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REVIEWS

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JazzMutant Lemur, Sibelius 4, and 9 more

A PRISM BUSINESS MEDIA PUBLICATION

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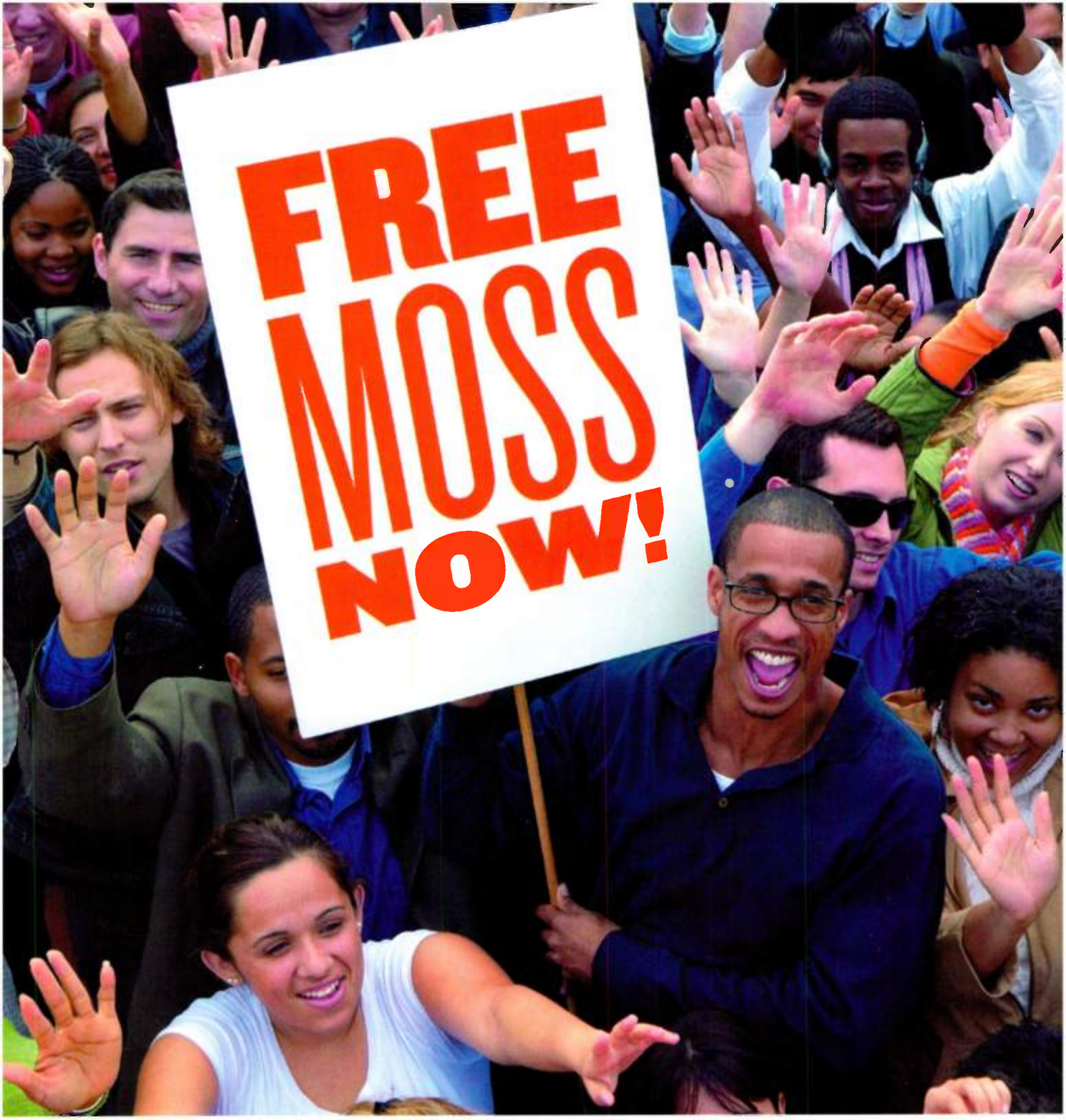

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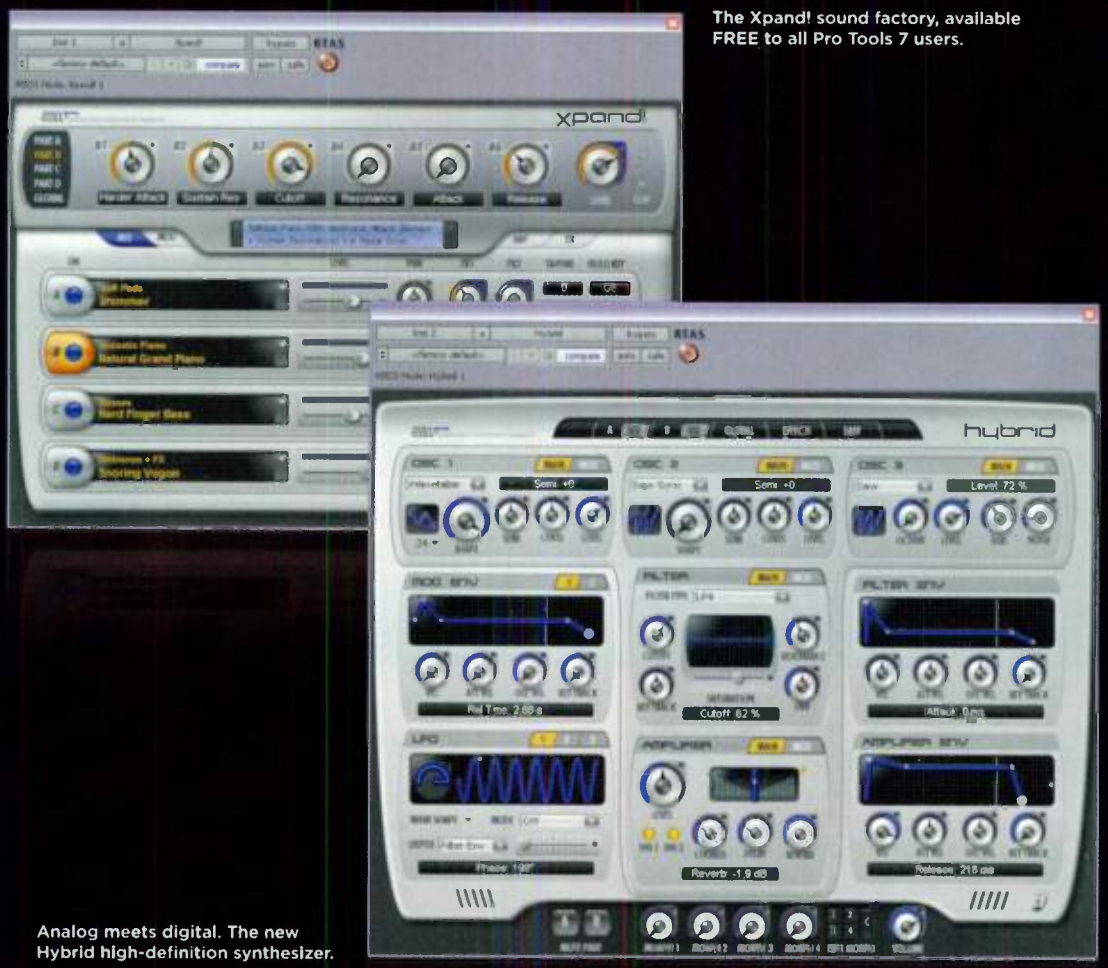

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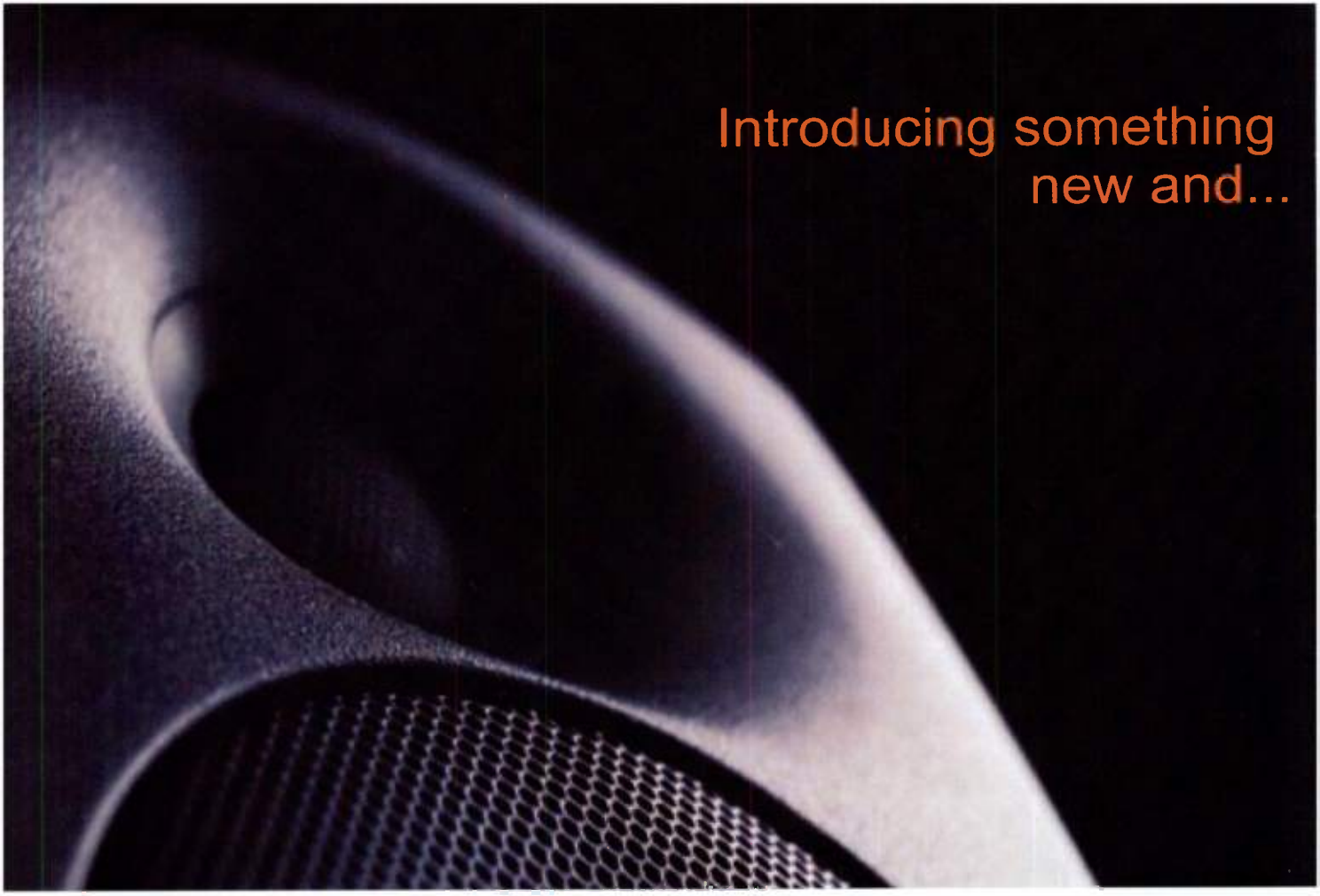

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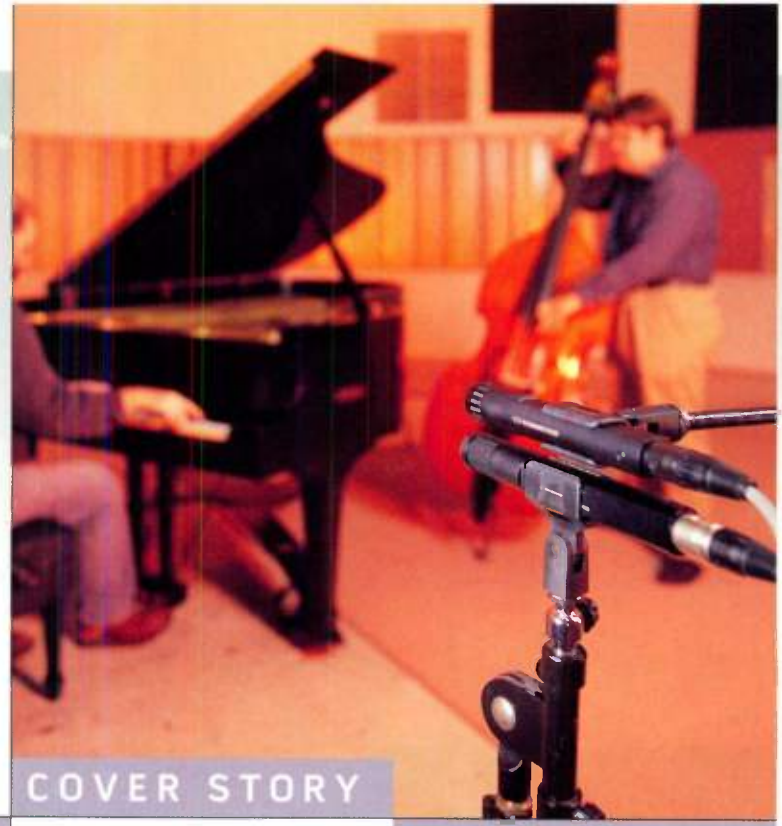
GENELEC®

Electronic Musician
March 2006

INSIDE



COVER STORY

COVER PHOTO BY ANTHONY PIDGEON
THANKS TO MICHAEL COGAN AND BAY RECORDS RECORDING STUDIOS IN BERKELEY, CALIFORNIA, AND TO MUSICIANS MATT INGALLS AND TIM DUFF.

FEATURES

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DigiTech DF-7 Distortion Factory modeling distortion pedal

PSP Audioware PSP 608 MultiDelay (Mac/Win) DSP software

Big Fish Audio *Rock Bass* sample library

Rayzoon Technologies Jamstix (Win) software instrument

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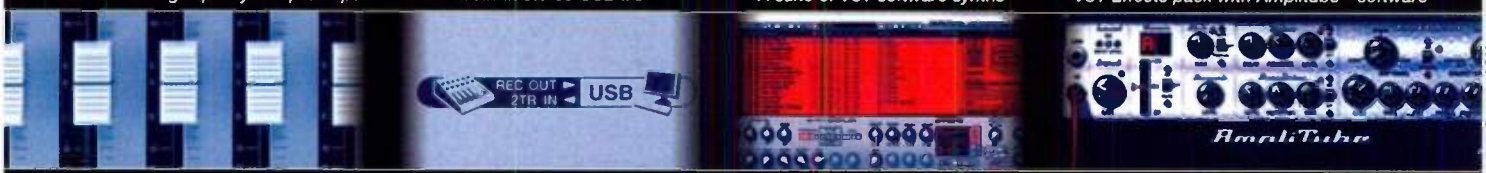

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Blog Is a Four-Letter Word

As I write this, we're preparing to head to Anaheim for the annual Winter NAMM show. Tomorrow, we'll participate in the preshow festivities, including our own Editors' Choice Awards reception. In some ways, this NAMM show will be like others in recent years: lots of press conferences and meetings and a mind-boggling array of new products and software updates. Just a few days after the show, you'll be able to read an entire show report on our Web site at www.emusician.com.

But we're trying something new, too. With help from our partners at *Mix* magazine (to be honest, we're riding their coattails), EM will create its first trade-show blog. The blog will stay on our Web site for a while, so you'll be able to check it out months after the show. And you should do so, because we'll have lots of comments from all over the industry about the show, the products, the trends, and whatever else is on the minds of participants.


JULIE BIRUM

Admittedly, ours won't be the usual blog. For one thing, it's sponsored by Intel (although Intel will not influence its content). And it won't be a classic single-author blog filled with cross-references to other blogs and Web sites. Rather, we'll have a blogging station at our NAMM booth, where the editors of *Mix*, *Remix*, and EM will comment on the show as it happens. We'll also bring in a variety of industry people with interesting and informed opinions and get them to add their two cents' worth. We'll also have a Podcast filled with live interviews, conducted by *Mix* technical editor Kevin Becka, which will be very cool. And it's online now at the EM, *Mix*, and *Remix* Web sites.

I also have a few announcements regarding EM staff changes. First, copy editor Lori Kennedy and assistant editor Matt Gallagher have moved on to bigger and hopefully better things. Fortunately, both are staying in-house and are working at our sister publications: Lori has moved to *Remix*, and Matt is moving to *Mix*.

Lori's name is probably unfamiliar to those who don't read magazine mastheads, but you've read her work in almost every EM article. Her sharp eyes and language skills have made our stories read a lot better. Matt's name is more familiar to EM readers because he often writes our "Pro/File" column and replies to an assortment of reader and manufacturer inquiries. For the past seven years, he has been a very versatile and reliable team player.

Now for the good news: we've hired a terrific copy editor. Marla Miyashiro isn't really new to our team, though, because she has copyedited for us many times as a freelancer. So we know firsthand that she is a top pro, and we're delighted to have her. In addition, we're about to hire a new editorial assistant, and the candidates have been impressive. We welcome our new staff members and wish Lori and Matt the best in their new jobs. And the beat goes on!

Steve Oppenheimer
Editor in Chief

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LAURA WILLIAMS

Letters

Role Models

Body Glove's advertisement in the January 2006 issue of EM (inside back cover) was distressing. In selling products to musicians, I think it would be best to lose the model in a bathing suit with high heels. Body Glove's attempt at humor with a double entendre ("About our model: tough, curvy exterior, yet soft on the inside. And we think Amber is pretty hot too!") was appalling. There is a reason that a professional magazine such as EM doesn't sell sex: it's cheap, shallow, and degrades both the exploited model and the consumer.

Dan Bradley
via email

I'm a longtime subscriber to EM and have always enjoyed its informative articles and reviews. But what's up with your advertisers' fascination with "models" in recent editions?

IK Multimedia's "Super Modeling" advertisement for AmpliTube 2, in the December 2005 issue of EM (p. 105), has been around for months. And now Body Glove's "Bombshell" advertisement, complete with its model named Amber, in the January 2006 issue (p. 139), leaves me wondering where EM is heading.

Don't get me wrong—I enjoy and employ my fair share of attempts at humor. But come on, folks, are we going to have pictures of scantily clad models in advertisements for every device that uses modeling algorithms? I hope not.

It's getting hard to find an appropriate place to read EM. At my day job, such pictures are considered offensive and referred to as "sexual harassment in the workplace." At home, my wife and three daughters would wonder what kind of magazine I'm reading with pictures like that. I also wonder if it makes your female readers feel like the world of music and music technology really is only for men, despite the articles you occasionally publish that try to persuade them otherwise.

So, please, have a talk with your advertisers. Give me the great articles and reviews that I need for wisely spending thousands of dollars on music equipment and for being productive. There are plenty of other magazines out there for those who want to see pictures of supermodels.

Dan Haas
via email

EM associate publisher Joe Perry replies: Our advertisers conceive and create their ads and buy the ad space, so they decide on the message and company identity they want to convey. That said, the publisher reserves the right to refuse advertisements that do not adhere to the publication's standards, policies, and principles. Publisher Dave Reik is ultimately responsible for this at EM, and he has delegated the immediate responsibility to me as the associate publisher. The editorial staff has no authority with respect to advertisements, although I respect their opinions. In fact, the editors usually don't see ads prior to publication.

Practically speaking, I have to use my judgment regarding "taste" in advertising. We serve a diverse readership, and the distinction between acceptable and potentially offensive can be hard to define. On occasion, we have advised advertisers that certain ads were unacceptable and have either refused the ads or induced the advertiser to change the offending portions.

The ads with the models obviously are intended to appeal to our 93 percent male demographic, and, according to the feedback I've gotten, these ads have been successful in that they resulted in product inquiries. On the other hand, some readers may feel that the ads are sexist or simply inappropriate for a technical magazine. But in my judgment, the ads are harmless and are well within mainstream American culture. Given that, I support the advertisers' right to communicate their messages as they see fit.

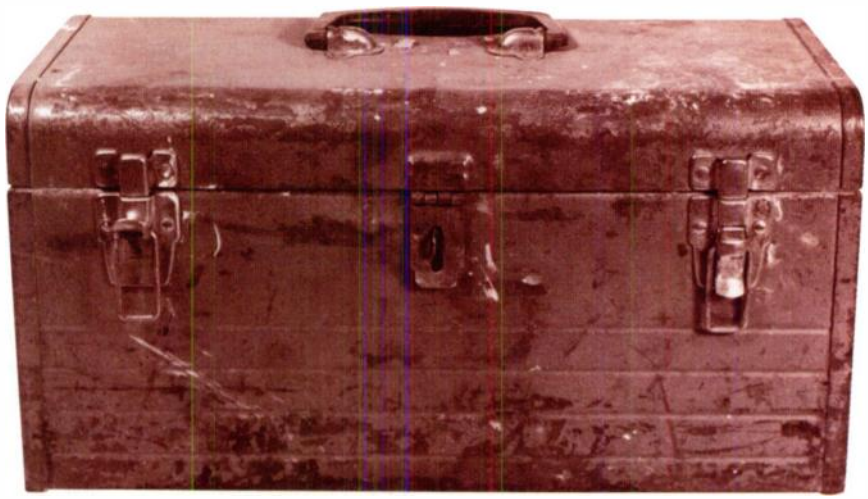
Claim Check

The 2006 Editors' Choice Award for Microphones (Condenser), in the January 2006 issue of EM, went to the Røde NT2-A ($699 MSRP). In support of EM's reasoning for giving the award, the mic's reviewer, Rob Shrock, said that the NT2-A's excellent sound was like "a quieter version of a Neumann U 67."

This key quote contains two assumptions that will be drawn by your magazine's readership: (1) the mic sounds like a Neumann U 67, and (2) the mic is appreciably quieter than the (noisier) Neumann U 67.

Such arguments are irresponsible and somewhat condescending toward your readers. Would a significant number of professional users who test both mics side by side truly come away with a similar

Production tools getting rusty?



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COVER

Hired Guns

We'll show you how to produce a killer demo by tracking with top session players while cutting costs by using efficient production techniques. Topics include planning, budgeting, pre-production, tracking strategies, and file export.

FEATURES

Backing Up's Not Hard to Do

Without a comprehensive backup strategy in your personal studio, a crash or other malfunction could easily wipe out irreplaceable music files. We look at various ways you can back up your files.

Production Values:
Daniel Lanois

Producer and recording artist Daniel Lanois talks about his ethereal production style, his most recent CD, his personal studio, his take on changes in the music industry, and more.

COLUMNS

Sound Design Workshop:
Extreme Processing

Use multiple plug-ins to achieve results that aren't possible using individual plug-ins.

Square One:
A Beginner's Guide to Monitors

EM looks at the various types of speakers commonly used in studio settings.

Working Musician:
Label Enablers

We discuss the several choices that personal-studio owners have for do-it-yourself CD duplication and label printing.

... and much more

Letters

impression? If so, I'm in the wrong business, and maybe on the wrong planet. Professionals who spend upwards of $4,000 for a healthy U 67 must be delusional, irrational, and fiscally irresponsible, because the equally good-sounding, but quieter, Røde costs only one-tenth of the U 67's street price. If beginners or semiprofessionals looking for a good microphone read this and, in good faith, base their purchasing decision on such promises, what will the outcome be the morning after, and in the long term?

Don't misunderstand my reflections. I'm not slamming the Røde product, which may or may not be a fine microphone in its price class. (I have no idea, because I've never heard it.) But if what EM claims in its Editors' Choice Awards write-up is true, then conventional assumptions of what a microphone is worth, as well as entire business and economic models, would undergo a revolution and seismic realignment worthy of front-page treatment in every audio and economic periodical on this planet.

Klaus Heyne
German Masterworks
Corbett, Oregon

Klaus—My comparison of the Røde NT2-A to a Neumann U 67 was intended as a broad stroke to convey the general flavor of the NT2-A to the EM reader. In fact, I still maintain that the NT2-A would be a good choice for recording applications where a U 67 would be used (such as vocals, cello, and so on). It's simply a matter of fact that the Røde microphone contains quieter electronics, and I'm sorry if that offends your sense of right and wrong. I have used U 67s a lot throughout my career, and—although I would not characterize them as a noisy microphone—the latest generation of microphones is setting new standards for both quiet electronics and price.

