films, television, books, and other media have been filled for years with devices, occurrences, and characters that violate the laws of physics. In spite of this, most people accept that in the real world, what goes up must come down and an object displaces its mass in water, regardless of how much technology and finesse is applied to make it appear otherwise.

All the more amazing, then, that in our fields of sound and music there seems to be such widespread difficulty in grasping the inviolate law of production, which is, of course, "Good, fast, cheap: pick any two."

Everybody wants it all. Well, sorry, only two to a customer; it's that simple. And yet, so many people seem to think they can get around it. This, in and of itself, doesn't really bother me. It's when my pinken little posterior gets positioned somewhere in the middle of the situation that I get riled.

The root problem is that choosing two out of three is a compromise, a dirty word to most creative types. On your own projects, you can usually choose to yield on "fast." But what about when you work for a client who can't bear anything less than perfection? Most commonly, that client presents schedule and budget as a fait accompli fixed at levels far below the reasonable. Regardless of any analysis of these factors, the client insists on pursuing the ideal scenario, but if no compromise on time, budget, or quality is accepted, there are only three possible outcomes.

The first is that the ship sails straight onto the rocks and never makes it to the finish. Money runs out, goodwill runs out, time runs out. Since I believe a large part of my job is bringing the project into port, I hate to see this happen.

The second is that compromise eventually occurs, however reluctantly. Maybe the schedule gets extended; possibly more money is kicked down. Or corners get cut. Compromising late in the game is considerably more expensive than budgeting realistically from the get-go.

The law of production demands that balance be maintained. If time, money, and quality are invariable and out of proportion, the resources to complete the project must come from somewhere.

Hence the third possible outcome: sometimes, sheer force of will and elbow grease can carry the day to get a project over a hump and brought to completion in apparent contradiction of the "Two of Three" law. Past a certain point, however, "sweat equity" turns to "blood equity," typically in the form of sound and music people sacrificing any semblance of a life for weeks or months until the project is done. Professional pride frequently drives musicians and engineers put in this position to see a project through even if it means losing money on it and feeling an adverse impact on personal and other business affairs. For such efforts was the phrase *burn out* crafted.

In the end, the law of production always holds. Trying to see how it was satisfied can be subtle: some people don't understand what happened in their project until long after it's done, while others are never able to distinguish compromises brought on by unrealistic production expectations. But life would be easier for all of us if people would realize this and take simple measures such as listening seriously to the assessment of necessary resources that they get from the professionals they have hired for their experience doing the job at hand.

The big question in all of our minds is how to show such a client the light. Unfortunately, this is a column of opinions and observations, of which I've got plenty, not of answers, of which I have few. This question of human nature is a real puzzler—if you ever come up with the answer, won't you *please* let me know?

Larry the O *provides music and audio services with his company, Toys in the Attic, and is a sound designer at LucasArts Entertainment.*

ILLUSTRATION BY STEVE CURL

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