EM's readership includes the semiprofessional and hobbyist markets, as well as professionals with personal studios and limited budgets. I'm not condescending to them when I inform them of a product that I believe brings them closer to the "big gun" products that are venerated by top professionals with larger budgets. In fact, that's my job as a reviewer. I appreciate—in fact, I own—many big-ticket items as well, fully understanding their practical and psychological benefits for the recording process.

However, the fact that many big-name producers regularly use modern, less expensive tools—including microphones that cost less than $1,000—for lead vocals on major-label productions further attests to the fact that, indeed, a revolution of both economic and technical models has already occurred. The fact that second-generation ADATs sounded better than digital recorders that cost 50 times more only a decade earlier is a trend that is continually playing out in all areas of audio and video production.

I love and respect the U 67, and I'm not implying that the Røde NT2-A is better, or even the same—only that they are broadly similar. The NT2-A makes a good option for those who cannot afford a large-diaphragm microphone of that caliber.—Rob Shrock

We Welcome Your Feedback

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Error Log

December 2005, "Quick Picks: Doepfer R2M," p. 116. The review does not mention that Doepfer products are distributed in the United States exclusively by Analogue Haven (www.analoguehaven.com).

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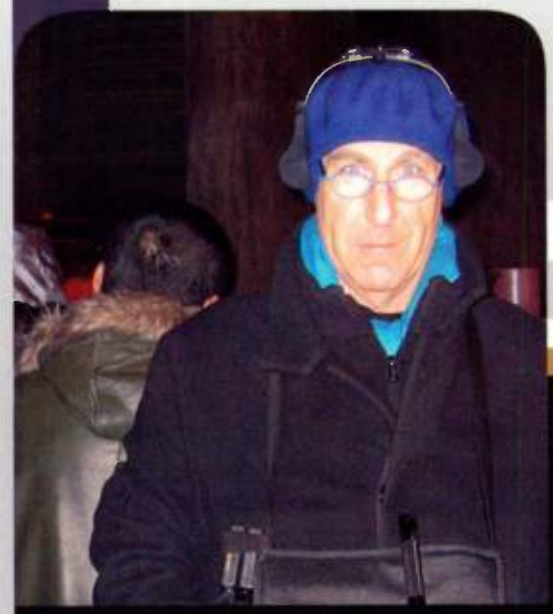

Yoichi Yamada

EMspotlight

In the Time of Bells with Steven Feld

Armed with his trusty recorder and microphone array, phonographer Steven Feld has traveled the globe in search of bells. The third installment in his *Time of Bells* CD series, *Musical Bells of Accra, Ghana*, captures a unique hybrid of ancient rhythmic textures and post-Coltrane jazz that flourishes on the Gold Coast. In this exclusive interview, Feld shares his insights on tracking elusive sounds in unusual places. By Gino Robair. emusician.com/em_spotlight

On the Home Page

EM Web Clips

A collection of supplemental audio, video, text, graphics, and MIDI files that provides examples of techniques and products discussed in the pages of *Electronic Musician*.

EM Guides Online

Get detailed specs on thousands of music-production products with our free online Computer Music Product Guide and Personal Studio Buyer's Guide.

Show Report

The 2006 Winter NAMM show is the biggest annual musical-instrument expo in the United States. Visit emusician.com for our report on the exciting new recording gear, music software, and electronic musical instruments unveiled at this year's show.

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The EM Seminars on Demand are an exciting way to see new products and learn new applications and techniques online and at your leisure. Korg USA's top sound designer, Jack Hotop, shows you how to customize a variety of synth sounds to suit your performing and compositional styles.
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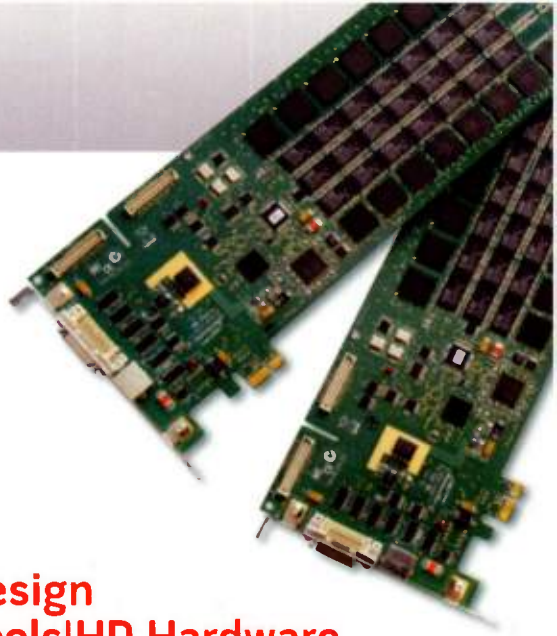
WHAT'S NEW

By Geary Yelton



iKey-Audio iKey

With the proliferation of the Apple iPod and other portable audio-playback devices, musicians naturally want to be able to record audio on them. To meet the musicians' demands, iKey-Audio (www.ikey-audio.com) has begun shipping the iKey ($229), a portable audio interface that connects to USB-compatible storage devices, including MP3 players and flash drives. Weighing approximately two pounds, the iKey supports USB 2.0, 1.1, and 1.0. Two gold-plated, line-level RCA inputs let you capture audio and convert it to MP3 or WAV format. A miniplug-to-RCA adapter cable is included.

The iKey can record 16-bit, 44.1 kHz WAV files and MP3 files at compression rates from 128 to 256 Kbps. An LED indicates the amount of disk space remaining on the target drive, and another LED indicates a low battery charge. The device runs as long as five hours using four AA batteries, and a DC power adapter is included.

Digidesign Pro Tools|HD Hardware

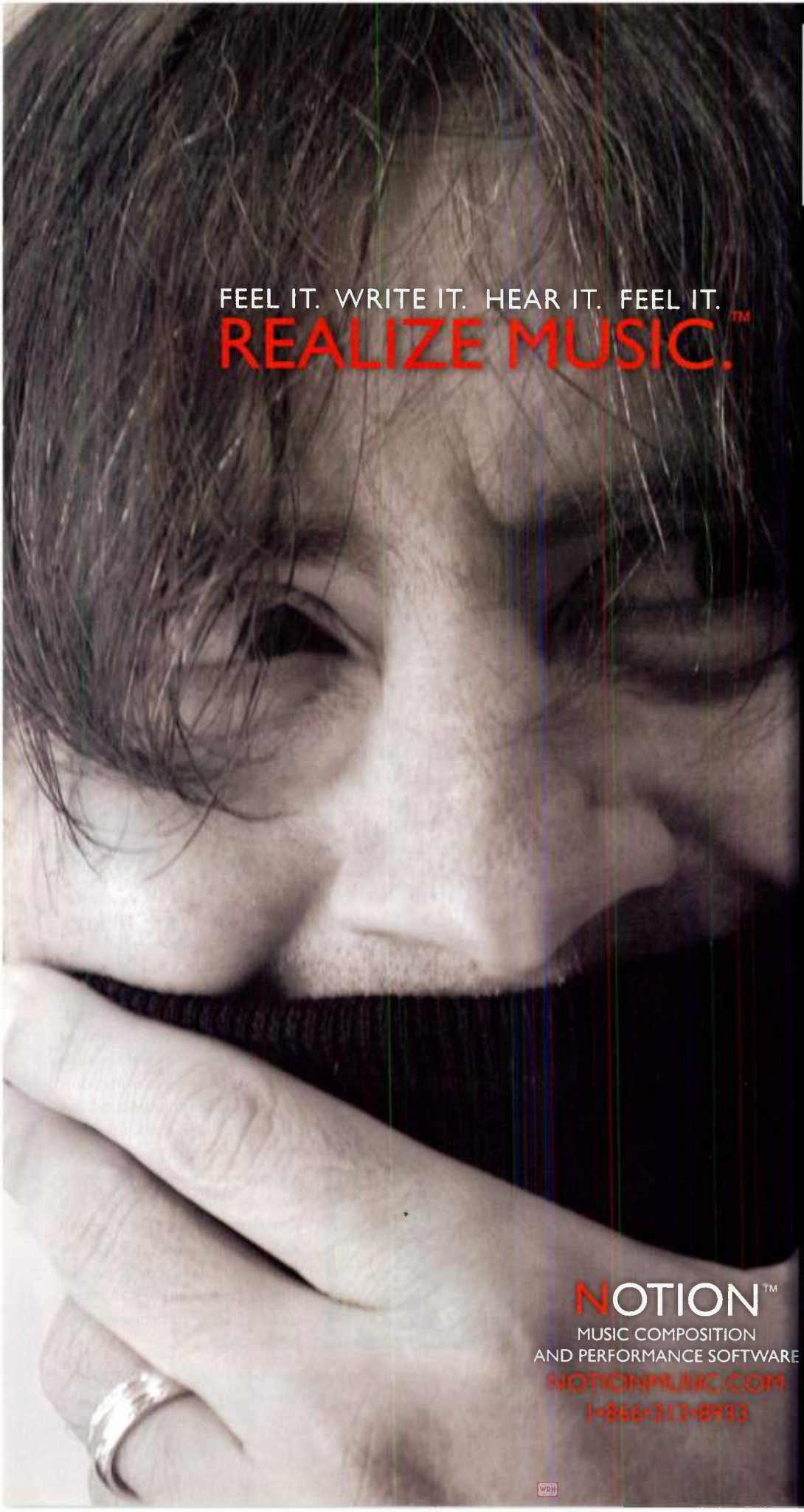
To maintain compatibility with Apple Computer's latest wave of Power Mac G5s, Digidesign (www.digidesign.com) now offers two versions of its Pro Tools|HD digital audio hardware systems. In addition to the PCI-based version previously available for PCI and PCI-X expansion slots, the company has announced a PCI Express version (Mac, $7,995 to $13,995) that will accommodate the PCIe slots in dual-core G5 Macs. Digidesign also plans to qualify its PCIe systems for Windows-based computers whenever PCs with three PCIe slots become available.

Because new Power Mac G5s have only three usable PCIe expansion slots, they limit users to a maximum configuration of three cards. If you need more, you'll need to use Digidesign's new Expansion|HD (Mac/Win, $2,395) with the PCI version of the card. The Expansion|HD is a six-slot expansion chassis that connects to the host computer using one of its existing PCI, PCI-X, or PCIe slots.

Best Service Chris Hein Horns

Best Service (www.soundsonline.com) has introduced Chris Hein Horns, vol. 1 (Mac/Win, $399), a 3.5 GB sample collection teamed up with Native Instruments' Kontakt Player. It features trumpet, trombone, alto sax, and tenor sax, with as many as 44 playing techniques per instrument and 20 Velocity layers per note. The emphasis is on jazz performance, and each instrument provides light, medium, and full versions to accommodate your computer's available resources.

Keyswitching allows you to vary your playing technique within each program. Tones have been sampled with and without vibrato, and they include techniques such as flutter tonguing, falls, trills, shakes, runs, and swells. Melodic ornaments in major and minor keys are on hand. MIDI control changes have been preassigned to control airstreams, growls, spit, and other details of articulation, as well as EQ, reverb, delay, and chorus parameters. Chris Hein Horns runs standalone or as an Audio Units, DirectX, RTAS, or VST plug-in.



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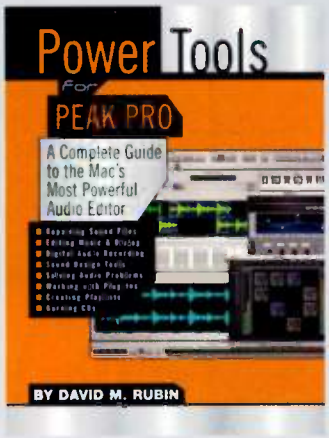
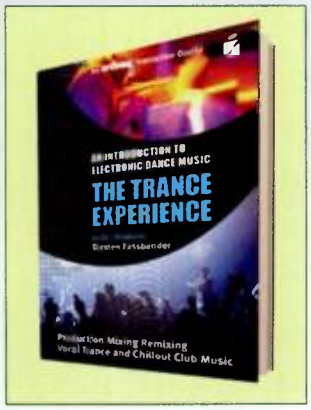


In his book *Power Tools for Peak Pro* ($22.95), former EM associate editor David M. Rubin explores the inner workings of the Mac's most enduring stereo audio editor. After outlining features exclusive to versions 4 and 5 and briefly discussing digital audio fundamentals, he dives into Peak's user interface, recording and importing music and dialog, and various editing tasks. Additional chapters cover topics such as creative sound design, using plug-ins, working with loops and regions, mastering CDs, and batch processing. "Tool Tips" appear throughout the text to familiarize you with Peak's capabilities. An appendix lists keyboard commands and file formats that the application supports. *Power Tools for Peak Pro* is published by **Backbeat Books** (www.backbeatbooks.com).

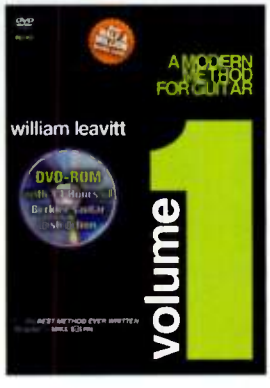
The Trance Experience ($80) is an interactive book, software, and soundware bundle from **Sound.org**, a division of Waves. Using the included audio tracks, samples, and plug-in presets with your DAW software of choice, German DJ Torsten Fassbender reveals his techniques for producing electronic dance music. In 175 pages, he walks you through composition, arrangement, signal processing, mixdown, and mastering, explaining details at every step in the process. Five DVDs furnish session files in all popular multitrack formats covering three electronic subgenres: vocal trance, remix, and chill-out. A sixth DVD contains installers for time-limited demo versions of Waves plug-ins, and video presentations about Waves Tune and GTR.



Peachpit Press (www.peachpit.com) has published another title in the Apple Pro Training Series, *Soundtrack Pro: Professional Sound Design* ($44.99), by Mary Plummer. Providing a thorough introduction to working with Apple's audio-editing application, the book shows you around Soundtrack Pro's interface and quickly progresses to arranging a multitrack project. You'll learn how to create suspense with sound design, build suspense with editing techniques, repair and enhance voice-overs, and create your own Apple Loops. More than 100 pages are devoted to using the Waveform Editor. The included DVD-ROM contains media files to accompany the step-by-step lessons.

At 680 pages, *Sonar 5 Power! The Comprehensive Guide* ($39.99) delivers just what it promises: a thorough tutorial and an extensive reference that will help you to master the powerful Windows-based sequencer. Author Scott Garrigus (www.garrigus.com) takes you on a guided tour that ranges from getting started to using advanced applications. After detailed examinations of Sonar's commands and layout, you will learn how to edit recordings, compose with loops, create sheet music, and master audio CDs. Chapters address software synthesis, effects processing, and surround, as well as using advanced features such as StudioWare and Cakewalk Application Language. Published by **Thomson Course Technology PTR** (www.courseptr.com), the book includes an interactive CD-ROM.

The DVD-ROM *A Modern Method for Guitar*, vol. 1 (Mac/Win, $29.95; bundled with book, $34.95), from **Berklee Press** (www.berkleepress.com), presents 14 one-hour lessons with Larry Baione, chair of Berklee College of Music's guitar department. Based on William Leavitt's book of the same name, the disc plays video clips on your computer in a QuickTime window. Each lesson is divided into chapters that teach music theory and have exercises for improving technique. Interactive duets and play-along segments let you sit in with the instructor. You'll learn to play songs that will strengthen your melodic, harmonic, and rhythmic understanding and enhance your performance.



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Korg TR

Three new synthesizers from Korg (www.korg.com) are the 61-key TR61 ($1,599), the 76-key TR76 ($1,899), and the 88-key TR88 ($2,499). The TR employs the Hyper Integrated (HI) sample-playback engine first developed for the Triton. Each model has 64 MB of waveform ROM that includes a new stereo piano. The TR is a true keyboard workstation, complete with a dual-polyphonic arpeggiator, 89 types of effects, and a 16-track, 200-song MIDI sequencer. A Cue List lets you assemble songs from sections such as verse and chorus. Thirty-two Template Songs memorize preset programs and effects. A media slot lets you store TR data on SD cards with capacities as large as 1 GB, and a USB port facilitates computer connectivity.

The TR's real-time controllers include a joystick and assignable knobs and switches, and the keyboard supports Velocity and Aftertouch. The Alternate Modulation Source (AMS) routing scheme brings life and expressivity to as many as 512 Programs, 384 Combinations, and 24 drum kits. The TR is compatible with many sounds for the original Triton, the Triton LE, and the Karma.

By adding the optional EXB-SMPL expansion board ($260), you can upgrade any TR to support 16-bit, 48 kHz user sampling and SCSI connectivity. The expansion lets you graphically edit sampled waveforms, crossfade loops, time-slice, and time-stretch, and perform other essential functions.

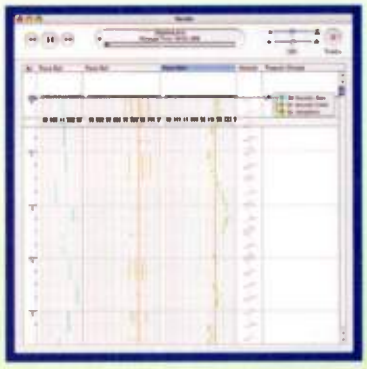
Download of the Month

RONDO (MAC)

Rondo 1.2 ($25), from fracturedSoftware, is a MIDI-file player for Mac OS X. Rondo automatically routes MIDI playback to the QuickTime Music Synthesizer, which is General MIDI–compliant, but you can also choose any MIDI port that has been enabled in OS X's Audio MIDI Setup utility. For example, you can use an IAC port to route playback to other running MIDI applications, such as standalone virtual instruments or DAWs running instrument plug-ins. But Rondo's main purpose is to quickly audition MIDI files, and QuickTime is fine for that.

Using Rondo is simple: you load a Standard MIDI File and press Play. Notes are displayed in a vertically scrolling piano roll, which makes it much easier to analyze what's going on than the more common horizontally scrolling variety does. The notes, which are color coded by track, pass under a movable play head with a resizable piano-keyboard graphic that animates its keys with the track colors as notes pass underneath. Individual tracks can be muted or soloed so that you see and hear only the parts you want. The visual feedback, together with the ability to slow down the tempo, makes it easy to analyze the parts being heard.

To further aid in analysis, Rondo provides playback start and end markers. Playback can be set to loop between the markers, play the marked region once, or simply jump back to the start marker each time you initiate playback. In an interesting twist, columns (including the piano roll) can be duplicated, and they have independent display settings. For example, you can display a piano part on one piano roll and a bass part on another, or have the Velocity column display only drum-track Velocities.



Rondo is not a file editor, and there's no save function, which would be convenient for saving different Rondo setups. But for quickly auditioning MIDI files on either the QuickTime synth or one of your own virtual instruments, Rondo is a useful tool and is well worth taking a visit to the fracturedSoftware Web site (www.fracturedsoftware.com).

—Len Sasso

G-70: Music Workstation Version 2 The best gets better with Version 2 software for the world's most powerful music workstation. Improve your tracks and expand your creativity with Guitar Mode, a new standard in guitar simulation, plus a MIDI File Style Converter, Piano-Roll Editor, and new Music Styles and Music Assistants. Get the facts at RolandUS.com/G70.

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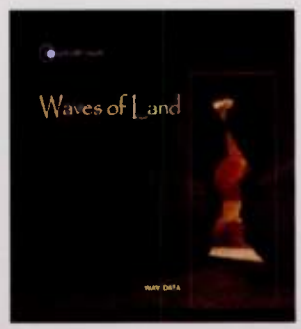
PERFORMANCE

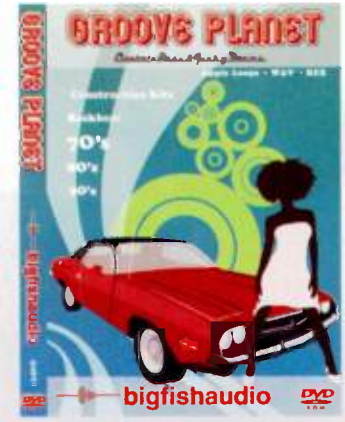
Sound Advice

The newest addition to the *Symphonic Collection* from **Sonic Implants** (www.sonicimplants.com) is *Harp* ($259), a 2-DVD set exclusively for GigaStudio 3. Featuring a huge variety of orchestral harp sounds, the more-than-5 GB sample library captures the timbral palette that the harp brings to any musical setting. Harp tones played with fingers are sampled at four Velocities, with three Velocities for finger mutes and two for playing with fingernails near the soundboard. In addition to normal play mode, you can use MIDI CCs to emulate a harpist's ability to retune each string using pedals. Tremolos and harmonics are also provided, as well as a generous assortment of sampled glissandos in major, minor, diminished, pentatonic, and whole-tone scales. Also included are eight GigaPulse halls to enhance your harp performances.

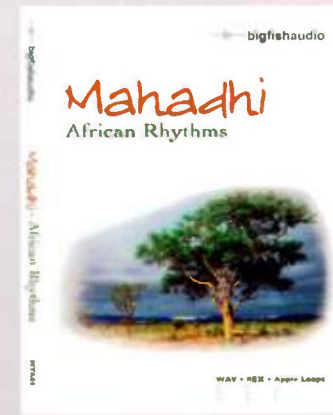
Less-traditional sample aficionados will be interested in *Waves of Land* ($60 WAV, $30 REX), a downloadable 16-bit sample library from **UniversumKiosk** (www.universumkiosk.com). The collection contains atmospheric textures and effects, rhythmic loops, and construction kits assembled from synthesized timbres. The developer calls them Ambient Inspiration Resources (AIR), because they are meant to push your music in abstract directions that you might not explore on your own. Created by sound designer Ton Driessens, most of the sounds are electronic in origin and heavily processed. Groups of files from *Waves of Land* are also available as individual sample packs ($5 to $10).



Prolific soundware developer **Big Fish Audio** (www.bigfishaudio.com) persists in its mission to enrich the world with sampled loops in every musical genre. *Groove Planet* ($49.95) focuses on funk, supplying plenty of drum and bass tracks in construction-kit format. The DVD-ROM delivers more than a gigabyte of content in the form of 16-bit WAV, Apple Loops, and REX files. A total of 22 danceable grooves with names such as "Harsky and Stutch," "Night Train," and "Tom's Hot Salsa" each provide as many as 17 mix-and-match loops played by a live drummer and bassist at tempos from 85 to 185 bpm. In addition, the disc provides two drum kits and five multisampled basses for Kontakt, two drum kits and eight basses for Reason NN-XT, and one of each instrument for GigaStudio. A demo version of Ableton Live 4.01 is also included.



Also from **Big Fish Audio**, *Mahadhi: African Rhythms* ($99.95) explores African percussion in a 2.24 GB sample library. The word *mahadhi* is Swahili for *rhythm*, and the collection draws from the rhythmic traditions of drummers from Algeria to Zimbabwe. The DVD-ROM contains 1,837 loops in 16-bit Apple Loops and 24-bit WAV and REX2 formats. Loops in each of the three formats are grouped into 27 folders arranged by tempo (from 90 to 170 bpm), and within those are folders arranged by individual instrument. Each track was performed by a live ensemble, and features traditional instruments such as bembe, conga, djembe, ekwe, kalimba, ogenne, and talking drum.



M-Audio (www.m-audio.com) is shipping a line of sample discs called the *ProSessions Producer Series*. Developed by Sonic Implants, each of the five DVDs delivers instrument samples in five formats: EXS24, HALion, Kontakt, MachFive, and Reason NN-XT. *Orchestral Strings* ($119.95) features world-class musicians from the Boston Pops and Boston Ballet Orchestra in violin, viola, cello, and bass sections and in combined ensembles. Likewise, *Orchestral Brass* ($119.95) features musicians from the same orchestras playing instruments such as trumpets, trombones, and French horns. Both orchestral libraries were recorded with natural ambience in a variety of playing styles. *Afro-Cuban Percussion* ($99.95) provides bongos, congas, cowbells, maracas, surdos, timbales, and many other Latin instruments, with articulated hits for each. The disc *Electric Bass, vol. 1, Rock* ($99.95) supplies five electric basses with several variations in assorted rock styles. *Heavy Guitars* ($99.95) concentrates on classic rock sounds, with variations ranging from tones and chords to sustains and effects.

Rev Up

VIRSYN TERA 3

Tera 3 (Mac/Win, $339; upgrade $119) is the newest version of the versatile software synthesizer from VirSyn (www.virsyn.com). It introduces a slew of new features, not the least of which is multisample playback. Each of the three audio oscillators can now play samples and preset waveforms, with as many as 32 individual samples per oscillator. The new Spectrum oscillator is capable of stereo wavetable synthesis; wavelet, operator, and Walsh synthesis are also available.

Tera 3 lets you modulate one VCA input with the amplitude of the other, and the VCA module has two extra inputs for stereo effects. You can route control outputs from the LFOs, and envelopes to the signal-processing modules, too. Other new modulation sources include Key Alternate, which produces a different sound for every other note, and Key Random, which produces a random control signal for each note. You can now copy and paste modulation settings in the modulation matrix. A new Terrain filter type lets you morph between two 16-pole vocal filters. You can save four multisegment envelopes as presets and load them independently.

Other new features include a smaller plug-in interface, RTAS support, an arpeggiator, 8-dimensional morph control, 35 MB of multisamples, and 256 new presets. Tera 3 buyers will also receive a copy of MiniTera, a less complex soft synth that's compatible with Tera's presets.

CELEMONY MELODYNE3

Celemony (www.celemony.com) is now shipping Melodyne3 Studio (Mac/Win, $699; upgrade $149 to $249) and Melodyne3 Cre8 (Mac/Win, $369; upgrade $99 to $199). Like previous versions, Melodyne3 specializes in high-quality time-stretching and pitch-shifting, but it's no longer limited to editing monophonic melodies. The Studio edition now handles polyphonic audio tracks and even entire mixes. Use it to transpose, correct, and reshape chord patterns and complex audio data. Speed up and slow down your tracks without changing pitch, or change pitch without affecting tempo.

By employing audio-analysis algorithms developed for Melodyne Uno, Melodyne3 offers much more reliable automatic note recognition, and you can create your own recognition templates. The Studio and Cre8 editions now support macros for correcting intonation and quantizing timing errors. Melodyne3 is faster and easier to use than previous versions, thanks in part to a redesigned graphical user interface with context-sensitive tools, logically arranged menus, and streamlined work flow in the Arrange and Edit windows. In addition, both editions now offer batch processing in Melody Maker and import sounds from Ueberschall Liquid Instruments.

IMAGE-LINE FL STUDIO 6

Image-Line (www.flstudio.com) is shipping four new editions of its virtual studio workstation, FL Studio 6 (Win): Express ($49 download), Fruityloops ($99 download, $139 boxed), Producer ($149 download, $209 boxed), and XXL ($299 download, $449 boxed). Downloadable upgrades are free for all existing download customers ($29 for box customers). FL Studio 6 features an assortment of new instrument and effects plug-ins. The Mixer is larger and resizable, and it offers enhanced track-routing capabilities. The Play List and the Piano Roll have been improved, and the Browser has new search features and recall functions.

FL Studio 6's rack of new plug-ins contains DirectWave, a sampler that offers Velocity layering, built-in effects, and the ability to sample audio tracks and other plug-ins. Fruity Reeverb 2 emulates acoustic spaces, and Fruity Delay Bank has eight filtering and delay stages. New effects include Fruity Squeeze distortion, Fruity Multiband Compressor, and the morphing equalizer EQUO. Another new plug-in, Chrome, can trigger graphical events with MIDI data. The XXL edition includes Sytrus v2, a sophisticated, versatile synthesizer. You can download a full-featured but save-disabled demo of FL Studio 6 from Image-Line's Web site.

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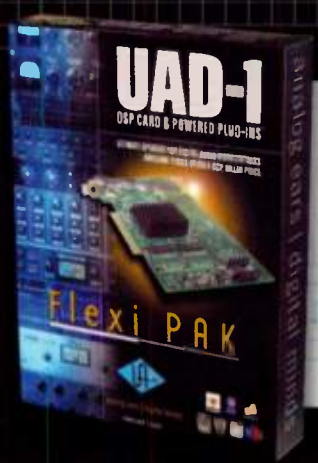

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FabFilter Twin

Dutch software developer FabFilter Software Instruments (www.fabfilter.com) has launched Twin (Mac/Win, $129), a virtual analog soft synth with 24-note polyphony and more than 500 factory presets. Featuring stereo processing throughout, Twin has two oscillators, two multimode filters, and unique drag-and-drop modulation capabilities. The oscillators generate standard waveforms with hard sync, phase sync, and stereo panning. The filters offer a choice of Raw, Gentle, or FabFilter One characteristics, with lowpass, highpass, and bandpass response. You can specify as many as 24 modulation routings simply by dragging an icon from a source to any knob whose value can be modulated.

Twin's 11 modulation sources include three LFOs and three envelope generators, and all their parameters can be mod destinations as well. One ADSR envelope has an additional Hold stage, and the other two have an initial Delay stage. The LFOs can sync to tempo in straight, dotted, or triplets mode. Other features include MIDI Learn, polyphonic portamento, and "smart" parameter interpolation. Twin runs as a VST or Audio Units plug-in.

Zero-G Outer Limits

The newest and most far-out virtual instrument yet from otherworldly synthesist Ian Boddy is Zero-G's Outer Limits (Mac/Win, $199), distributed by EastWest (www.soundsonline.com). Following in the footsteps of Morphology, Outer Limits is a 4 GB sample collection with a front end based on Native Instruments' Kompakt Instrument. Either on its own or as an instrument plug-in, Outer Limits delivers a huge assortment of truly surreal atmospheres, textures, and sound effects, with no traditional instrumental timbres at all.

The range of electronic sounds that Outer Limits can produce is remarkable. Classic synth effects, eerie ambient loops, and dreamlike atmospheres join vocal transformations, dense soundscapes, and drones of every description. Anyone interested in film scoring, sound design, and an assortment of other applications will find a fertile field of aural ideas in this extensive timbral collection.

JBL LSR4300 Series

One challenge of speaker design in recent years has been to overcome the limitations of the acoustic environments in which the speakers are placed, especially personal studios. Consequently, some manufacturers of studio monitors have developed sophisticated systems to analyze and correct problems in smaller recording spaces. JBL recently responded to the challenge with the introduction of its LSR4300 series, two new designs that further refine the company's Room Mode Correction (RMC) and Linear Spatial Reference (LSR) technologies. According to the manufacturer, the LSR4326P ($1,399 per pair) and LSR4328P ($1,699 per pair) make it possible to create accurate mixes in any work space.

The LSR4326P is a biamplified model with 150W driving a 6.25-inch woofer and 70W driving a 1-inch dome tweeter. Specifications state a frequency range from 55 Hz to 20 kHz, ±1.5 dB. The LSR4328P has the same amplification driving an 8-inch woofer and a 1-inch tweeter, resulting in a response of 50 Hz to 20 kHz, ±1.5 dB. Both models have a continuous SPL of 106 dB and a peak SPL of 112 dB. Audio inputs include balanced ¼-inch TRS, XLR, AES/EBU, and coaxial S/PDIF.

Built into each speaker are acoustical analysis capabilities that can measure and automatically compensate for low-frequency problems caused by standing waves and boundaries. To calibrate the system, you connect a supplied calibration microphone and press a button on the front panel or on the supplied remote control. You can view and change calibration settings on your computer using a USB connection and LSR4300 Control Center software (Mac/Win). The speakers also feature the Harman HiQnet networking system to simplify surround setup. EM

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Musical Profiling

By Scott Wilkinson

DSP-based recommendation services help artists.

Online technology is uniquely suited to help new artists gain an audience, thanks to music-recommendation services offered by the likes of Amazon, iTunes, RealNetworks, and Napster. As consumers shop for albums and songs, the e-tailer recommends selections they might like based on what they buy.

One fundamental approach to predictive music recommendation is to have human experts identify certain attributes of songs, albums, and artists, and enter that information into a database, where it is compared with subsequent user purchases. This can provide good results, but it's labor-intensive, and there are hundreds of thousands of recorded artists.

Another approach is called *collaborative filtering*, a statistical method used by Amazon. Collaborative filtering assumes that people naturally fall into "taste clusters"; if you bought album A, you might like album B, because many other people have bought both. This technique can identify connections between recordings that don't necessarily sound much alike. On the downside, you need a huge number of data points (lots of titles and users) to get good results. Amazon's database is certainly big enough to do a good job, especially with popular material, but artists who aren't top-tier may get short shrift in the recommendation department because fewer people buy their stuff.

Gracenote (www.gracenote.com) is working on an interesting answer to these problems. The company supplies music-database tools to various music-service providers, who package these tools for their customers. For example, Gracenote's long-standing CDDB database helps consumers catalog their CD collections by recognizing a CD and sending back its metadata (title, artist, label, and so on). Recently, the CDDB service evolved into MusicID, which adds the ability to extract and identify unique "fingerprints" of individual recordings by "listening" to the audio data.

This year, Gracenote expects to release its Discover music-recommendation service, which combines human analysis and collaborative filtering with a new approach that uses DSP algorithms to extract descriptive attributes from audio files. The algorithms used by Gracenote are derived from those found in unrelated fields, such as gene sequencing and radar, which also rely on finding patterns in large, multidimensional data spaces.

The easiest attributes to identify are low-level features, such as the spectrum envelope and zero-crossings. They provide some idea of the song's timbral structure, but they do not correspond to anything humans would consciously identify. High-level attributes, like tempo, instrumentation, melodic structure, and character of the vocals, are more immediately apparent to human listeners, but they are much more difficult to extract from an audio file using DSP algorithms. The first commercial version of Discover will extract low-level attributes and at least one high-level feature—tempo—but even that basic attribute is not easy to identify consistently across all types of recordings. Other high-level attributes, such as dominant melody and vocal character, are expected to come later, as the algorithms are refined (see **Fig. 1**).

To enable the Discover service, Gracenote will first run the analysis on millions of recordings and store the resulting attributes in a database. Then, when a music fan requests a recommendation, it will be generated based on any "seed" song, album, or artist in the music service's catalog or the user's personal library. Furthermore, new songs can be added as users rip CDs that are not already in the database. Initially, the service will not be available for already-encoded files because the algorithms work best on uncompressed audio.

This technology is meant to fill the gaps in what can be done with human analysis and collaborative filtering. It can also be used to recommend items without much data, especially new artists and recordings, which is good news for aspiring musicians seeking an audience. **EM**

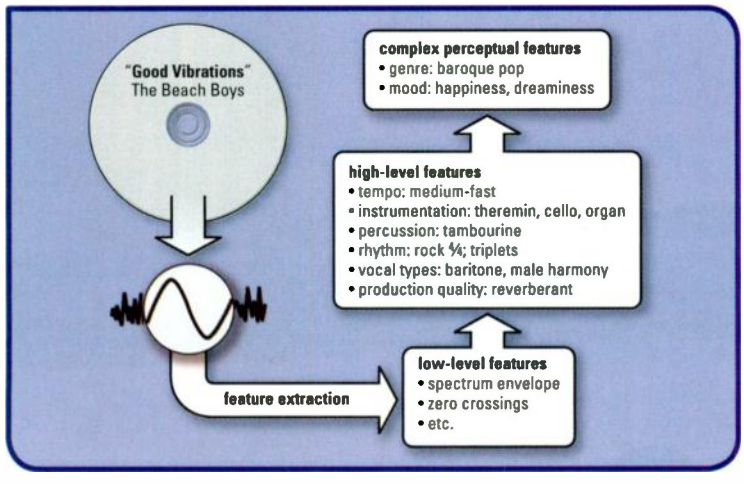

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FIG. 1: Gracenote expects that its Discover service will eventually be able to extract lots of high-level attributes in addition to low-level features from a recording, as illustrated in this flowchart.

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The Eclectic Company

By Matt Gallagher

Z.O.N.K. mixes guitars, turntables, and harmonica.

Z.O.N.K. is an alternative pop band on the rise, having opened nationwide for acts such as the B-52's, Concrete Blonde, and De La Soul. *Half Human* (Harmony Machine, 2005) showcases the band's signature blend of trip-hop, dance, rock, and pop styles. Z.O.N.K. consists of Nic McFiendish (vocals), Dave L (guitars, keyboards, programming), Uncle Bergie (bass and harmonicas), and Walt Dizzle (turntables and vocals).

Z.O.N.K. produced *Half Human* using a Digidesign Pro Tools/24 Mix system and a dual-processor 867 MHz Mac G4 running Ableton Live, BIAS Peak, Propellerhead Reason, and Digidesign and Waves plug-ins. The band's hardware includes a Digidesign 1622 I/O audio interface, Focusrite Platinum VoiceMaster Pro single-channel voice processor, Line 6 Pod Pro guitar direct box, Røde NTK and Shure SM58 mics, Genelec 1030A monitors, Pioneer CDJ-1000 and Technics SL-1200 Mk II turntables, and a Stanton SMX.501 DJ mixer.

"Our studio is in a room in a 1920s flat in San Francisco," Dave L says. "Vocals are recorded in a small room next to it. I would have liked to use a guitar amp, but I just put the Pod Pro in manual mode and dialed in the sounds. Bass was recorded through the VoiceMaster Pro, turntables were recorded out of the DJ mixer, and MIDI sequencing was done mostly with Reason.

MITCH TOBIAS

Half Human/Z.O.N.K.

"A song could start with samples, a guitar riff, a bass line, a set of lyrics, a chord progression, a jam, or a concept," Dave L says. "We record all the parts together, usually against a rough beat. Then I do an arrangement in Pro Tools and we'll tweak it. I'll work on the beats and add keyboard parts, other textures, and samples. Then we rerecord everyone's parts, unless the initial takes were good enough. I'll do any final editing.

"Usually we do a few [complete vocal] takes, and then work on specific areas," he continues. "I then comp them together. Sometimes we double-tracked the vocals." Effects processing plays a prominent role in Z.O.N.K.'s compositions. "The lines between sound design and music started to blur for me after listening to Brian Eno, the Orb, and Future Sound of London. I do all my effects processing with plug-ins, mostly in Pro Tools, taking advantage of effects automation."

The band also created effects from scratch. "The vocal-stutter effect in 'Radio Me' was achieved by drawing in mutes on different divisions of the beat in Grid mode in Pro Tools," Dave L says. "Audio Caffeine" includes a recurring, rapid-fire audio collage. "I drew in 32nd-note drum hits and a volume curve, played a chromatic organ glissando, freaked out on the guitar with a wah-wah pedal, and did pan automation."

Uncle Bergie's harmonica adds warmth to Z.O.N.K.'s sound. "We had him hold an SM58 in his hands with the harmonica and sent that through the VoiceMaster Pro with compression," Dave L says. "I turned up the gain all the way. I used some pretty drastic EQ in Pro Tools and got a raunchy sound full of overtones. For 'Dark and Light' we went for a cleaner sound, so I rolled off a lot of high end and used a cheap-sounding mono reverb."

In mixing *Half Human*, Dave L carefully balanced and distinguished dense layers of electronic sounds. "I double-tracked most of the guitar parts and panned them left and right," he says. "Vocals are usually center unless they're doubled. Every track has a different set of pan positions for synths and other textures, and sometimes they move within a track. I cut a lot of low end out of the bass tracks to give the kick drums space. Uncle Bergie plays the bass melodically anyway, so it works." EM

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LAURA WILLIAMS

Six Creative Uses of Plug-in Automation

By Nick Peck

Automation can go a lot deeper than recording changes in a track's volume and pan position. Just about any parameter of any synth or effects plug-in can be automated. You can exploit automation for sound design, tracking, and mastering. I'll describe six creative uses of DAW automation that can enliven your tracks. Though I happen to use Pro Tools as my DAW, these techniques will work just as well with any DAW and a basic suite of plug-ins. The parameter names may be different, but the concepts are the same.

There's a lot more to automate than pan and volume.

I'm a huge fan of physical controllers—for me, there is something about moving a fader that is more musical than drawing lines with a mouse. Many of these automation ideas lend themselves to twists and pushes of knobs and faders. It's well worth the time to program a fader on your MIDI controller to work with the parameters discussed here.

Fun with Filters

Filtering is the most powerful and most often used tool in any engineer's arsenal. Equalization (EQ), which emphasizes or deemphasizes different parts of a sound's frequency spectrum, is often necessary to use to make different elements coexist peacefully in a mix. Static EQ is enormously powerful, but you can further enhance sounds by automating the parameters of EQs and other types of filters (see **Web Clip 1**). Filter effects such as a swept lowpass filter darkening and brightening an arpeggiated synth line have become musical clichés for a reason—they sound great.

There are dozens of different ways to use automated EQ effectively. In film work, dialog editors are often required to match dialog recorded by different mics at different times. To smooth over the differences, an editor creates snapshots of EQ settings that are applied as the lines play back.

In a more musical context, you can automate EQ to change the timbres of instruments at different sections of the music in order to create contrast. You can, for example, use a highpass filter to drop the low end out of a track or a mix for a couple of bars, creating a floating moment that crashes back to earth when the low end returns. Or you can filter out the highs and the lows, creating a midrange peak to simulate an AM radio effect.

For a real change of pace, try a frequency fade-out. Slowly lower the cutoff frequency of a lowpass filter instead of dropping the level, and then use a volume fade at the end to finish it off.

You can apply LFOs to a filter plug-in's cutoff frequency, adjust a midrange EQ's gain or center frequency over time, or even split tracks into multiple EQ bands that can move around the stereo space independently. There are endless ways that you can use filtering to animate sounds.

Going Away

The ratio of direct to reflected sound and the sound's high-frequency content are cues to the distance from the source to the listener. High frequencies, because of their shorter wavelengths, are more directional in nature—the farther you are from the source, the more the high frequencies scatter and the less you hear them. Here's a trick I like to call the Going Away effect (see **Fig. 1** and **Web Clip 2**).

FIG. 1: Changing the wet/dry reverb mix, increasing the decay time, and adding a lowpass filter creates the effect of sound receding into the distance.

Insert a reverb plug-in on a vocal track, and choose a nice room, hall, or plate algorithm, with a reverb time of around three seconds. Start with the wet/dry mix set to 100 percent dry, and slowly move it to 100 percent wet. Done slowly enough, this effect can be quite subtle. If your reverb plug-in can handle changing the reverb-decay time in real time without creating unpleasant glitching artifacts, you can enhance the effect by slowly increasing the decay time as well. Try starting with a decay time of 700 ms and increasing it to 3 seconds or more.

FIG. 2: The Beatles utilized multiple tape recorders for Automatic Double Tracking, but the plug-in method is much easier.

You can further enhance the sense of moving away from the listener by slowly rolling off the high frequencies of the dry signal. If your reverb plug-in includes an input EQ, you can use that; otherwise, simply insert an EQ or lowpass filter plug-in before the reverb.

Electronic-music pioneer Alvin Lucier created the ultimate artistic statement of this technique, with his piece "I Am Sitting in a Room" (Lovely Music, 1980). He recorded himself speaking in a room, and then rerecorded that recording, repeating the process 32 times. In the end, there was nothing but a wall of reverb.

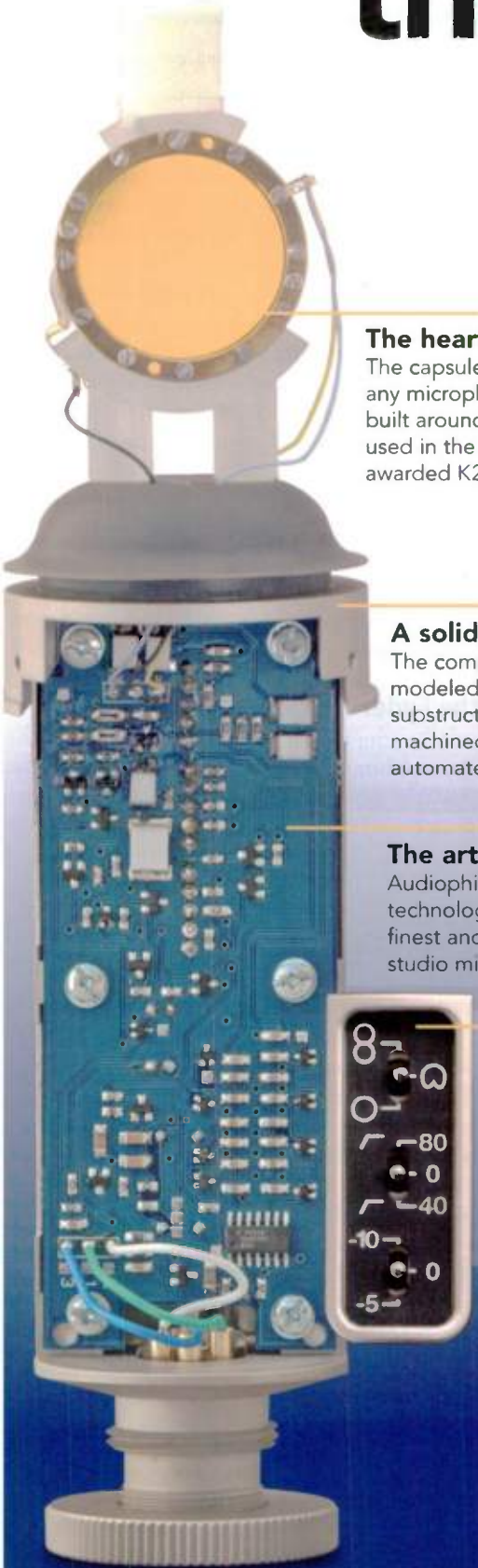
Simulated ADT

The Beatles are known for their gorgeous vocal sounds. At first, they precisely rerecorded the vocal parts in synchronization with the first vocal pass—a process known as *double tracking*. John Lennon wanted to find a way around this tedious process, and in 1966, Ken Townshend, a recording engineer at Abbey Road, invented Automatic Double Tracking (ADT) as the solution. ADT amounts to making a copy of the original vocal on another tape recorder, and then playing both recordings simultaneously for the mix. Millisecond variations in the playback speeds create the doubling effect.

You can now achieve a similar effect more easily with digital delay. Insert a delay plug-in on the vocal track, set the wet/dry mix to 50 percent with no feedback, and find a delay time that sounds about right (settings of 30 ms or less usually work best). Keeping the delay time of the doubled track static doesn't quite create the desired

Beauty is more than skin deep.

First impressions can be misleading. On the surface, many microphones appear to be the same, it's only when you look deeper you see the truth!



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doubling effect. You can get closer by subtly automating the delay time.

To better capture the spirit of ADT, duplicate the original vocal and place it on a second track, with a delay plug-in set to 100 percent wet (see **Fig. 2** and **Web Clip 3**). As you automate the delay time in your piece, notice the colors that appear. Below 10 ms, you get comb-filtering effects that color the frequency response of the signal. Between 10 and 30 ms, you get a more pronounced phasing effect, and from 30 to 80 ms, the tracks start to break apart and appear as separate elements.

Many delay plug-ins produce audible clicks when the delay time is changed on the fly. If your plug-in does that, you may be able to minimize the problem by adding a lowpass filter after the delay.

Remember, you're not limited by a physical number of tape recorders. Try using three, four, or even five separate delays, modulating each one's delay time by hand. You'll get a big sound with lots of animation and subtlety.

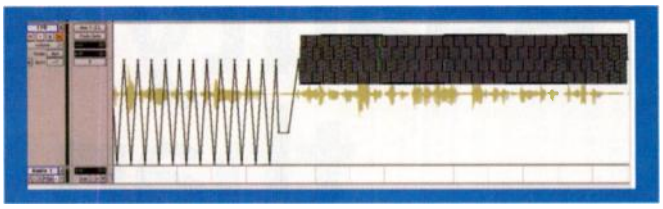


FIG. 4: Drawing rapid automation curves to modulate pan and level parameters can create interesting effects.

Mid-Side Vocal Widening

I'm a big fan of mid-side (MS) recording. I use this technique all the time for field recording, because it allows me a great deal of flexibility back in the studio. I can use a mono on-axis, a mono off-axis, or a stereo version of the same recording, and can even tailor the stereo width to the center strength of the signal. All of that power is encoded in just two channels.

For those unfamiliar with MS recording, it uses two microphones placed at the same location. The mid-channel (also called the center-channel) microphone uses a cardioid or omnidirectional capsule, whereas the side-channel microphone has a figure-8 polar pattern (see **Fig. 3**). Matched pairs of mics are best, but any two mics will do the job, as long as they have the appropriate polar patterns. The mid channel is recorded on one track; the side channel is recorded on another. The side-channel track is duplicated, and one instance is flipped 180 degrees with respect to the other. The two side channels are then panned hard left and right, while the mid channel is set to the middle.

NICK PECK

FIG. 3: Any two mics can be used for mid-side recording, as long as they have the appropriate polar patterns.

You can take advantage of the MS technique for vocal or instrument tracks by automating the levels of the stereo and center elements to enhance the drama in the music. For example, you could use the mid channel alone for the verse, and then slowly fade in the side channels for the chorus. That will open up the track and build momentum.

For contrast, try using only the side channel for the bridge. Or, to create a subtle punctuation, use automation to bring in the mid channel on selected words or phrases. Those same techniques can bring an animated stereo image to electric guitars, keyboards, and sound effects (see **Web Clip 4**).

Riding the Fader

Dynamics processing has had a gargantuan impact on popular recording techniques over the past 40 years. Legendary boxes such as the Fairchild 670, the LA-2A, the 1176, and the Distressor have pumped, squished, and slammed tracks and whole mixes. The result can be cohesive mixes, big fat drums, huge bass tracks, and vocals that soar over everything else. On the other hand, you can have too much of a good thing.

Overcompression can lead to tracks and mixes that are highly squeezed, removing all trace of dynamics on both the micro and macro levels. Loudness is context specific; it has meaning only relative to softness. Contrast in music is a good thing. But because there is always the challenge of making vocals work over backing tracks that usually don't change much in volume, compression will always be with us.

The great engineers of yesteryear solved problems in level matching by riding the faders during recording and mixdown (see **Web Clip 5**). That may sound obvious, but if you are paying attention to the loud and soft moments of a vocal or an instrument during a tracking session, you can avoid the need for heavy dynamics processing by adjusting the input level as you go. Of course, that is much easier to do during overdubs than while tracking a whole band.

You can use automation to ride a vocal track's level during mixdown, increasing gain for whispered passages and reducing it for louder ones. If you can't react quickly enough to satisfactorily ride the levels, try playing back the track at half speed, assuming your DAW supports that. Your automation data will play back correctly at full speed, and your tracks' dynamic levels will rise and fall smoothly.

If you want to use some compression while still having a bit of manual control, automate the compressor's threshold control. Play through the piece varying the threshold, bringing in more or less compression at different moments. Once again, you can use this technique to create contrast between different sections of music, perhaps having an open, dynamic sound during the verse, and a squashed, beefy sound during the chorus.

LFO-like Audio Modulation

Low-frequency oscillators (LFOs) have been used to modulate synth parameters since the beginning of electronic music. You can stretch beyond the traditional uses of LFOs (controlling an oscillator's pitch or a filter's cutoff frequency) by creating LFO-like automation on digital audio tracks in your DAW.

The technique is easiest to apply on systems that have some form of draw tool that supports waveshapes, such as Pro Tools' pencil tool. But the technique is worthwhile even if you have to manually approximate LFO waveshapes, because you can apply it to any audio parameter you can think of.

Perhaps the most obvious example is panning. Using a sine or triangle waveshape, draw pan automation, bouncing between far left and far right at the highest possible resolution of automation. (To ensure that you get the highest resolution, turn off any thin track-automation preferences.) The result is a warbly, animated effect (see **Web Clip 6**).

You can apply the same technique to volume, creating rapid tremolo effects. Or, instead of volume, try using the mute parameter for a choppier effect (see **Fig. 4**). If your system can play back this kind of automation at more than 20 Hz, you can even get into the range of amplitude modulation, creating audible sidebands.

Try other shapes: square waves create sharper edges, and pulses of different levels with varying timing create turbulence and randomness (think sample and hold). Finally, you can just freehand it—you never know what kinds of weird effects you might get by scribbling with a mouse or a fader.

The ability to change effects parameters over time gives you myriad possibilities for experimentation. So dive into your DAW, pull up some plug-ins, and start exercising that fader finger. EM

EM WEB CLIPS

Nick Peck is a sound designer, engineer, composer, and Hammond organist living in the San Francisco Bay Area.

A practical guide to mid-side recording and processing.

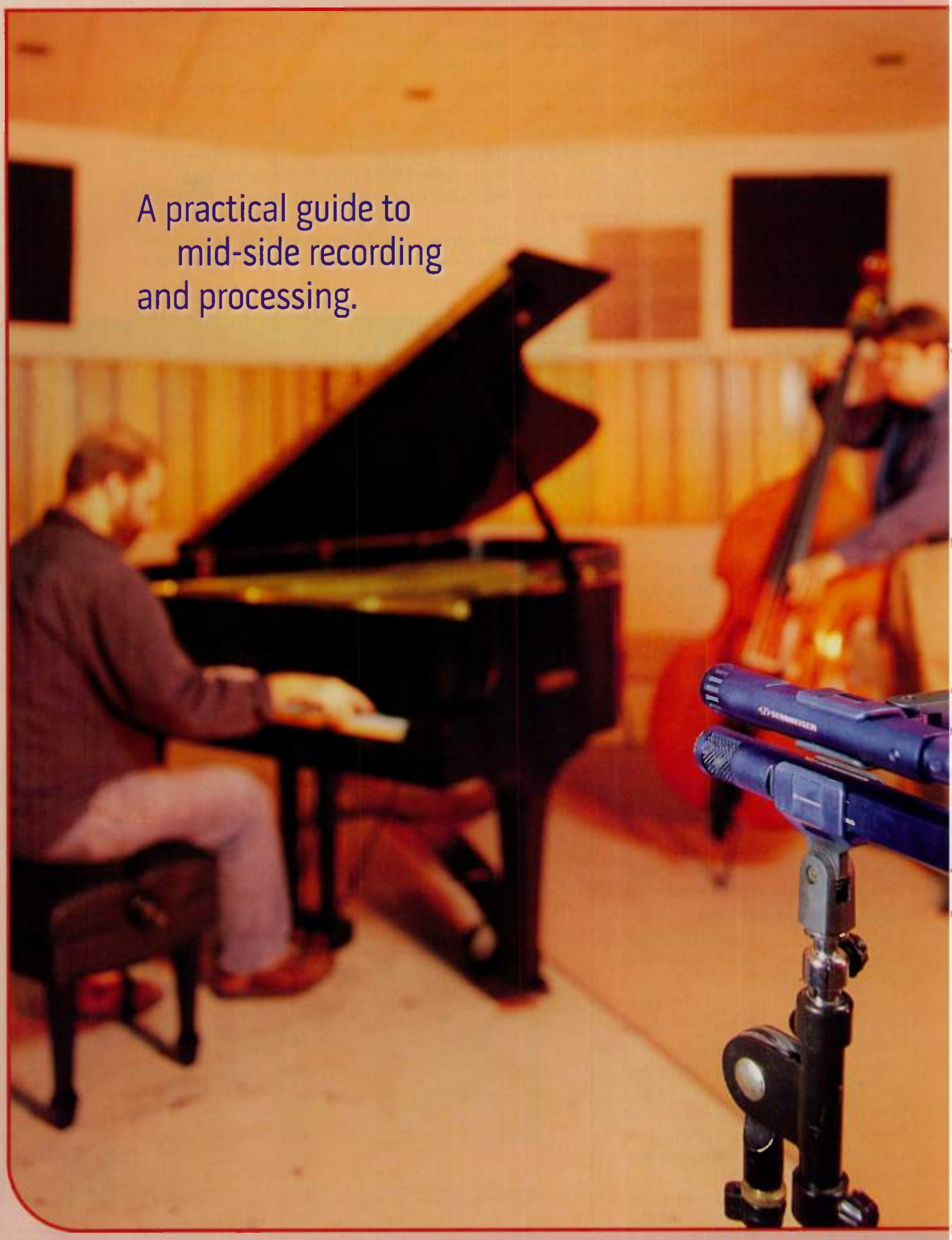


PHOTO BY ANTHONY PIGEON

Front and Center

By Brian Heller

When we think of stereo recording, left/right stereo typically comes to mind: two independent channels of audio, one carrying information from the left of the soundstage, the other carrying information from the right. There is another way to record in stereo, however—one that gives you plenty of options during post-production.

The *mid-side* (M-S) stereo recording technique puts the center of the soundstage (mid) in one channel and the side information in the other. These mid and side channels can be adjusted and reconfigured to accurately represent a conventional left and right stereo image (on their own, they do not give you a left/right image). The M-S technique gives you more control over the width of the stereo spread than other miking techniques do, and you can make adjustments at any time after the recording is finished.

Surprisingly, the M-S technique is one of the oldest concepts in stereo audio. The theoretical basis for it appears in Alan Blumlein's seminal 1934 patent, although it wasn't until the stereo experiments of Danish State Radio engineer Holger Lauridsen in the 1950s that the technology caught up with the theory.

Although M-S is most commonly used as a microphone technique, it can also be used on stereo recordings in a mastering situation. It has long been used in broadcasting, largely because properly recorded M-S is always mono-compatible. M-S continues to be a popular technique for studio and concert recording, and its convenience and flexibility make it a good choice for field recording.

Coincidentals

Although it may seem complex at first, the M-S miking technique is surprisingly logical and low-tech. The basic concept is that one microphone acts as the center, or *middle*, channel, picking up sounds from the front of the soundstage. Because the mid mic is aimed straight ahead and panned center, equal amounts of it are sent to the left and right sides of your mixer's main bus. Typically, a cardioid mic is chosen for the mid mic, but any mic can be used.

The side microphone must be bidirectional. When sounds hit the front of a bidirectional mic's diaphragm, it sends a positive voltage; the mic sends a negative voltage when sounds hit the back of the diaphragm. When using a figure-8 mic in an M-S setup, point the front of the mic to the left, so that one of the null spots is aimed forward, toward the sound source (see Fig. 1).

This position allows the figure-8 mic to pick up ambient and reverberant information coming from the sides of the soundstage, although the sides will have opposing polarities. Note that the bidirectional mic is not a stereo mic: if you turn the middle mic off, you will not hear independent left and right channels from the side mic.

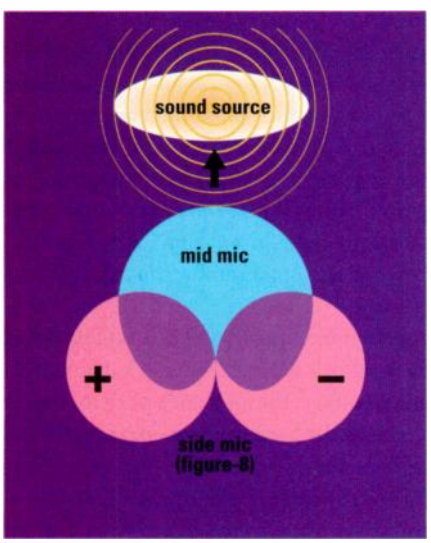
Like the XY and Blumlein techniques, M-S is a coincident stereo technique. The two mics should be as close together as possible—they are usually placed one on top of the other—to avoid any phase-related coloration (see Fig. 2). In many coincident setups, much of the soundstage is picked up off-axis, where a directional mic may not sound its best, due to mic position. With an M-S setup, each mic is optimally positioned for its polar pattern and function. A stereo mic that gives you both a directional and figure-8 pattern, such as the AKG C 426, makes M-S miking easy. The Shure VP-88, a stereo mic that includes an internal M-S matrix, makes it even more convenient.

The signal from each microphone is then recorded to its own track. For you to hear a proper stereo image when listening to the recording, the tracks need to be *matrixed* and *decoded*. (I will explain how these processes work in a moment.) Otherwise, you will hear the direct image in one ear and the room sound in the other.

Do the Math

Although you have recorded only two channels of audio—the mid and the side—the next step is to split the mono side channel into two separate channels. This gives you three channels—the mid signal (center channel) with the two sides—that you can balance to re-create a stereo image. For M-S to work properly, however, all three channels must be present in some amount.

FIG. 1: The middle-side stereo technique, also referred to as mid-side or M-S, employs a directional (mid) mic aimed at the source and a bidirectional (side) mic positioned to pick up sound coming from the sides. The signals must be properly decoded to create the final stereo image.



CHUCK DAHMER

M-S decoding requires a sum-and-difference matrix (see Fig. 3), where you add one side signal to the mid signal to get the sum, and subtract the other side signal from the mid signal to get the difference. To do this, one of the side channels is shifted 180 degrees in polarity: when the polarity-shifted signal is added to the mid signal, you get the difference between the two signals. Using M for mid, S for side, and –S for the polarity-shifted side, the math looks like this:

M + S = left channel
M + (–S) = right channel

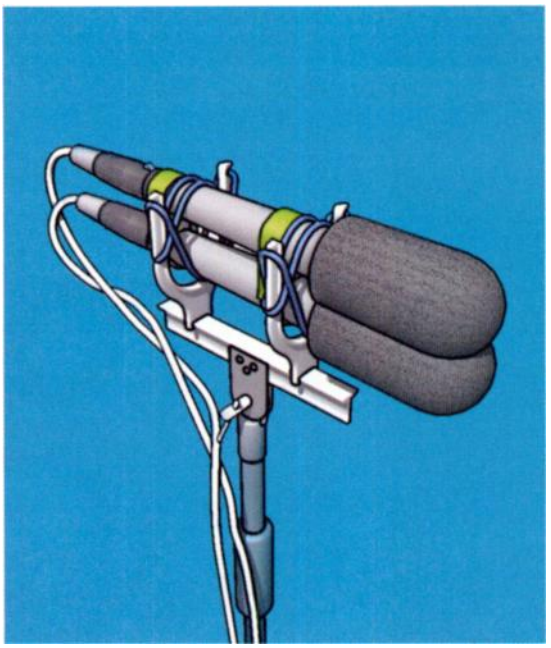


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FIG. 2: Typically, directional and bidirectional mics are placed on top of each other for M-S recording.

In other words, when two positive signals are added, you get a reinforcement of the signal—just like 1 + 1 = 2. When a positive signal is added to a negative signal, the signal is reduced—for example, 1 + –1 = 0.

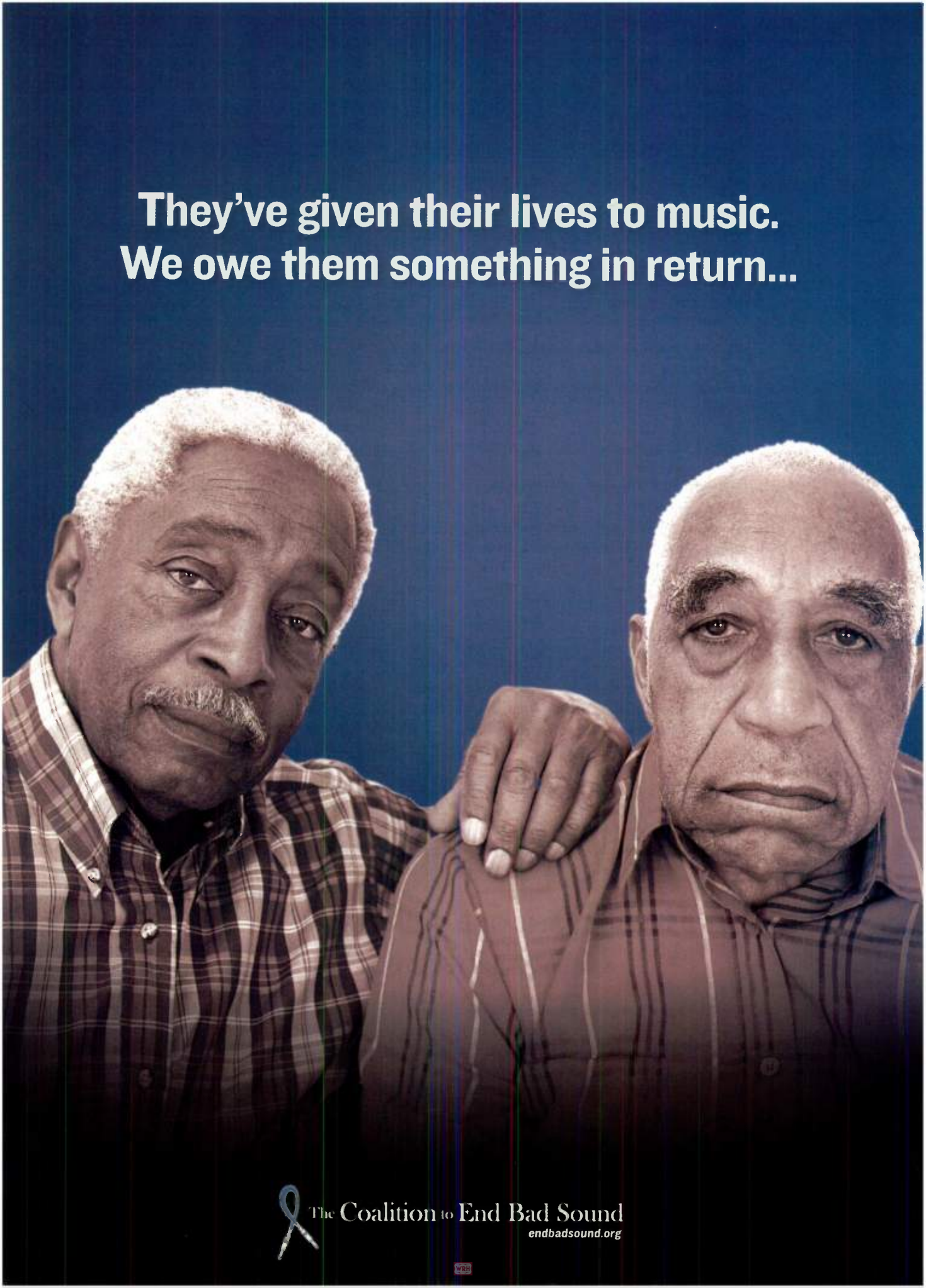
In M-S decoding, the left bus of your stereo mixer sums the signal from the mid mic (panned center) and side mic (panned hard left), while the right mix bus sums the mid mic to the polarity-shifted mic (panned hard right). The concept can be further clarified, albeit in an overly simplified manner, by studying how the mics treat sound coming from three different directions.

In Fig. 4a, an instrument in the center of the stage makes a sound at 0 degrees, which enters the cardioid mic on-axis but hits the null spot of the bidirectional mic. Because the cardioid is panned center, the signal is sent equally to the left and right mixer buses and speakers, giving you a centralized image.

In Fig. 4b, an instrument makes a sound at 45 degrees to the left. Because the front of the side mic is facing left, the mic's output has a positive polarity. The side mic's positive polarity adds with the positive polarity from the mid mic in the left mixer bus, resulting in imaging on the left side of the reproduced soundstage.

Over on the right side of the mix bus, the positive polarity from the side mic gets flipped to negative. When it is summed with the mid mic, the two signals with opposite polarity are combined, causing the sound level to be reduced.

In Fig. 4c, an instrument makes a sound at 45 degrees to the right, which also enters both mics. Because the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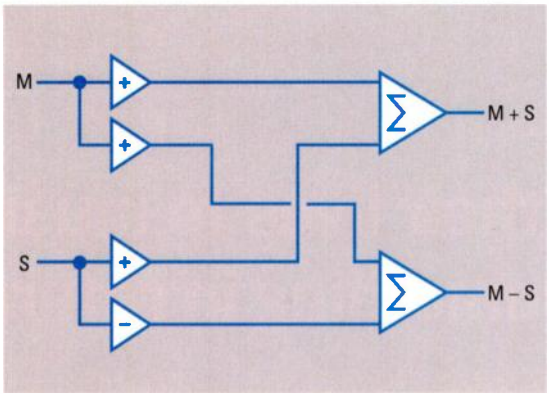

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FIG. 3: M-S decoding requires a sum-and-difference matrix that adds one of the side signals to the mid signal to get the sum, and subtracts the other side signal from the mid signal to get the difference.

negative lobe of the side mic is facing right, the side mic has an output with negative polarity. In the mixer, the negative signal is flipped in the channel that is panned to the right, making it positive. When summed in the mix bus with the positive polarity of the mid mic, it is reinforced and the sound is localized on the right. In the left channel, the negative polarity from the side mic is summed with the positive polarity of the mid mic, causing the sound level to be reduced.

Of course, in the real world the behavior of sound is a lot more complex when you consider frequency, room reflection, and phase. But these examples illustrate that, with the M-S technique, sounds are localized by differences in intensity at the microphones and by how the channels interact with each other electrically.

Decode Yourself

You can easily set up a matrix to decode M-S signals using a mixer. Some preamps, however, offer built-in M-S decoding, such as the M-Audio Octane and the Grace Design Lunatec V3. Standalone decoders are also available. Both types of devices usually accept mic- or line-level input from mid and side mics, matrix them internally, and give you a left/right stereo output. Typically, controls for mid and side levels are available on the box. A dedicated M-S decoder is especially handy in a remote- or a field-recording rig, because it allows you to monitor your signals without a mixer.

Although the signal path will vary depending on your equipment, the following is a generalized setup for recording and matrixing/decoding in M-S, using the mixer in Digidesign Pro Tools LE (see **Fig. 5** and **Web Clip 1**). Begin by setting up the cardioid and bidirectional mics at 90 degrees to each other. Arrange your microphones so that the cardioid is pointing directly at the center of the sound source, and the front, or positive, lobe of the bidirectional mic is pointing to the left, as in Fig. 1.

Next, split the output of the bidirectional mic and assign each signal to its own mixer input. Ideally, that is done at line level, if you have outboard mic preamps, but it can be done at mic level as well. The split can be done in the analog world by making an XLR splitter cable, or in the digital world by assigning the same input to two different channels, either in a DAW or in a digital console.

In the mixer, pan the cardioid mic to the center and pan the two channels of your bidirectional mic in opposite directions—hard left and hard right. Invert the polarity of the bidirectional channel that is panned to the right. Many consoles and mic preamps have a button labeled ∅ that does that. If not, you can easily make a short adapter cable that changes the polarity by reversing the hot (XLR pin 2) and cold (XLR pin 3) wires on the input end. (Be sure to clearly label the cable if you do this, so that you don't mistakenly use it later.)

In a DAW, you can use a plug-in to reverse the polarity of the signal, which I did in the Pro Tools mixer in this example using an RTAS plug-in. Now, assign all three channels to the mixer's main stereo bus, and your M-S decoder is complete.

Stereo Explorations

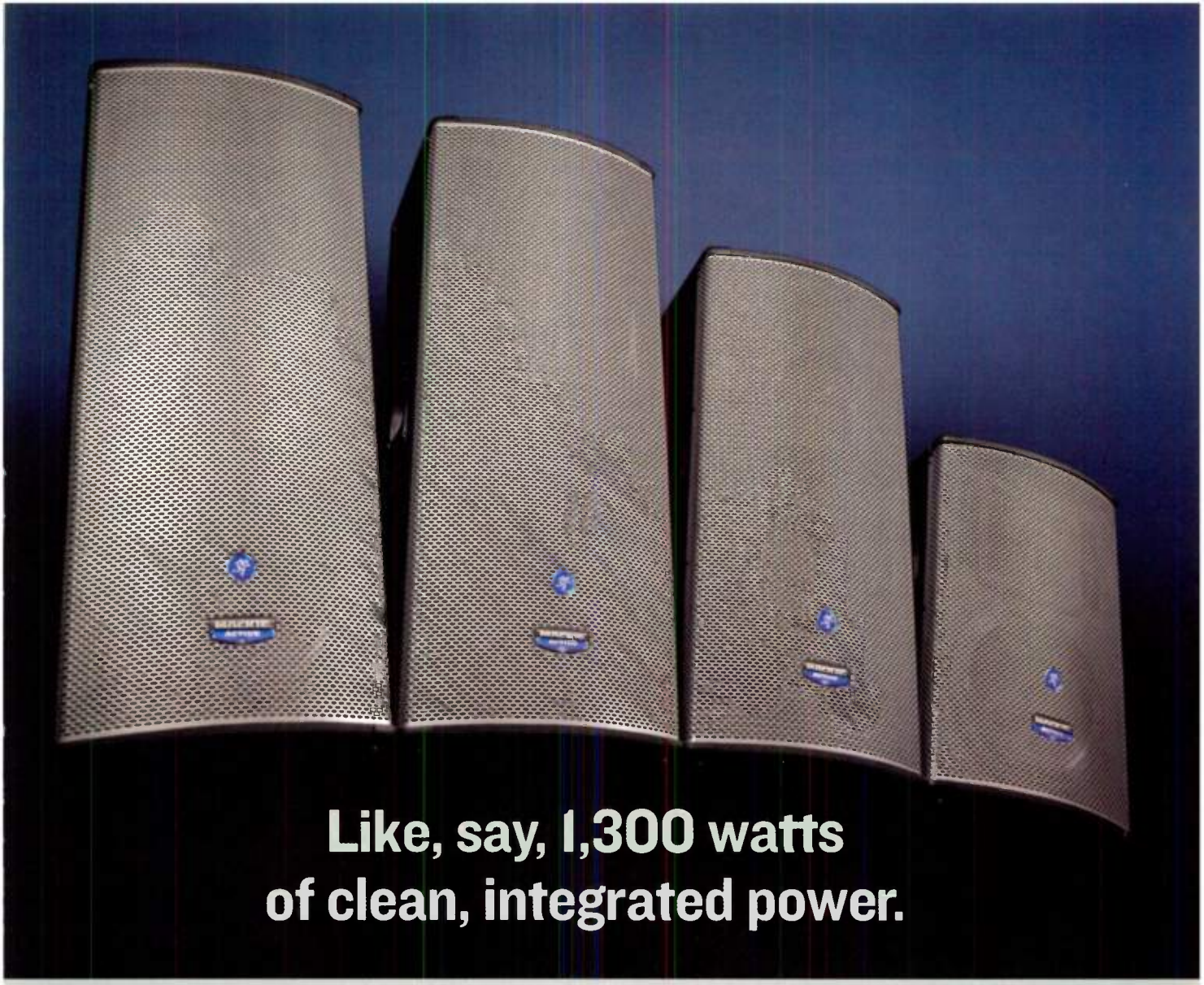
Once you finish recording, take a moment to explore the stereo image. Bring up just the cardioid channel on the mixer. It should contain more direct sound than room reflections.

While dropping the level of the cardioid, bring up the two bidirectional channels together. As you do that, you'll notice that the stereo image gets dramatically wider as the center sound moves farther away. Eventually, when the mid fader is completely down and the side faders are up, you'll hear a wide, out-of-polarity signal, made up mostly of room reflections. To verify that everything is connected and working correctly, you can pan the two side channels to the center, and they should cancel each other out.

> The M-S technique is one of the oldest concepts in stereo audio.

Neither of these extremes is an ideal stereo image, but they illustrate the different role that each mic plays in an M-S recording setup. By starting with the mid mic and mixing in the sides, you can find the right balance between the direct and reflected sound, as well as a nice stereo spread (see **Web Clips 2** and **3**).

As I noted earlier, one of the benefits of the M-S technique is that you can make these adjustments at



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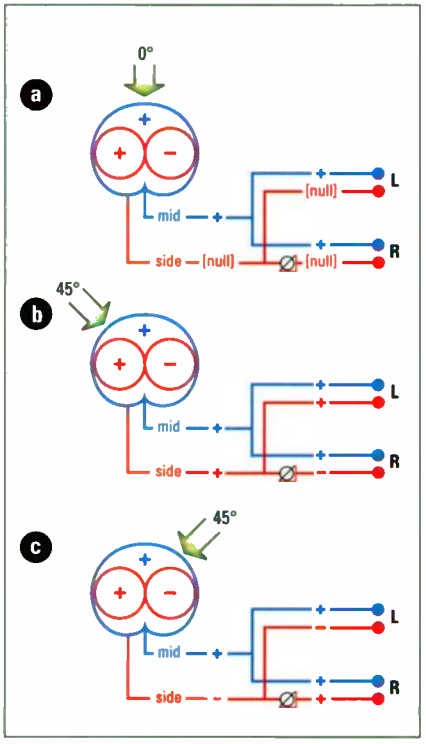
any time after the recording is done, without adding extra processing, such as EQ, delay, or spatialization. Consequently, M-S recording is great for situations where fine-tuning the mic setup is not practical, such as when you're field recording. And because the decoding process can happen at any time, M-S is a popular choice when monitoring in less-than-ideal situations. Simply record the mid and side tracks to the two tracks of your stereo recorder, and balance them later using your matrix and decoder.



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FIGS. 4a–c: a. An instrument in the center of the stage makes a sound at 0°, which enters the cardioid on-axis and hits the null spot of the bidirectional. b. An instrument makes a sound at 45° to the left, which enters both mics. Because the positive lobe of the side mic is facing left, the side mic has a positive output, as does the cardioid. On the right side, the positive polarity from the side mic has been flipped to negative. When it combines with the mid mic, the signal is reduced. c. An instrument makes a sound at 45° to the right, which enters both mics. Because the negative lobe of the side mic is facing right, the side mic's output has a negative polarity, while the cardioid mic is positive. In the side output with the reversed polarity, the negative signal is flipped and becomes positive. Then it is reinforced when it sums with the positive voltage from the mid mic. On the channel panned left, the voltage from the side mic remains negative. When it combines with the positive mid mic, the signal is reduced.

Back to Mono

As I mentioned earlier, an added benefit of M-S is perfect mono compatibility, because the stereo signal collapses to mono when the two bidirectional channels of opposite polarity are summed. In addition, the amount of reverb will also be decreased, which can add clarity when you are reducing a mix to mono. Even though other coincident techniques are theoretically mono-compatible, phase problems can still occur if the mics are not perfectly positioned. That is one reason why M-S has been used in broadcast recording for so many years.

An M-S setup also provides more options than other stereo techniques when it comes to microphone selection, because it does not require a matched pair. This gives you the freedom to choose different mics for the mid and side without negatively affecting the stereo image.

For example, in a particular situation you may prefer the clarity of a condenser for the mid mic, but the body and warmth of a ribbon for the side reflections. That kind of flexibility is great for solo instruments large and small (such as acoustic guitar and drum overheads). With large ensembles, however, which have a wider stereo image, the side mic may capture a significant amount of direct sound from the instruments. In that case, matched mid and side microphones are a good choice, in order to keep the stereo image even and stable.

Classical and jazz engineer Steve Bellamy, who is based at the Banff Centre in Canada, says he prefers M-S "on acoustic guitar and on piano recordings when I want to be close. When the source is large, like a piano, the M and S mics should be matched because they are sharing panning information. For smaller sources, like guitar, you can color the wider, spacious sounds differently than the center image simply by choosing a brighter side mic."

Everyone's a Critic

Of course, every recording technique has its trade-offs, and M-S is no exception. It is important to listen to how your recordings sound when using the mid-side setup technique, to make sure it is the right choice for the situation. If the microphones are not positioned well, no amount of tweaking will make them sound better. The best idea is to fix any problems before you record.

Begin by listening for sonic artifacts such as an imbalance between the mid and sides. If there is too much mid, the image will collapse to near-mono; if there is too much side, the image will sound unnaturally wide, unfocused, and phasey. Having to fight with the mix to get a good tone, balance, and imaging may mean that M-S is not the right choice for that recording situation.

Like other coincident techniques, M-S has been criticized for lacking the spaciousness that a spaced pair of omnidirectional mics offers. Although there are trade-offs with spaciousness (primarily in precise imaging), there are techniques that pros use to make the imaging of M-S more dramatic. For example, you can use an omni as the mid mic instead of a cardioid. That allows you to have the control that M-S offers, with the spaciousness of an omnidirectional mic. Using an omni in the center also gives you the benefit of an extended low-frequency response, which cardioids don't typically offer.

You can use any polar pattern you want for the mid mic—hypercardioid, omnidirectional, or bidirectional. Many engineers supplement a center M-S setup with an additional pair of omni mics further out in the room. That allows them to use the M-S setup for focus and the omnis for added spaciousness and width.

As part of his research in developing the Lexicon reverbs, David Griesinger suggests that the spaciousness of M-S can be increased with what he calls a "spatial EQ." Because low frequencies have been found to increase the level of the listener's envelopment in the sound (particularly when they are different in each channel), he recommends boosting signals below

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400 Hz by 2 to 4 dB in each of the side channels, often with a corresponding cut in the mid channel.

Dial M-S for Mastering

M-S is not only a recording technique, but also a way to process stereo audio. As a result, it's one of the mastering engineer's secret weapons.

M-S lets you separate the sounds in the center, which are common to the left and right channels, from the sounds on the sides, which do not share information with each other. For example, that allows you to change the balance of the final mix by encoding a pair of left/right stereo tracks to M-S tracks, processing or adjusting them independently, and decoding the altered M-S tracks back to left/right stereo.

How the M-S processing is used varies with the material at hand. The most common application is to

FIG. 5: An example DAW mixer set up to separate a finished left/right mix into M-S, adjust levels, and decode back to left/right. The Trim plug-ins are used for polarity reversal.

change the width of the stereo image, either collapsing it for mixes that are too wide by increasing the volume level of the mid, or widening narrow mixes by increasing the level of the sides. That technique works best in small doses, because a little can go a long way.

Once the tracks are isolated as mid and side components, they will each contain different instruments, which can be used to your advantage. For example, you can brighten up a lead vocal in the center without affecting the stereo percussion on the sides (see **Web Clip 4**); move a solo piano a little closer in the mix by minimizing the reverb from the hall; or add a small amount of compression to tighten up the rhythm-section elements in the middle, while not adding audible artifacts to the cymbals at either side.

Some mastering processors have built-in M-S processing capabilities. For example, the TC Electronic System 6000's BackDrop broadband noise reduction allows you to take a noise print and work on it in either left/right or M-S, automatically doing the encoding and decoding for you. With processes as program-dependent and touchy as noise reduction, the M-S option can make all the difference.

"M-S processing can help you control aspects of the mix that are hard to reach with standard left/right processing," says Greg Reierson, principal of the Minneapolis-based Rare Form Mastering. "The most obvious is widening or narrowing a mix, but other image-specific issues arise. I sometimes find I want to adjust frequency-specific content of the side channel without affecting the center, or vice versa. If done carefully, M-S allows more specific control of one channel without adversely impacting the other."

Several DAW plug-ins are available that convert M-S sources to left/right stereo and vice versa. That can save channels and mitigate error in DAW routing. A plug-in like the Waves S1 offers M-S processing—with and without spatial EQ—as part of its spatialization algorithms.

Be sure to listen carefully when making these kinds of mastering changes. There can be a tendency to overdo the side levels in hopes of creating an impressively wide image. Not only does this change the musical balance, but it could also leave you with phasey-sounding sides and a weak center.

Getting There and Back Again

The process for encoding left/right to M-S and decoding it back again requires a few more channels than M-S recording, but the tools and concepts are the same (see **Web Clip 5**). The goal is to separate everything that the left and right channels have in common from what is unique to one channel. That means reversing the sum-and-difference matrix used for recording.

Begin by making two copies of the original left/right mix. Duplicate them onto four mono tracks on a DAW or split them to four channels on the console.

Pan the left/right tracks from one copy of the mix to the center. If you like, send them to a mono subgroup in order to control them with one fader. That will become your mid, or sum, channel.

Next, pan the tracks from the other copy of the mix to the center, and reverse the polarity of the right channel. Send them to a mono subgroup, too, if you like. That will become your side, or difference, channel.

Now you have independent mid and side channels from a left/right mix, which can be balanced and processed independently. Because you'll want to listen in left/right stereo, take the mid and side channels and decode them as if they were M-S mics.

What you've done is add the two left/right channels together to make a mono signal that creates the mid channel. The side channel was also created by adding the left/right channels together. When the polarity of one of the channels is reversed, however, any sound that was common to both gets canceled.

You can prove the math by taking a left/right pair, encoding them as M-S, and then decoding back to left/right without making any changes. Assuming your gear is operating as it should, the original and post–M-S left/right tracks will be exactly the same. (To be sure, try flipping one pair out of polarity against the other.)

Summation

These are just the basics of the mid-side technique. If you want to get deeper into the technique, read The New Stereo Soundbook, by Ron Streicher and Alton Everest (www.stereosoundbook.com). It goes further into the details and is an excellent resource on microphones and stereo sound in general.

The best way to become adept at the mid-side technique is to experiment with it for recording and mastering. Armed with a bit of confidence in what M-S can do, you will have another powerful tool in your arsenal, and a new way to think about stereo. EM

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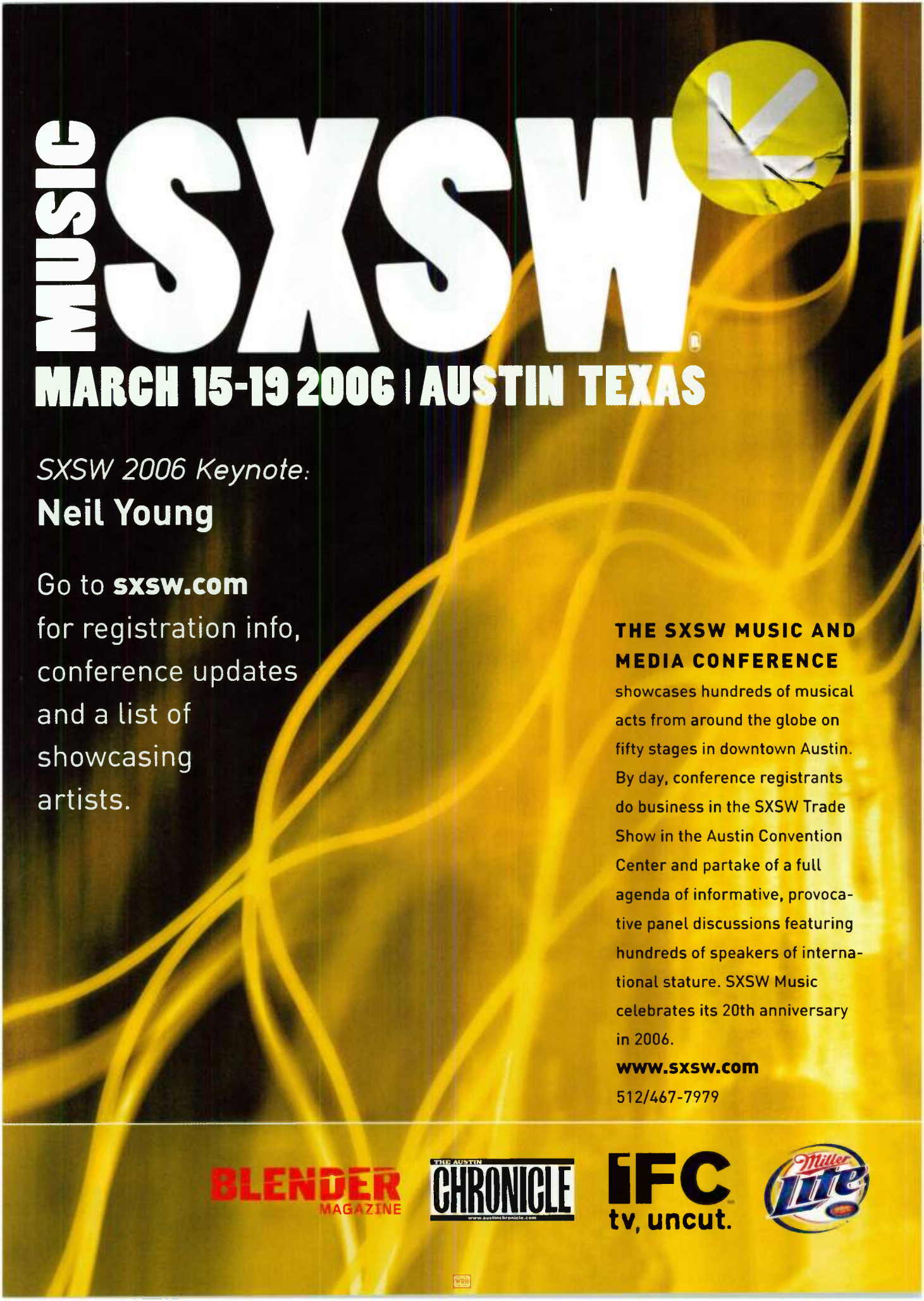
Brian Heller (www.whattheheller.com) is a composer, engineer, and tech based in Minneapolis. He teaches in the Sound Arts program at Minneapolis College but is currently staff engineer at the Banff Centre, in the Canadian Rockies. He would like to thank Greg Reierson of Rare Form Mastering and Steve Bellamy of the Banff Centre for their feedback.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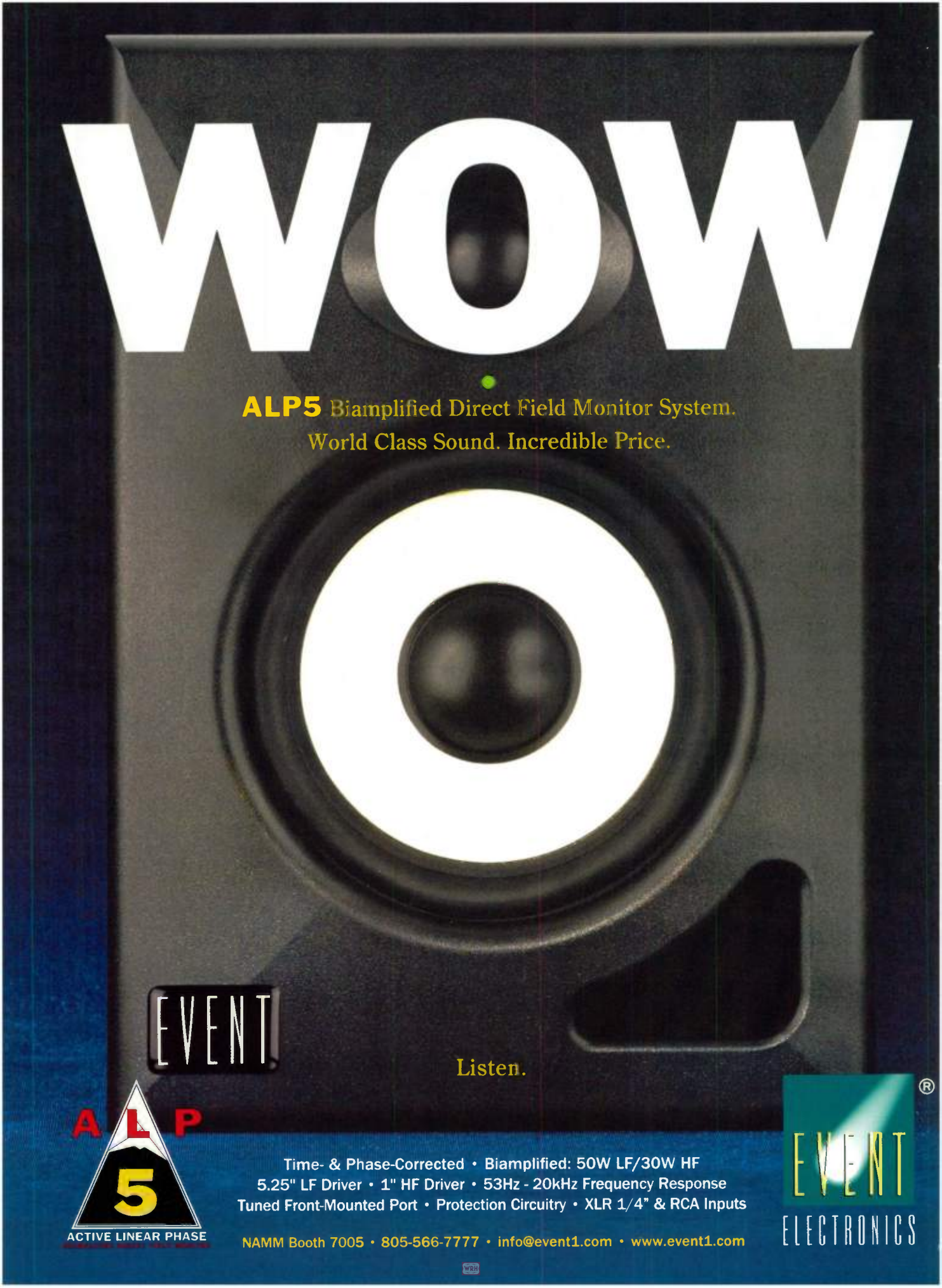
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MR. BONZAI

From Tangerine Dream to the Big Screen

By Mr. Bonzai

Paul Haslinger scores big in the world of movie music.

Austrian-born Paul Haslinger has built a reputation for composing film scores that incorporate robust classical elements and compelling electronica. Formally trained in his hometown of Linz, Haslinger ventured to Vienna after high school, where he continued his classical studies while exploring the new domain of electronic music. Auditions for the band Tangerine Dream led to a five-year collaboration, four albums, and a number of film soundtracks, including *Miracle Mile* (Columbia Pictures, 1988), *Near Dark* (Anchor Bay, 1987), and *Canyon Dreams* (Simitar, 1989).

After leaving Tangerine Dream in late 1990, Haslinger released three solo albums and scored two landmark animated science-fiction films, *Planetary Traveler* (Winsor, 1997) and *Infinity's Child* (Winsor, 1999). He continued perfecting his film-music skills as the programmer for Graeme Revell, supplying memorable textures and atmospheric style to films such as *Blow* (New Line Cinema, 2001), *The Negotiator* (New Regency, 1998), *The Siege* (Fox, 1998), *Pitch Black* (Interscope, 2000), and *Tomb Raider* (Paramount, 2001).

Haslinger earned his first solo credit as a film composer for the 2000 movie *Cheaters*, directed by John Stockwell. Since then, he has worked on Stockwell's *Crazy/Beautiful*, *Blue Crush*, and *Into the Blue*. He also composed and produced musical segments for Steven Spielberg's thriller *Minority Report* (20th Century Fox, 2002). In 2003, Haslinger scored Len Wiseman's

Underworld, his first film to open as No. 1 in the United States (see the sidebar "Paul Haslinger Soundtracks").

He recently completed work on *Far Cry: Instincts*, a new Microsoft Xbox game from Canadian developer Ubisoft. Haslinger's immense vision embraces film, games, albums, and live performance in a modern musical world without boundaries.

FIG. 2: Haslinger extols the virtues of Stylus RMX by Spectrasonics, citing the "unprecedented" control it gives him.

What is the first music you remember?

I don't specifically recall, but I'm sure it was the usual assortment of "music for children" that parents all over the world use to expose their offspring early on to really bad musical structure. The first time I actually paid attention was when I was six years old, and my sisters—eight and ten years older—took me to a band rehearsal. I remember being nearly deaf for several days from the noise. I decided right there and then that that was what I wanted to do when I grew up (I'm still waiting for that to happen!).

Who were your musical heroes early on?

Whatever my older sisters exposed me to—unknown Austrian bands, Ike and Tina Turner, the Who, Hendrix.

Who do you respect and admire today?

DJ Krush, UNKLE (James Lavelle), Jon Brion, Arvo Pärt, Funkstoerung, TerraNova, Kruder and Dorfmeister, Jon Hassell, David Sylvian, Timbaland, Sigur Ros.

Talk about your early musical education.

I started piano lessons at age six but didn't take music seriously until I was a teenager, when I thought about a career in music. I studied classical music, and my instruments were guitar and piano. I played keyboards in bands, and after high school I went to Vienna to study at the Academy of Music. I also became a session player, which culminated in my work with Tangerine Dream.

FIG. 1: Haslinger recently scored *Far Cry: Instincts*, an Xbox game by Ubisoft.

How did you get into electronic music?

I always had a fascination with electronic instruments. Part of the mystique was that nobody could afford them when I was growing up. But by the early '80s, one thing I could afford was an Atari 1040 ST, one of the first PCs to make music applications available to musicians. I guess I was in the first group of kids to take advantage of that. It was a little like learning a new language early in life. By 1985, when I met Tangerine Dream, I was the "computer kid" who could easily work with sequencers and synthesizers. I auditioned for their U.K. tour, and afterwards was asked to join the band.

Who was in the group at that time?

Traditionally, it is a three-piece ensemble. I joined Edgar Froese, the original founder, and Christopher Franke, and stayed with them for five years. I worked with them at their studios in Germany and Austria, and we also toured a lot during that period. Two U.S. tours and two European tours—it was a great way to get my feet wet, and to be exposed to the international music business.

A good portion of my work with Tangerine Dream at the time involved film music, and I remember approaching it as any 23-year-old would—without much fear or respect. Also, Tangerine Dream was typically asked to deliver a monochromatic kind of score, the electronic-analog trademark sound that TD had become famous for following landmark films such as *Sorcerer* [Universal, 1977], *Thief* [MGM, 1981], and *Risky Business* [Warner Brothers, 1983].

After moving to Los Angeles in the early '90s, I started looking into "music for picture" more seriously and in broader scope. My collaboration as a programmer and arranger with Graeme Revell exposed me for the first time to the full spectrum of film music, including the hectic demands of orchestral scoring and the power politics surrounding the finalization of any score for a major motion picture in Hollywood.

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Do you have a philosophy of scoring?

Watching a movie should be an experience. The music's most important role is to enhance, and in some cases even establish, that experience. To do that, a composer needs to understand the history and craft of storytelling and its application in film. And in this day and age you must be aware and have an understanding of psychological effect.

Other than that, it always helps to simply write a good piece of music. Keeping an open mind is equally important, because there are many ways to score a picture effectively. I always try not to make my position a singular one.

It's very emotional, isn't it?

Emotion has a lot to do with it. In Bach's time there was a theory of how music affects emotional states. It's interesting, because Bach is not thought of as being very melodic. He didn't use a lot of recognizable themes—there was something else that pushed the emotions. His music is more subtle and, for me, more effective. I feel that I am working in a long tradition of using music to change emotions.

It's important to note, though, that it's not good if you notice a film's music. If you don't notice it and the effect is created, that's what we are striving for. It involves understanding the psychology of storytelling and having skill with music to do it well. I love the fact that in film you are part of a bigger entity and that you are not out in front.

MR. BONZAI

In Haslinger's studio, this juxtaposition of laptops and an acoustic piano is emblematic of the mixture of electronic and classical influences in his music.

How do you prepare for your scoring work?

That is my favorite part of the job. I love to dive into projects, to research and explore the possibilities. I typically spend weeks establishing a sound and a musical vibe for a project—building project-specific sound libraries, experimenting with musical ideas—throwing paint against the canvas and seeing what sticks in context.

Do the rules of film scoring apply in scoring for video games?

There are many parallels, and the starting points are similar. At the core of both, there is storytelling. With video games, the difference is that you are working with nonlinear techniques and tools—working with loops, samples, with fragments, and multitracking.

In Vienna, when I was studying classical music, I would do little demonstrations of studio technology at the Academy. People were blown away because I could mix eight tracks in different ways and create completely different pieces of music. In scoring video games, I take musical fragments and combine them in different ways and create different effects. For me, that is the way I write, anyway. I love to experiment with modules, and recombine them to take on new appearances and forms. My way of working is paying off with games, and I am having a lot of fun.

Do you find game scoring as satisfying as film scoring?

Yes, but it depends on what you are working with. Just as there are boring films, there are boring games. I am working with Ubisoft now on a game called *Far Cry: Instincts* [see **Fig. 1**], which is an ideal situation. They give me the framework of the game and how the music will be applied, and lots of freedom with it. With film, you are working with a picture, but with games, there is only an assignment—a minute of action music, a minute of music leading up to the action. With film, you are a slave to the picture and the timecode. With games, it is more like animation, where the visuals may be created alongside the music.

Because there is a higher fragmentation of musical elements and, as you say, more "modularity" in games, what does that mean for the production process? It just puts higher demands on production organization, or *production architecture*. The number of software platforms, intermediate and final recordings, samples used, and so on, produces a data accumulation that can quickly turn into chaos. Add to that tight schedules and delivery demands.

With my crew, we typically run everything through a central database that can be accessed through a server or the Web pretty much from anywhere on the planet. That way we can all share and update data from various locations, and it gives everybody a clue

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4. PERFORM - Use the integrated GrooveMatrix to trigger individual loops, one shots, and MIDI patterns, or sets of loops and patterns. Great for performing or arranging.

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cakewalk

as to where we stand in overall production development. The key here is not only to have network and broadband communication access, but also to know how to use it most effectively for the purpose of the project at hand.

Do you notice aesthetic differences between Japanese games and those designed in the West?
There is a culture of animation in Japan that is unlike anywhere else in the world. Manga and anime have a lasting influence on everything else. There is even an art theory called Superflat. If the Western approach to art is vertical, the Japanese approach is more lateral; it goes toward a spherical feeling more than a pinpoint feeling.

I've always felt very inspired by and fascinated with Japanese art. It places an importance on emptiness and empty sound that I don't find anywhere else. More importantly, it uses that absence of sound just as effectively as sound itself and, in doing so, is closer to my own understanding of art as "choice of balance" rather than "genius of creation."

What did you learn from Tangerine Dream?
It's difficult to point out any one thing, since that was such an important engagement and time for me. But probably it was the "photographic" principle applied to music, which I first started to comprehend when working with them. Until then, I was under the impression that "creation" of music is all there is. After working with TD, I began to see "selection" as the other complementary principle to coming up with something, and one that is becoming even more important, as we are flooded with options afforded to us by advances in technology. You often have a lot of musical options, and there is a skill in selecting that one moment that is actually worth something.

We spent a lot of the time in the studio together, and because it involved producing, I was exposed for the first time to a world beyond being a session player. I began to understand the selection process involved in creating music, and over the years it has become more of an issue for me. How do you select the data and manage it?

DIGITAL DREAM STUDIO

Haslinger's studio (see **Fig. A**) contains nine networked computers. Of those, seven are PCs, the flagship of which is an AMD Dual Opteron 2.0 GHz. He also has a Mac G4 and a Mac G5. The various audio interfaces for all his computers are patched through his Yamaha DM2000 digital mixer and centrally clocked with an Apogee Big Ben master clock.

His primary sequencing software is Steinberg Nuendo 3.1, which he runs on the AMD CPU. "I ran Nuendo on a [Mac] G5 before, and I ran them parallel for a while," says Haslinger. "My impression was that I got more bang for the buck on the PC. So at some time I just switched over to PC."

His other six PCs are used for Tascam GigaStudio. He uses his G5 for running Mac-specific music software (such as Audio Ease Altiverb 5), and his G4 for his Digidesign Pro Tools|HD system (he's currently running v. 6.7). Although it might seem redundant to be running both Pro Tools and Nuendo, Haslinger has a definite division of labor between his two main DAW systems.

"The musical playground is Nuendo," he says. "It's all self-contained, and I have my own stock of files in there. The exchange platform to the outside world, more or less, is Pro Tools, simply for the fact that every music editor in town, every post-production [facility], everybody is on Pro Tools."

Haslinger typically slaves the Pro Tools system to the Nuendo system using MIDI Time Code. He loads the picture into Pro Tools and does his composing and arranging in Nuendo.


MR. BONZAI

FIG. A: There is no shortage of computers in Haslinger's studio. The audio from nine of them is routed and mixed through a Yamaha DM2000 console (far right).

He runs his Nuendo system at 44.1 kHz, because that's what his 2-terabyte collection of samples is saved at. He runs the Pro Tools system at whatever sampling rate the project dictates. He's able to easily move files back and forth between the two systems, thanks to a real-time sampling-rate conversion card installed in his DM2000.

"When I record audio and am working on a cue or working on a song and I want to record a stereo mix of it, I usually record it into both Nuendo and Pro Tools," he explains. "I record it into Pro Tools because there I have it in the project-specific sampling rate—mostly 48 kHz or something. I record it into Nuendo because there I've had it at 44.1 kHz, and can use it in the continuation of the project."

—Mike Levine

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Describe your studio setup.
My studio is basically one big computer network. I record and play certain instruments directly into the computer right here—guitars, bass, and so on—but I also love going to traditional recording studios and working in the vibe of that studio for a particular recording session. And L.A. is full of great studios! [For more on Haslinger's gear, see the sidebar "Digital Dream Studio."]

What is the primary software you use when you are scoring?
Steinberg Nuendo 3 with HALion 2 is my main writing and production environment. I apply most effects and mix treatments directly within Nuendo; I prefer the sound of Nuendo's audio engine over comparable sequencing products out there—it helps that it was written with surround sound in mind. I utilize HALion 2 as a quick-shot sample-management tool that allows me to pitch and adapt audio in real time.

What other music software do you use?
Steinberg Wavelab 5. The UAD-1 Ultra Pak from Universal Audio, the Max Bundle from Elemental Audio, and plug-ins from CamelAudio. All the virtual instruments from Spectrasonics, Applied Acoustics, and Native Instruments.

What's your favorite virtual instrument?
I have to single out Stylus RMX [see **Fig. 2**]; it is the most powerful and most versatile instrument I have in my arsenal. It offers an unprecedented amount of control and creative options for rhythm arrangements, and it allows import of REX files, which means it can be customized to even play tonal loops with all the sound control of its main engine. Amazing and very inspiring to work with.

Steinberg was bought by Yamaha, so Steinberg software and Yamaha hardware work nicely together. There is a software extension in Nuendo that allows direct control of the console. The mixer itself allows me to run a lot of live signal coming from samplers (outboard modules that I am still using) into a digital matrix where all signals can be routed according to project needs.

Do you have to replace tools every year?
Not as much anymore. Most of the changes happen in software these days. Think of it as a network, and you are changing parts off the network quite regularly. We are so dependent on orchestral emulation, which in programming terms is dependent on having a lot of colors available to us in real time. That means you need stacks of PCs running GigaStudio with all the stuff online and ready to go. I don't want to be loading up a sound when I have an idea—I want it ready to go. Being able to operate in real time is a major factor in catching the moment.

Any new stuff that helps you with your work?
There are no big surprises, but good-sounding technology and good-sounding software is finally here. My current new favorites: Stylus RMX, Nuendo 3, Eqium, Wavelab 5, Absynth 3, Tassman 4.

What is wrong with the music industry?
It's only partially the fault of the big bad corporations. The main problem is one of evaluation and judgments about music, which are going through a change. The music industry and the audiences haven't fully acknowledged that yet.

On top of that, there is an ideological disconnect because music doesn't quite "mean" as much anymore. Life is more experience based now, which is why live concerts are thriving. Hopefully, music can and will find new grounds and new environments, like the next generation of games, in which it can flourish and develop again past the narrow confines of the pop song. There is nothing wrong with pop songs—they just become overbearing as a format at times.

I think we are in a period of shifting paradigms of taste and judgment. In the past, we had critics and A&R people who would attempt to pinpoint what is good and what is bad. Today, I see this in flux. The PC laptop revolution has allowed many people to make their own records, whether they're good or not. I do think there is a lot of good music being made today, but it may be overshadowed by the huge amount of mediocre music. Where are the magazines, and where are the radio shows you can go to and find worthwhile work?

Who is your best musical friend?
Brian Williams, also known as Lustmord. He was a member of SPK, as was Graeme Revell. I met him in L.A. during the early '90s. We collaborate quite a bit, and we hang out and talk about music, film, and games. He has introduced me to a lot of music I wasn't aware of over the years, such as the Mo' Wax label out of the U.K., founded by James Lavelle. This led me to DJ Krush, one of the most influential artists in recent years.

PAUL HASLINGER SOUNDTRACKS

TV show
Sleeper Cell (Showtime, 2005)

Video game
Far Cry: Instincts (Ubisoft, 2005)

Films
The Girl Next Door (New Regency/Fox 2000, 2004)
Underworld (Screen Gems, 2003)
Blue Crush (Imagine Entertainment, 2002)
Into the Blue (Sony Pictures, 2002)
Crazy/Beautiful (Disney, 2001)
Picture Claire (Alliance Atlantis, 2001)
Cheaters (HBO, 2000)

Is there any filmmaker that you would like to work with?
Hideo Nakata, who made *The Ring* [Dreamworks, 2002] and is now working on a film version of one of my favorite books, called *Out*, by Natsuo Kirino. I also admire Akira Yamaoka, sound designer and producer of *Silent Hill*, and would love to collaborate with him in some fashion. Some of my other favorite filmmakers include Danny Boyle [*28 Days Later* (Fox Searchlight Pictures, 2002), *Trainspotting* (Miramax, 1996)], Jonathan Glazer [*Sexy Beast* (Fox Searchlight Pictures, 2001), *Birth* (Fine Line Features, 2004)], David Fincher [*Seven* (New Line Cinema, 1995), *Fight Club* (Regency Enterprises/Fox 2000, 1999)], and Gaspar Noe [*Irreversible* (Lions Gate Films, 2002)].

If you could go back in time before recording, what would you like to hear?
Different tunings. I'd love to hear J. S. Bach conduct his choir in Leipzig at Sunday Mass. Singers tune naturally toward each other, and it would be amazing to hear how Bach shaped that choir to tune and sound at that time.

Which film scores do you consider important steps in your development as a composer for film?
I feel *Miracle Mile* was the best of my scores with TD. It is also one of the most used temp tracks. Another one was *Cheaters*, the first film under my own name. At that time, I had spent ten years living in Hollywood, honing my skills in film writing and establishing a new vocabulary for my music. *Cheaters* was the first real sign that all that hard work was starting to pay off, and it was the start of a beautiful creative working relationship with director John Stockwell, who I've worked with ever since.

Any advice for getting a good start in the music-scoring business?
Try to start out working on the front lines, but in a noncombative position—that is, work with a composer as a programmer, a copyist, and so on. My work with Graeme Revell proved invaluably helpful, not just in working with orchestras and tight schedules, but more importantly because it allowed me to observe and learn how politics play out on high-profile projects. No school can prepare you for that.

What's coming up for you?
Freedom of Noise, a project about the connection between experience and memory. Over the years I've spent time in many cities, and one of my favorite things to do is go for long drives at night, somewhere between midnight and 6 a.m. I always find a certain tranquillity, a weightlessness, on these drives, and I can recall the experience, the vibe of these moments, very clearly, because they have attached themselves to my core memory. And there is a music that goes with that atmosphere—a music that is between worlds, between time, between circumstances. In happy isolation.

My plan is to release this as a collection of material. Initially, it will be a regular album release (CD and download), but I plan to expand its release cycle with image and text elements, and audio complements. Think of an updated version of the old radio-play format, popularized by such creators as Orson Welles. The cities I've chosen to select memories from for this particular film are Tokyo, Paris, and Los Angeles. Their common denominator is the vast differences of landscapes within these cities, the worlds within worlds, the ethnic multiplicity, and their sense of tranquil despair, which is so fitting for the times.

What music would you like played at your funeral?
Bach: BWV 82, *Ich habe genug;* Debussy: *Claire de Lune;* Bobby McFerrin: "Don't Worry, Be Happy." EM

Mr. Bonzai is an award-winning photographer and writer. His new book, Faces of Music: 25 Years of Lunching with Legends *(Thomson CoursePTR/ArtistPro, 2005), includes over 400 photographs and 160 interviews. Visit www.mrbonzai.com to learn more.*

Under the Surface

By Eli Krantzberg

Edit and automate using your control surface.

Your mouse and keyboard provide a great degree of precision, control, and accuracy for interfacing with your DAW. Hardware control surfaces, with their ability to simultaneously edit multiple parameters, can add an extra degree of expressiveness and creativity. In this article I'll look at some interesting ways of using a control surface in your productions.

I use a Mackie Control Universal (MCU) with Apple Logic Pro 7, but the suggestions here can be applied to most control surface and software combinations. If your hardware is not directly supported by your host application, two excellent pieces of software, LC Xmu and LC Xview (available at www.opuslocus.com), let you operate generic control surfaces in Mackie Control emulation mode with various audio-sequencing applications. (Both utilities require Mac OS X 10.3.8 or higher.)

Off to the Tracks

Using a control surface at the tracking stage is a good way to begin familiarizing yourself with the unit's various assignment and display modes. With Mackie Control, you can create tracks in your arrangement by holding down the Option button along with the corresponding Select button. You then assign physical inputs in Track Assignment mode. Group parameters can also be set for multiple miking situations, such as recording a live drum kit. Instrument Assignment mode enables you to scroll through and select from your available soft synths.

The hardware buttons cover all the basic transport and record commands. Cycle and Drop zones can be set by locating end points with the jog wheel and then pressing either the Cycle or Drop button, along with the Rewind button (for the left locator) or Fast Forward button (for the right locator). For punching in on the fly, controller assignments can be programmed for tasks beyond the functions of the transport buttons, like record toggle or record repeat.

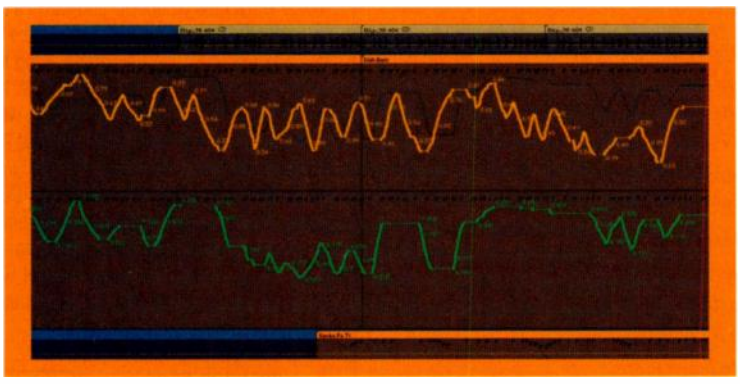

FIG. 1: Automation written in real time with two faders of a Mackie Control Universal. The faders controlled the filter cutoff and resonance of a bass line generated by Logic's EVOC plug-in.

Move to the Groove

For editing, a Mackie Control Universal and MCU emulation offer a fair bit of control for executing routine editing tasks. The jog wheel doubles as a scrub wheel, which, along with the built-in marker features, makes it easy to position the sequence at any point for editing. The eight function buttons, in conjunction with the Cmd/Alt button, give you access to basic cut, copy, and paste functions and various selection commands.

Once selections are made, holding down the Shift and Nudge buttons puts you into Large Nudge mode, where the V Pots can be used for moving selected objects to designated points (by either bar, beat, format, tick, frame, or sub frame) or to the current song position. The function buttons can also be programmed along with the other modifier buttons for user-specific tasks. Commands for splitting sequences at rounded and nonrounded song positions are useful for routine editing tasks.

Control surfaces really shine when accessing multiple parameters. For example, you can breathe life into static synth parts by automating parameters to give some movement and color to repetitive sections. On the MCU I used the following procedure to create this type of effect by simultaneously automating cutoff and resonance settings of a Logic EVOC plug-in (see **Web Clip 1**).

In Instrument Assignment mode press the V Pot button of the currently selected track to bring you into Instrument Edit mode. There the synth parameters are displayed horizontally across the entire LCD in multiple scrollable pages. The two-character Assignment mode LCD will display a dot at the bottom right to indicate the Edit mode, in which all V Pots can edit the visible parameters. Engaging Flip mode will allow you to edit the displayed parameters using the touch-sensitive faders, and "play" your controller to the music in real time to create free-form automation (see **Fig. 1**).

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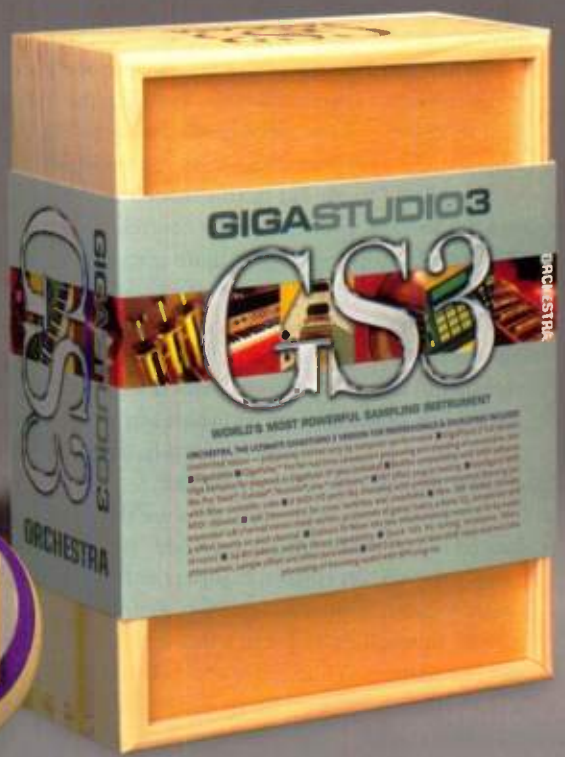

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Riding the Mix

Another multiparameter editing job that the MCU makes easier is fine-tuning EQ bands. Multi Channel view is a quick and easy way to access and edit one parameter at a time, but the real control comes when you switch to EQ Channel Strip view. In its default mode, the EQ parameters are laid out horizontally across the full LCD (see **Fig. 2**). All of the V Pots, or faders when in Flip mode, control the various parameters of a single EQ plug-in. You can sweep gain, frequency, and width amounts simultaneously, which offers obvious advantages over locating each individual parameter with a cursor.

When you hold down the EQ Assignment button, other options become available. Pressing the V Pot 7 button allows you to enter a mode similar to EQ Multi Channel view, where a single band can be adjusted for each available track. But in this mode the V Pots control the frequency while the fader controls the gain, and the mute button toggles the mute status of the individual band.

Holding down the EQ Assignment button in conjunction with the V Pot 8 button enables this same functionality in Channel Strip view, where all the EQ bands for a single EQ plug-in are accessible across the display. This feature is a great tool for quickly controlling and editing all the bands from one screen.

An interesting application is to use the control surface to sweep the frequency and width (also referred to as *Q*) together, on either a highpass or lowpass EQ, to create a resonant-filter sweep effect. For example, I took a small mix and routed it to a bus with EQ on. Then I put the controls in Flip mode and rode the faders, controlling frequency and width of a highpass EQ to generate some interesting nonlinear movement to the groove (see **Web Clip 2**). These types of effects are great for fade-ins, fade-outs, or breakdown sections.

FIG. 2: The LCD of an EQ in Multi Channel mode. All the parameters of a single EQ plug-in are displayed horizontally across the V Pots (or faders when in Flip mode).

Live at the Sends

Controlling multiple sends, either on a single channel or across multiple channels, can also yield gratifying musical sound-design results. A common technique used in many pop productions is to increase the reverb level as a track is fading out, to create the illusion that the music is moving farther away as it gets quieter. This is accomplished by subgrouping your tracks, and then setting up a send from the subgroup to feed the signal into a bus with reverb applied (see **Web Clip 3**). Using Send Assignment mode in Multi Channel view on the MCU is a perfect way to set this up. Global view lets you easily navigate to the bus and call up a reverb plug-in. Then you can simultaneously fade the track down with the fader while increasing the reverb-send amount with the V Pot.

FIG. 3: The Send Assignment display in Destination/Level Channel Strip view. In this mode, you can control all send slots for the selected track.

The same mode in Channel Strip view allows for editing several sends for a single track. This can be very effective if you have different effects set up on various buses (see **Fig. 3**). You can use your control surface as a real-time sound-design tool to shape the different effects sends over time. This can be particularly useful with rhythmic, time-based effects on the buses (such as delays, modulation, and filters synced to tempo).

I set up a drum loop on a track with four sends (see **Web Clip 4**). Each send was assigned to a bus with a different effect applied. I used Global Bus view and Plug-in Assignment mode to quickly load the plug-ins on the buses. I engaged Flip mode, so as to use the faders, and rode the different amounts of effects sends on the four buses in real time in order to create some intriguing rhythmic variations of the drum loop over time.

Control This

As feature packed as modern control surfaces are, there will be times when you come across unassigned or unsupported parameters. The MCU (or hardware operating in MCU-emulation mode) provides six programmable user modes for these situations. These are accessed by holding down the Shift button along with one of the Assignment buttons. Once in this user mode, adjust the application's onscreen controls for the actions you want the control surface to learn. Enable your host software's "learn" function, and move the V Pot to which you want to assign the function.

With eight V Pots and six Assignment modes, you can assign 48 unique parameters. Add to this the ability to program function buttons with different modifiers, and you have an almost unlimited degree of mouse-free control at your fingertips. Now, to reach nirvana, shut off your monitor! EM

EM WEB CLIPS

Eli Krantzberg is a Montreal-based drummer, vibraphone player, bandleader, and home-studio owner.

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Revisiting the LFO

By Brian Smithers

Realistic vibrato and other slick modulation tricks.

As a saxophonist, the first time I heard a keyboardist use LFO vibrato I almost choked. That was a long time ago, but in most synth patches, the LFO is still used for a steady pitch warbling that is laughable. In this article, I'll discuss how you can get more musical pitch modulation out of an LFO (using a bit of imagination), as well as other ways to broaden the LFO's horizons.

When an acoustic instrumentalist creates vibrato, pitch isn't the only thing that's varied. Timbre and volume also vary, and in different proportions, depending on whether the vibrato is created with the lip, diaphragm, wrist, or throat. You can emulate those playing techniques with a sine-wave LFO.

The Real Deal

You can create timbral variation by mapping the LFO to modulate the cutoff frequency of your synth's filter. Set the default cutoff frequency and modulation amount so that the effect is subtle—too much modulation will create a wah-wah sound.

For volume dynamics, map the LFO to modulate the amplifier's output level, but again, keep it subtle to avoid a tremolo effect. You might even mix in a little noise, keeping the level low enough to be noticeable at only the loudest point of the LFO cycle. Properly done, that adds a subtle breathiness to the sound and gives it an organic character.

Instrumentalists often slightly delay the onset of vibrato, allowing the tone to settle in first. If your LFO has a delay control (sometimes called *hold*), you can use that; otherwise, you might be able to apply an envelope generator with a slow attack to the LFO frequency or modulation amount. All these techniques are illustrated in **Web Clip 1**.

FIG. 1: The Voltage Processor on the Arturia ARP 2600 V lets you combine modulators and audio signals in novel ways.

Limitless Fun

A preprogrammed LFO will do the job, but there's no substitute for real-time expressive control. You can achieve that by using MIDI controllers. For example, mapping the Modulation Wheel to control LFO amount is a much more natural way to delay the onset of vibrato. If your synth has a MIDI Learn mode, setting up the mapping is a piece of cake. Even if you need to dive into the manual to find out how to set it up, it's well worth the effort.

In **Web Clip 2**, I mapped the same controller to LFO frequency and filter-modulation amount, so as I raised the slider, I got a faster and more pronounced timbral variation. The low static drone starts to shift slowly, morphing into a rhythmic pulse and then into a wild effect—all under the control of a single slider on my keyboard.

Beyond Vibrato

The Arturia ARP 2600 V's Voltage Processor allows you to combine modulation and audio signals in novel ways (see **Fig. 1**). In **Web Clip 3**, I set up oscillator VCO2 as an LFO to modulate the amplitude of oscillators VCO1 and VCO3, which are tuned an octave apart. I inverted the polarity of the modulation applied to the higher-pitched oscillator, causing the sound to morph between the two octaves. As a variation, you can modulate the amplitude of only one oscillator, so that one tone stays constant while the other comes and goes. It's a good idea to map a MIDI controller either to the speed or to the amount of the modulation; otherwise, the effect becomes monotonous on sustained notes.

Many programmers apply a slow LFO in small amounts to vary pitch in order to keep a patch from sounding too mechanical. If doing that makes the patch seem out of tune, try applying the LFO to only one oscillator in a multi-oscillator patch. The static oscillators maintain the pitch center, while the modulated oscillator introduces a variable sonic grind. Choose your LFO frequency and amount carefully to avoid obvious beating between the oscillators. You may also find that you prefer a triangle wave to a sine wave for that effect.

Although I've only talked about applying the LFO to pitch, filter cutoff frequency, and amplitude, any target is fair game. Most synths now offer some form of modulation matrix that lets you target a variety of parameters, including the settings of other modulators. Experiment a bit, and you'll be surprised by what you can get out of the lowly LFO. EM

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Brian Smithers teaches audio workstations at Full Sail Real World Education and music technology at Stetson University. He is the author of Sonar 5 Ignite! *(Thomson Learning, 2005).*

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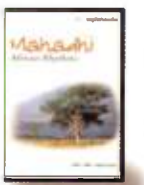

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Vive la Différence

By Mark Ballora

Dynamics, condensers, and ribbons—oh my!

We've all heard it: "If you want to do a job right, use the right tool for the job." If your job is to record something, your main tool is the microphone. Choose which one to use and where to put it, and 80 percent of the job is done.

But how do you choose the right tool, given the range of microphone designs and prices? Though mics have been covered extensively in these (and other) pages (see "More than the Sum" in the June 2003 issue of EM, "Smokin' Condensers" in the March 2004 issue, and "Ribbon Revival" in the November 2005 issue), I'll explore a few topics that should help you get a better sense of what mic will work for you.

At the Heart of It

Recording technology involves taking acoustic energy—air molecules oscillating back and forth—and mapping it to alternating-current electrical energy—electrons moving back and forth at a corresponding frequency and amplitude. Microphones accomplish this with a *diaphragm*, which is typically an extremely thin sheet of Mylar coated with gold, aluminum, or nickel. The diaphragm vibrates in response to air-pressure changes, and this motion is converted to electrical current with a *transducer*.

Transducers come in two varieties: *dynamic* (also called electrodynamic, electromagnetic, ribbon, and moving coil) and *condenser* (also called capacitor). Dynamic microphones operate by magnetic induction. This principle states that if you poke a conductor (such as a wire) into a magnetic field, you'll get a jolt of current in it. Pull the wire out, and you'll get a jolt of current in the opposite direction. Move the wire in and out continuously, and the result is an alternating current flow. In a dynamic mic, metal coils attached to a diaphragm extend into a magnetic field. As the diaphragm moves back and forth, alternating current is produced in the coils (see Fig. 1). Dynamic mics are sturdy, inexpensive, simple to design, and a good choice for onstage amplification.

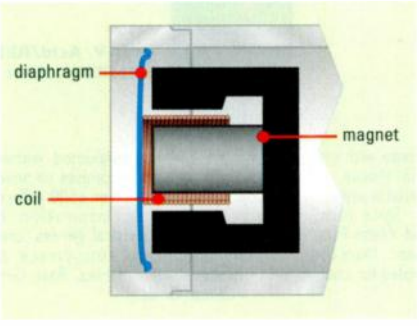

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FIG. 1: This image is a cross-section view of a dynamic transducer. As the diaphragm moves back and forth due to air-pressure changes, the coils (conductors) connected to it move back and forth within the magnetic field, producing an alternating current.

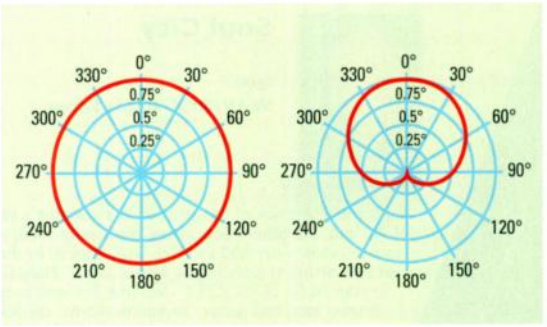

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FIG. 2: Microphone directionality is shown using polar plots that illustrate the mic's relative sensitivity to signals from all directions. Here are plots for an omnidirectional microphone (left), which is equally sensitive in all directions, and for a cardioid microphone (right), which is oriented to the front with some signal from the sides, and very little from the rear.

Condenser mics are more complex in design than dynamic mics. Unlike dynamic microphones, which generate their own current, condenser microphones require an external source of electrical current because they operate by means of a capacitor and therefore need to be charged. The current can come from a battery or from an external power supply in the form of *phantom* power.

Condenser mics are good at picking up sharp transients, such as those from pianos or drums, and they tend to have a wider frequency response than dynamic mics, making them the preferred type for use in recording studios. Historically, condensers were more expensive than dynamic mics, but high-quality, lower-priced models have emerged in recent years.

Point Me in the Right Direction

A microphone's pickup pattern, or directionality, can be seen in a polar chart. **Fig. 2** shows two common polar patterns: *omnidirectional* (left) and *cardioid* (right). Omnidirectional microphones, or omnis, can receive signals at equal magnitudes from all directions. They can be effective when used in groups, especially when recording an ensemble in a large room. The mics are often arranged in an arc, spaced a few feet apart. The instruments' signals reach the microphones at different times, and the phase differences among them create a nice diffuse spaciousness. (For more about phase, see "Square One: About Phase" in the May 2004 issue of EM.)

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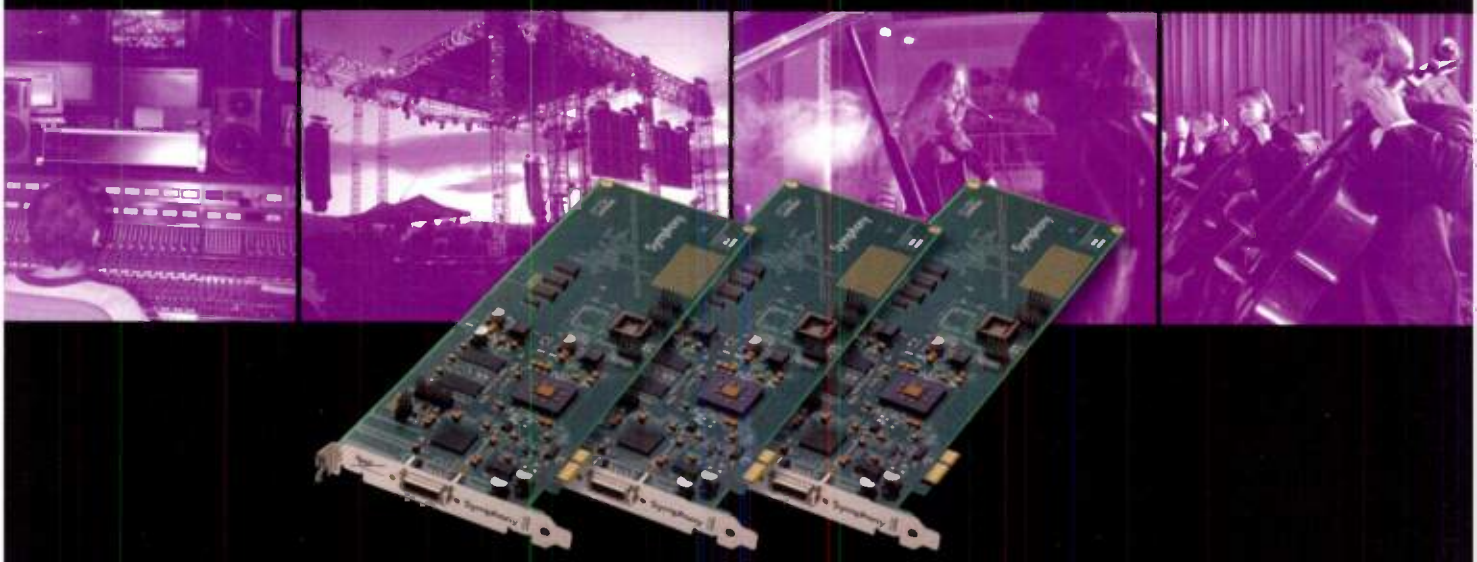

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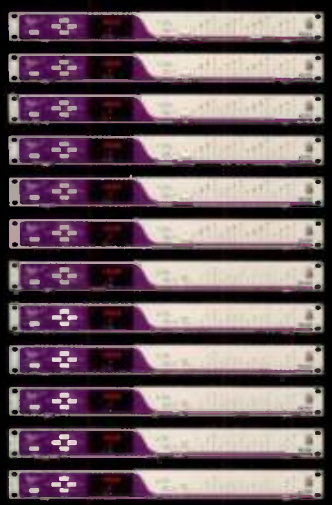

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Cardioid condensers are often used in a *coincident pair*, or XY, configuration, in which two mics are placed at the same location, one pointing 45 degrees to the right and the other pointing 45 degrees to the left. The fine pressure differences received by each of the mic capsules produce precise spatial imaging. The XY configuration is effective for recording small groups of instruments or for collecting location sounds. Stereo mics, such as the Røde NT4, make the process convenient by putting a coincident pair of diaphragm units on a single body.

Stereo recordings using coincident configurations, however, can lack the spaciousness found in recordings using spaced omnis. As a compromise, many engineers use an ORTF configuration, a near-coincident technique where the mics are separated by about 7 inches, each angled about 55 degrees away from center.

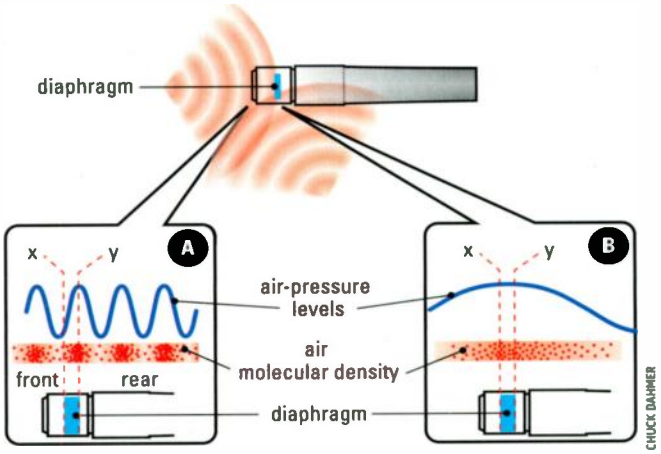

FIG. 3: Because the far side of a microphone diaphragm represents a longer travel distance for the wave front, the front and rear respond to the same signal, but with a difference in phase. A high-frequency signal (lower left) produces a big difference in pressure levels, while a low-frequency signal (lower right) produces minimal difference in pressure at the two sides of the diaphragm.

Getting Low Down

The frequency response of a directional mic depends on its construction. The output level depends on differences in pressure (called a *pressure gradient*) hitting the front and rear sides of the diaphragm. (In contrast, omnis have diaphragms that receive pressure changes on one side only.) The greater the pressure difference, the greater the microphone's output. The difference depends on two things: phase and intensity drop-off.

Phase differences are due to the time difference between when the wave first reaches the front of the diaphragm and when it refracts around and hits the diaphragm from the back. They are much more significant with high frequencies than with low frequencies (see Fig. 3). High frequencies, with their short wavelengths, may be at very different levels in their cycle on either side of the diaphragm. Note the difference in pressure shown between points x and y in the lower left of Fig. 3. This creates a big difference in pressure levels, which means the microphone has high output. But low-frequency pressure waves, with long wavelengths, create nearly equal pressure levels at both sides of the diaphragm, as shown at points x and y in the bottom right of Fig. 3. With minimal pressure differences, the output signal is not as strong. Thus, a cardioid or figure-8 has a natural tendency to roll off low frequencies and emphasize the highs. Mic manufacturers compensate for this by adding an output transformer that has a frequency response curve mirroring the mic's curve. The sum of the two produces a flatter response.

Intensity drop-off results from the longer distance to the far side of the diaphragm, and the drop-off is significant at close range. This is because acoustic intensity drops according to the inverse square law: basically, a change of an inch means a lot when you are close to the sound source but means next to nothing at a distance.

At close ranges, the added distance from the front to the rear of the diaphragm adds a significant percentage to the travel path. As a result, there is a stronger gradient because the intensity drops significantly by the time the wave front reaches the far side. This intensity difference affects all frequencies, including the low ones, meaning that the lows have greater presence at close range. When combined with the corrective curve of the output transformer, the result is a bass boost at close range. This is known as the *proximity effect*, or bass tip-up. It was exploited by crooners and announcers in the 1930s and 1940s with ribbon mics, a classic bidirectional dynamic design. They learned to make subtle adjustments in their distance from the microphone to give a husky sound to their voices.

Different types of cardioid mics have different frequency response curves built in. Most handheld dynamic cardioids, such as the Shure SM58, are designed to give a flat response at 4 to 6 inches from the source. If it's closer, there can be too much of a bass boost, resulting in a muffled sound and popping plosives. If it's positioned farther away, there can be a roll off of lows and a thin, AM-radio-like sound. On the other hand, small-diaphragm condensers, such as the Shure KSM137, Røde NT5, and Neumann KM 184, have a response that is designed more for distant positioning.

Are You Experienced?

Mic choice and placement is an art learned over time—like playing an instrument—and there's no substitute for experience. But with the basics presented here, you can get started on your road to recording virtuosity. EM

Mark Ballora teaches music technology at Penn State University.

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Have Knowledge, Will Travel

By Jeffrey P. Fisher

Parlay your know-how into seminar and training gigs.

Several months ago, this column looked at how to leverage your recording smarts into technical-writing gigs (see "Working Musician: Landing the Write Job" in the September 2005 issue of EM). This month, I'll examine another route for profiting from your knowledge and experience: training others through speeches, seminars, clinics, and college courses.

These sessions can range from a 15-minute trade-show demo to a three-hour training seminar to a course that lasts several days, weeks, or months. Some events pay only a token honorarium or travel expenses, while others—usually longer seminars and college courses—can yield decent money (see the sidebar "Seminar Pay Scales").

Beyond the pay, seminars often bring you indirect benefits such as product sales, postsession consulting work, and other music gigs. Teaching also builds your industry reputation and extends your network of contacts, both of which can pay dividends over time.

Opportunities abound for teaching others what you already know. You can speak locally on music topics at community events, clubs and organizations, park districts, and music stores. You could even host a session yourself. For example, if you're a studio owner, you might offer free or low-cost clinics on recording techniques as a way to bring potential new clients to your facility.

Regionally and nationally, the AFM, NARAS, ASCAP, and BMI all sponsor events and hold training and informational seminars. There are national trade shows such as AES and NAMM, as well as shows in related industries. Topics can range from music-career advice and promotion strategies to a host of technical subjects. Community colleges and technical schools offer production-related classes—both noncredit and for credit—that need teachers. For instance, at the College of DuPage, near Chicago, I teach Audio Production for Film and Television.

Hone Your Skills

To be an effective trainer, you need broad-based and extensive experience with your chosen topic. Some situations require that you have a college degree or other credentials, such as Pro Tools certification. But most trainers earn their reputation through real-world expertise and experience. You'll also need to develop good presentation skills.

In most cases, seminar attendees are there because they want to be. They want to hear your opinion and learn your techniques, and it's up to you to deliver the goods. Think of the most positive learning experiences you've had, and try to use similar methods.

Think of training as being like performing. You have to convey information in a pleasing, entertaining, and compelling way. Just as with a musical performance, rehearsal is the key to success. There are plenty of books on the subject of giving seminars, and you can learn a lot by watching others. However, nothing replaces actually giving presentations because the more you perform, the better you will become. Consider videotaping a few rehearsals or actual sessions with the goal of improving both your content and your presentation.

Presenting is demanding and requires lots of mental and physical energy. Strive to provide what the audience wants despite your own moods or other distractions. It can be exhausting because you are constantly "on." Between travel and lugging any necessary gear around, you're likely to have some long days.

LEAVE THE DOOR OPEN

Make sure to let attendees know how you can continue to help them after the session concludes. The real money is rarely made from the seminar itself. You can use your appearance to get new clients, additional music projects, other speaking engagements, consulting jobs, and back-end product sales. To do this effectively, collect contact information from attendees so that you can follow up later.

One method is to offer a "tip sheet" that complements what you presented, which you'll give to any attendee in exchange for his or her business card or contact info. Don't just parrot what you said in the seminar. Feature more detail about a specific topic or explore an adjunct to your main presentation. I have articles that explain compression and provide composition exercises to supplement my sessions on mixing techniques and scoring to picture. Be sure that your giveaway contains your contact information and an indication of other products and services that you sell.

"I Got a Deal with a Top New Age Label Because I Joined TAXI."

Chuck Henry – TAXI Member

I don't make what you'd call "mainstream" music. I like to call it, "Ethno-Electronica" with a jazzy edge. Record companies aren't exactly coming out of the woodwork trying to find artists like me, if you know what I mean.

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As if that wasn't enough, I've also placed music with several publishers through TAXI. And as a result, my music has been in TV shows, commercials, and films all over the world.

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So many people think that TAXI is just a songwriter organization, but it's actually much, much more. They help artists get deals, songwriters get cuts, and film, TV, and commercial work for writers and artists in just about every genre of music.

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Are you making great music? Do you have the connections to get it to the right people, or is it collecting dust on a shelf?

I wasn't making many solid contacts either. Then I joined TAXI, and my life literally began to change. I've made phenomenal connections, and landed just about every kind of deal imaginable. The most important part is that I'm making money doing what I love doing most – making music.

I don't know why every musician on the planet hasn't joined TAXI yet. It's exactly what you've been waiting for, and it's been sitting right under your nose since 1992!

Does TAXI sound too good to be true? Let me be the first to tell you that it's everything they claim it is – and a lot more. I even used TAXI's feedback to help me get the songs for my CD whipped into shape.

It's like having your own team of experts and collaborators at your disposal for less than a dollar a day. You probably spend more than that on designer coffee.

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SEMINAR PAY SCALES

How much you can make from a seminar depends on the particular circumstances. A one-hour session could pay anywhere from nothing to a few hundred dollars. An all-day seminar will likely range from a few hundred to maybe (big maybe!) a thousand dollars.

The pay for college courses depends a lot on the institution. Expect to make somewhere in the neighborhood of a few thousand dollars as a part-time instructor. If you're putting on your own seminars, the sky's the limit, but there's also the potential that you might lose money.

Find an Opening

I wouldn't suggest trying to develop your own seminars when you're just getting started. First, find built-in audiences by approaching organizations that already host training events. Let them do the promoting, registering, and handling of attendees, leaving you to focus on content and delivery.

Start with organizations that you're already affiliated with or have some knowledge of. If they offer training, look for the topics they've featured in the past and devise a course that is unique or fills a gap in their current lineup. If they don't offer training, propose a pilot program.

Find out who is in charge of booking presenters. Make contact by phone or email to gauge interest and to ask for class-proposal guidelines. Typically, your proposal should contain the program title and length, a paragraph or bulleted list about what you'll present, a recommended target audience and its skill level, and a brief bio that shows your credentials. Also include a promotional blurb about your seminar, and copies of any handouts, demo files, or CDs that you plan to use.

If you do put on your own event, consider teaming up with sponsors or other presenters. Music-equipment manufacturers may cough up some dough to have you promote their products for them during your presentation. Joint ventures with other speakers can spread the expenses around and bring in a wider audience. For example, a recording studio might team up with a graphic designer, duplicator, and publicist and present a course on creating and marketing an indie-music release.

Break It Down

When developing the content of your presentation, remember that people want information that is applicable to their situation. Therefore, use your knowledge to solve problems for them in a logical, practical way. Start by stating the problem or challenge, and then present step-by-step details for overcoming it. For example, in my scoring-to-picture sessions, I explain how I break the scoring process down into a predictable sequence of tasks. Obviously, you can present more detail in a longer session.

Playing before-and-after music examples will help to drive points home. You might deconstruct a recent project and then show how you pulled it together.

For a seminar on mixing techniques, you could play a song demo, switch to the raw multitrack, and then rebuild the final mix for the audience. Discuss your mixing approach, step-by-step. Be generous with your advice and don't leave anything out. At the end, play the finished mix for the audience and then take questions. When teaching how to score to picture, I replay the same video with four different musical soundtracks. I then dissect each approach and explain why it may or may not be the best musical solution.

People prefer active, hands-on learning to straight lecture, so try to involve the audience by soliciting questions, polling opinions, and doing group activities. Encourage attendees to share their experiences; tell your own only when they reinforce specific, practical points.

People attend seminars because they prefer learning that way. Prepare and give an entertaining, interactive live presentation. If you use audiovisual aids such as Keynote or PowerPoint, they should complement and augment your presentation, not be the whole thing.

Props are another way to involve your audience. Doing a session on recording techniques? Bring the mics, stands, and other accoutrements you need, and show specifically how you set up and record. Ask audience members for assistance, or set up hands-on situations.

Defeat the Butterflies

Most presenters get nervous before the event. The key to beating the butterflies is to be totally prepared. Thoroughly rehearse your presentation in advance. Audiences will forgive a little nervousness on your part if you look prepared and your content is good.

Focus on your audience and their needs. Poll them at the beginning for insight into what they are really expecting from your session. Tailor your presentation based on your findings, altering your content on the fly.

Start and finish at the appointed times, and ignore latecomers and those who leave early. Be wary of humor at other people's expense and stick to self-effacing examples. Be open to taking questions, or reserve time at the end for them instead of interrupting your prepared presentation. Don't fall prey to unexpected technical glitches. Check and recheck everything before the audience arrives. Also, have backups for crucial files and any software programs that you use. Most of all, enjoy your moment in the spotlight and the good feelings that come from helping others make their music and careers better. EM

Jeffrey P. Fisher's latest book is Cash Tracks: Compose, Produce, and Sell Your Original Soundtrack Music and Jingles *(ArtistPro/Thomson, 2005). He can be reached through his Web site at www.jeffreypfisher.com.*

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Sound Advice

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Their friendly and informative Sweetwater Sales Engineer also told them that by investing a little more money, they could get the Pro Tools LE Factory bundles—arming them with even more professional plug-ins. They jumped at the opportunity.

The band now uses BFD Lite on nearly every song to build super-realistic drum tracks. Green loves how the BF-3A compressor sounds on her vocals. Tiffany can't get enough of the Tel-Ray® Variable Delay on her guitars. And Richard digs the backing tracks he's been creating with Reason Adapted 3, Live Lite 4 Digidesign Edition, and SampleTank 2 SE. Run Don't Walk has never sounded better.

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CAKEWALK Sonar 5 Producer Edition (Win)

A Windows powerhouse breaks the 64-bit barrier.

By Allan Metts

Autumn is a wonderful time. Each year we get cooler weather, football season, colorful leaves, and, of course, a new version of Sonar. As usual, Cakewalk has found powerful new features to add to an already capable product.

This time around, Cakewalk has given us 64-bit processing, convolution reverb with hundreds of impulses, powerful vocal processing, more soft synths, and a host of other improvements (see Fig. 1). Also included is a second version of Sonar 5 that is optimized for use with the 64-bit edition of Windows. And all of this new capability comes at a price that is $150 lower than Sonar 4's.

FIG. 1: Sonar 5's interface has been enhanced and now includes a Piano Roll view directly in the main window. New envelope types have been added, and you can now assign icons to represent tracks.

Do the Math

Cakewalk explains the math behind its new 64-bit processing technology in a technical white paper on its Web site. In a nutshell, using the 64-bit double-precision audio setting reduces the calculation errors that can occur when audio number crunching is at its highest. Typically, this is when you are mixing tracks, or using software synthesizers and plug-in effects. What does a 64-bit end-to-end signal path sound like? I generally found the improvement to be subtle, and at times I couldn't hear any difference at all. Most people describe the improvement as an increased clarity, fullness, or smoothness, especially in passages with a wide dynamic range or complex harmonic content. The difference was perhaps the most noticeable for me when I called up some complex patches in the Pentagon I soft synth and fed them straight into the Lexicon Pantheon reverb. The sound was richer with 64-bit processing enabled than without it.

GUIDE TO EM METERS

5 = Amazing; as good as it gets with current technology
4 = Clearly above average; very desirable
3 = Good; meets expectations
2 = Somewhat disappointing but usable
1 = Unacceptably flawed

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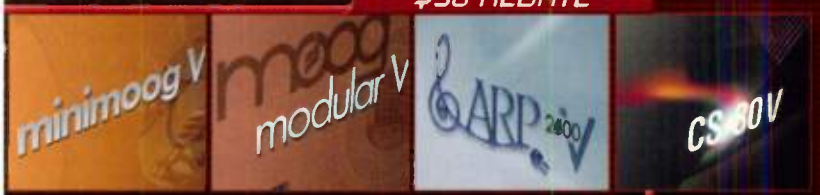

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The double-precision audio engine is available to all Sonar users, whether or not you're running a 64-bit operating system. On my 32-bit machine running Windows XP, I noticed about a 10 percent increase in CPU usage with the 64-bit mix engine switched on.

A Higher Order

Cakewalk has included a 64-bit copy of Sonar that requires a computer running Windows XP Professional ×64 Edition. On this platform, the 64-bit version is automatically installed. (Unfortunately, I wasn't able to evaluate Sonar on such a computer.) In addition to giving you access to 64-bit pointers and 64-bit floating-point registers (a subject that is beyond the scope of this review), running Sonar as a 64-bit application allows it to take advantage of much more system memory. Unlike 32-bit Windows, which supports less than 4 GB of RAM, Windows XP Professional ×64 Edition allows an application to use up to 128 GB of physical memory, and even more virtual memory.

How does more memory benefit Sonar? The program can access data held in RAM much more quickly than it can by grabbing data off the disk drives, so increased RAM can improve Sonar's overall performance considerably. Software samplers also benefit by having access to more RAM, because it reduces the need to stream samples from disk. Moreover, Sonar, like any application running under 64-bit Windows, benefits from a reduction in CPU utilization. All told, running Sonar under Windows XP Professional ×64 can provide as much as a 30 percent performance enhancement.

I should note that several significant features do not work with the 64-bit version of the program. These include ReWire support to 32-bit applications, 32-bit DirectX plug-ins, QuickTime, MP3 importing, and MPEX time-stretching and pitch-shifting. (According to Cakewalk, all of these issues will be resolved when the various manufacturers and developers update their technologies to be ×64 Windows compliant.) On the other hand, Cakewalk has developed an innovative technology called BitBridge that lets 32-bit VST plug-ins run under 64-bit Windows. In any event, make sure you are ready for life on the "bleeding edge" before jumping on the ×64 bandwagon.

FIG. 2: Sonar's V-Vocal processor provides high-quality pitch correction, formant control, dynamics adjustment, and time-shifting.

FIG. 3: Sonar's improved Piano Roll view shows controllers in the same pane as note events. You can edit both time and value with great precision by dragging the controller handles.

The Audio Melting Pot

Sonar 5 sports a number of enhancements in the audio arena. You can now include audio of different bit depths within a single Sonar project, and choose different bit-rate settings for importing, exporting, rendering (bouncing, freezing, and applying effects), and recording. For example, in the middle of a project, you could switch to a new sampling rate for recording, and everything you recorded from that point forward would be at the new rate.

Buses and synth tracks now display their audio waveforms during playback, giving you a visual representation of the audio you're hearing. These waveforms can be scaled just like audio tracks. All audio in the Track view (whether an audio track, a synth track, or a bus) now displays Peak Markers, which show the highest peaks found during playback. I found Peak Markers to be quite useful for identifying exactly where clipping was occurring.

There are several new offline processes, including a new Gain command that not only lets you boost or cut, but also lets you swap channels and invert the phase of either channel. In fact, you can mix any amount of the original left and right channels into the new audio. A new DC offset removal tool is also available, as is normalization to levels other than 0 dB.

Sonar 5 can stream video to an external device via FireWire, provided that device is supported by Microsoft's AV/C drivers; most modern devices are. You can use this feature to either reduce the processing load on your computer (by displaying the video using something other than Sonar) or export video to a digital recording device. In the latter case, Sonar provides a transport tool for controlling the remote device.

Being Vocal

Sonar includes a powerful vocal processor (called V-Vocal) that is based on Roland's VariPhrase technology. (For a discussion of this technology, see the review of the Roland VP-9000 in the May 2001 issue of EM.) You can use V-Vocal to perform pitch correction, alter dynamics or formants, and easily perform pitch-shifting or time-stretching on a phrase-by-phrase basis.

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When you right-click on a clip or selected region and choose Create V-Vocal Clip, Sonar will mute your selection and overlay a new clip in its place—one that represents the processing of your original selection by the V-Vocal processor (see Fig. 2). (Offline editing operations such as normalization and splitting cannot be performed on a V-Vocal clip, but you can move and copy clips.) You then pick from one of the processor's four editing modes: Pitch, Time, Formant, and Dynamics. Each mode plots your audio in the main screen and provides appropriate tools for navigation (zooming and panning) and making adjustments (selecting and drawing).

When viewing pitch, you see the actual detected pitches in red, the corrected pitches in yellow, and the center pitches (for determining where vibrato is centered) in white. You can draw in your own pitch correction and vibrato, and you can even fade in a vibrato to make it sound more natural.

Rather than draw the pitch correction, V-Vocal can automatically correct according to the parameters you specify, which include scale selection and allowable notes. You can dial in subtle pitch correction, extreme robotic pitches, or anything in between.

V-Vocal's time-stretching seems ideally suited to correcting slight timing errors in individual notes or phrases (although you can use it to adjust entire passages as well). Click the starting and ending points of the phrase you want to shift in V-Vocal's display, and vertical bars will appear at these points in your audio. To time-shift the passage, you drag a vertical bar: if you drag forward in time, audio in front of the bar gets compressed, and audio behind it gets stretched. Because this operation affects only what lies between the bar you're dragging and the bars before and after it, you can easily leave other phrases in place by adding more bars.

FIG. 4: Sonar 5 includes several great-sounding plug-ins. Shown here is the PerfectSpace convolution reverb, which includes impulse files for many interesting spaces.

You can also use V-Vocal to adjust formants and dynamics, which are controlled by drawing envelopes on the audio. The LFO tool, which is used to create vibrato in Pitch mode, can be used for periodic formant or dynamic shifts as well. There are also global adjustments for formant control: one to set it and one to control how strongly formant follows pitch. I could dial in everything from Alvin the Chipmunk to Darth Vader with the formant controls. After experimenting with a range of wacky formant-control options, I settled on a global formant adjustment that was slightly deeper and richer than my real voice.

Overall, V-Vocal is a very powerful addition to Sonar. I am not a singer, yet I was able to transform my subpar vocal performance into something downright respectable. And unless I took it to the extreme, I noticed no unpleasant artifacts with V-Vocal's time-shifting or dynamics processing. I only wish the tool were a bit more integrated into Sonar itself. In particular, I wanted the ability to specify the reference for pitch correction using a MIDI track.

MIDI Mayhem

Given that Sonar 5's ancestor was a MIDI sequencer, it's not surprising that there are plenty of significant improvements to MIDI editing in this version. A new Snap to Scale feature lets you restrict note placement to the notes found in a chosen scale for a chosen key (scale and key are selectable on a per-track basis). There are dozens of scales to choose from, and you can create your own.

In the Piano Roll view, note Velocities and MIDI controller events appear inside the piano roll itself (see Fig. 3).

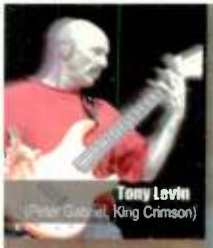
GETTING TO KNOW YOU

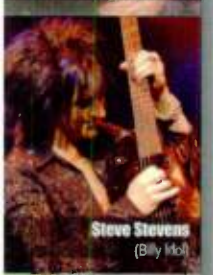
Any program of sufficient complexity can create a market for third-party instructional materials. I reviewed one such product while working with Sonar: Digital Music Doctor's *Sonar 5—Know It All!* course ($29.95; www.digitalmusicdoctor.com). This application provides a comprehensive tutorial on the use of Sonar 5.

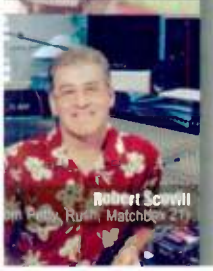
Sonar 5—Know It All! is available via DVD or download, and includes 22 lessons organized into four sections. The course takes about two and a half hours to complete, and you'll be spending nearly all of that time watching Sonar screen activity while listening to a voice-over. Though the activity-based approach is very helpful for learning how to use the program, I felt that the tutorials could have used a few more charts and pictures to show the concepts *behind* the program.

The training course covers all aspects of Sonar, including digital audio and MIDI, effects, loops, soft synths, and working with video. The plug-ins themselves are not covered to any significant depth, however, so don't buy this product if you're expecting to learn how to use the TTS-1 synthesizer or the PerfectSpace convolution reverb (some basic uses of Sonitus compression and EQ are covered, though).

Nevertheless, *Sonar 5—Know It All!* provides an excellent guide to using Sonar. Even experienced users will find something of value. I felt I already knew the product rather well, but I still picked up a trick or two by watching how someone else performed a specific task. The files used in the tutorial are included, allowing you to experiment further with the examples you see during the course presentation.







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Radial J48 Active DI - suggested list $199 US
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Radial JPC Stereo DI - suggested list $199 US
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This maximizes screen real estate and allows you to align controller events with notes. There are settings to determine which controllers are displayed, and an option to use the separate controller pane from previous versions.

Each controller event in the Piano Roll view has its own editing handle, which means you can change an event's temporal position or value. A floating window displays a real-time update of time and value as you make changes. Holding down the Shift key restricts your changes to either the time or value dimension.

Many of these same features are now available directly in the Track view when the Inline PRV (Piano Roll View) feature is enabled. With Inline PRV on, the track's MIDI clips are replaced with a piano-roll representation of the data. Nearby editing tools let you add or edit notes and controller events, and a zoom control lets you see just the range of notes you're interested in. (You can use the Fit to Content feature to set the zoom automatically.)

The Convenience Store

Several of Sonar's improvements are intended to make existing features more efficient. For example, you can create temporary Quick Groups by clicking the appropriate selectors in Track or Console view. Quick Groups let you change similar controls on multiple tracks in tandem. Once you're done, a single mouse-click decouples the controls.

There are a number of new track-management tricks: You can insert multiple tracks and set their properties simultaneously, and you can use Track Templates to bring in an entire group of tracks with their settings (such as bus routings, effects, and synths) preconfigured. Tracks can also have icons attached to make them easily distinguishable. Dozens of music-related icons are provided, or you can create your own with nearly any image editor.

PRODUCT SUMMARY

CAKEWALK Sonar 5 Producer Edition

digital audio sequencer
$799
MPEG encoder license (required after 30-day trial) $19

PROS: 64-bit processing engine. Efficient user interface. Powerful vocal processing. Excellent plug-ins. Lower price than previous version.

CONS: Vocal processor doesn't allow MIDI as a pitch reference. Not all plug-ins have comprehensive documentation. A $19 fee to register the MPEG encoder is required after a free 30-day trial.

FEATURES
EASE OF USE
AUDIO QUALITY
VALUE

MANUFACTURER
Cakewalk
www.cakewalk.com

In the "Why didn't they think of this before?" department comes the Add Nodes at Selection feature for envelopes. I often have the need to raise or lower an envelope over just the range of time that I have selected, and this feature makes easy work of it. Also new is the ability to draw envelopes in a variety of shapes (sine waves, square waves, random changes, and so on) and sync the shapes to a project's tempo.

There are many other improvements in this version of Sonar—among them, step-recording improvements, seamless integration of VST plug-ins, effects bins on individual clips, and a host of usability improvements. My favorite usability enhancement is Sonar's ability to turn any of its floating windows into a tabbed pane in the Track view. Once you've opened more than a few effects or alternate views, screen clutter becomes a serious problem. When you make part of the Track view multitabbed, the clutter problem goes away.

And then there are the plug-ins: Sonar has always shipped with lots of extra goodies in plug-in form, and this version is certainly no exception. Among the new soft synths (each of which could warrant an entire review) are the PSYN II subtractive synth, the Pentagon I vintage analog synth, the Roland GrooveSynth, the SFZ SoundFont Sampler, and the RXP REX Player groove box. The Cakewalk TTS-1, Cyclone, and DreamStation DXi2 synths from previous Sonar versions are still here.

The excellent suite of plug-in effects from Sonar 4 is still in place, including the Lexicon Pantheon reverb and the full set of Sonitus processors. A notable addition is the PerfectSpace convolution reverb (see Fig. 4), which comes with numerous impulse files that simulate spaces as diverse as cathedrals, mine shafts, and the inside of a piano. (See the article "Trading Spaces" on the EM Web site, at www.emusician.com, for the rundown on convolution for reverb and other purposes.)

All in all, Sonar 5 represents a solid upgrade. As usual, the product's documentation is complete, and the online help is context sensitive, although not all of the plug-ins are documented as thoroughly as the main product (see "Getting to Know You" for a description of a third-party support product). If you're an existing Sonar user, especially one who deals with vocal tracks, you'll want this upgrade. And if you're a new shopper in the DAW marketplace, give Sonar a good, hard look. It certainly holds its own with any other modern high-end DAW product.

Allan Metts is an Atlanta-based musician, software/systems designer, and consultant. Check him out at www.sonicbids.com/AllanMetts.

FIG. 1: Korg's D3200 has a 16-pot Knob Matrix (located directly under the LCD screen) that performs various functions depending on which mode is selected. The knobs make it easy to work with the onboard drum sounds and effects.

KORG D3200

A new level of personal digital recording.

By Rusty Cutchin

After producing popular entry-level portable digital studios such as the D4 and D1200mkII, and then making a splash at the deep end with the D32XD, Korg has found fertile middle ground with the D3200. The unit should appeal to those who want a complete studio in a box but don't need every last bell and whistle. The D3200 offers a comprehensive feature set for basic live and studio recording, mixing, and burning. The recorder delivers pro-quality sound along with convenient portability.

The D3200 offers an extensive set of editing and mixing features.

A combination of traditional and unique controller tools on the D3200 (see Fig. 1) makes it easy to use its many features. You get a jog wheel (the Value Dial) for basic data entry and scrubbing. The Edit Controller section has a fingertip joystick (the ClickPoint) and several navigation buttons for quickly positioning the cursor on the tilting 320 × 240–pixel LCD screen. The most welcome tool is the 16-pot Knob Matrix, a 4 × 4 grid of controls with functions that change according to the assignments that are onscreen. It provides an elegant way of quickly reaching the D3200's many options.

Into the Matrix

The D3200 gives you 16 tracks of simultaneous playback and as many as 12 tracks of simultaneous recording at 24-bit, with 44.1 kHz or 48 kHz resolution. If you drop it down to 16-bit (either 44.1 kHz or 48 kHz), you get as many as 32 tracks of playback and 16 tracks of simultaneous recording. The D3200 also gives you eight virtual tracks for each playable track (even the stereo mixdown tracks), so a total of 272 tracks can be stored at one time.

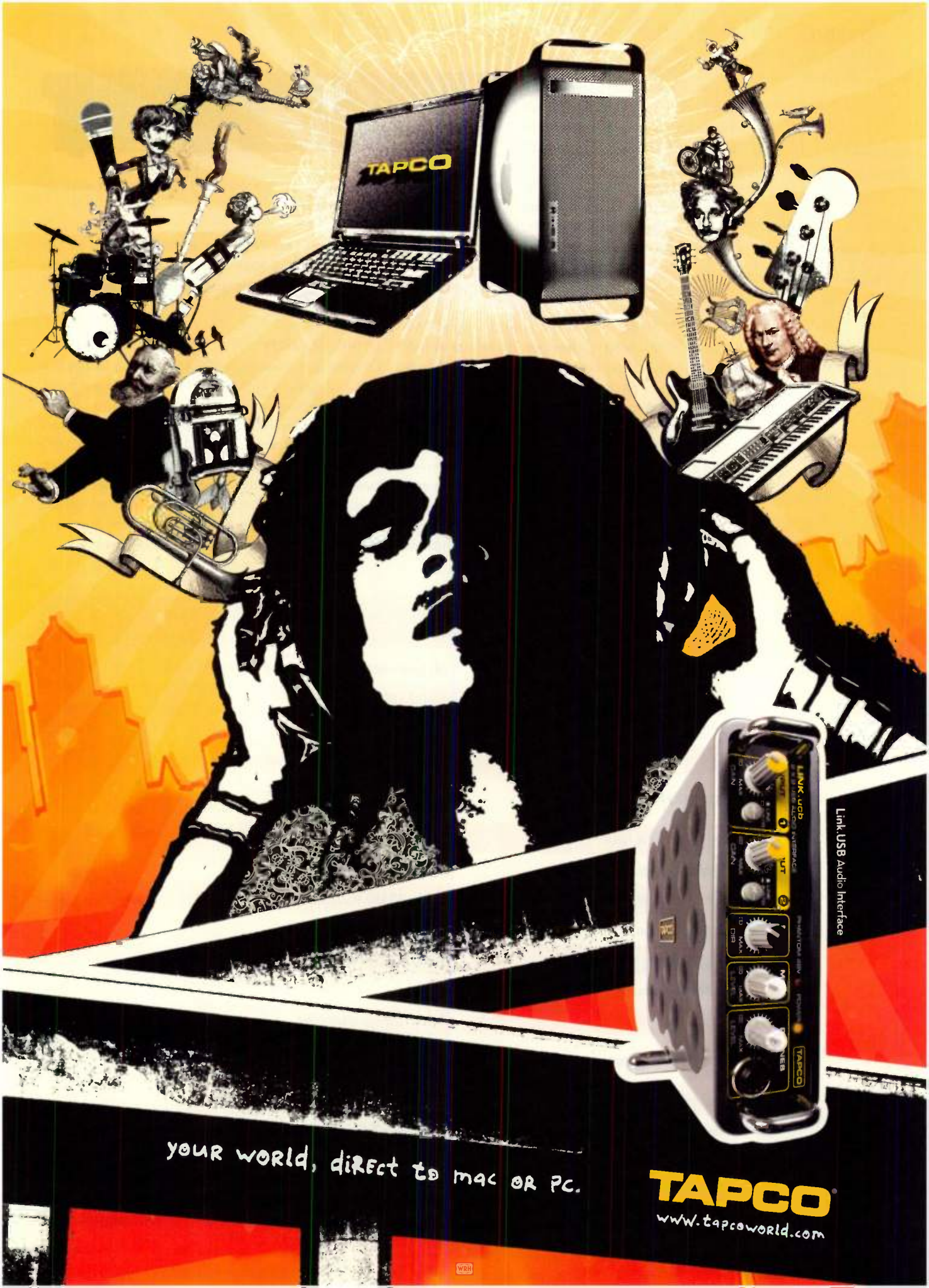
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FIG. 2: The D3200's rear panel contains MIDI and USB ports, jacks for a footswitch and expression pedal, and S/PDIF connections.

It's easy to set up a multiple-mic session online with the D3200. All 12 of its input channels have balanced ¼-inch connectors. Eight of the channels have XLR jacks with 48V phantom power, which can be switched on or off independently, allowing you to use condenser and dynamic mics on the same take. Each of the 12 input channels has a –26 dB pad switch and trim pot. A dedicated guitar jack on channel 1 lets you plug your axe in quickly and take advantage of the unit's onboard effects.

The D3200 has 52 effects algorithms and uses Korg's REMS modeling technology. The effects include a range of typical processes such as reverbs, delays, modulation and pitch-shift effects, and special effects in mono and stereo groupings. Stacked multi-effects are available in mono.

The Knob Matrix gives you convenient control of the effects, which can be applied in several ways. Insert effects can be separately applied to as many as eight simultaneous channels. You can also have two Master effects (auxiliary or bus) that are accessed from the individual channels' effects sends. In addition, a Final effect can be applied globally on the master bus. You have access to 128 preset effects programs, 128 user programs, and 32 user locations within each song.

Drum Session

Composing is aided by the D3200's Session Drums feature. You click one button to enter Session Drums mode, and then use buttons and the Knob Matrix to access the collection of loops built into the unit. The Group and Session knobs let you choose the overall musical style (everything from a simple metronome to complex electronica loops with odd time signatures), and the Variation knob lets you zero in on a pattern. You can access fills and ending patterns with one click and fine-tune the feel of patterns with the Shuffle and Humanize knobs.

D3200 SPECIFICATIONS

Analog Inputs	(8) balanced XLR; (12) balanced ¼" TRS; (1) unbalanced ¼" TS
Digital I/O	S/PDIF, USB 2.0
MIDI I/O	In/Out
Additional Ports	expression pedal input, footswitch input
Frequency Response	44.1 kHz: 10 Hz–20 kHz (±1 dB) 48 kHz: 10 Hz–22 kHz (+1/–2 dB)
Signal-to-Noise Ratio	103 dB
Dynamic Range	103 dB
Total Harmonic Distortion + Noise	20 Hz–20 kHz (0.02%)
A/D Conversion	24-bit, 64× oversampling
Input Impedance	4 kΩ (XLR); 10 kΩ (TRS); 10 MΩ (instrument)
Output Impedance	150Ω
D/A Conversion	24-bit, 128× oversampling
Sampling Frequency	internal: 44.1 kHz/48 kHz external: 44.1 kHz/48 kHz (±6%)
Record/Playback Resolution	16-bit/24-bit
Dimensions	21.5" (W) × 4.5" (H) × 14.6" (D)
Weight	23 lbs.

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For each preset pattern you can choose any of the unit's built-in drum kits. Within each kit you can modify the pan, level, and tuning of each instrument. A dedicated fader next to the master fader controls the overall volume of the Session Drums section.

You can string the patterns into sequences that play along with the recorded audio. The audio from these sequences does not have to be recorded onto tracks and can be mixed down internally along with your recorded tracks. Many of the internal drum sounds are processed and sound good enough for a final mix (see **Web Clip 1**). If, however, you wish to use the D3200's insert effects or any external effects on your drum parts, you can record the drums onto their own audio tracks.

Because the drums feature doesn't depend on recordable tracks for the final mix and includes stereo EQ and sends for the Master (aux) effects, the D3200 is as valuable as a demo-composing tool as it is as a straight recorder.

PRODUCT SUMMARY

KORG D3200

portable digital studio
$1,799

PROS: Excellent sound. Convenient layout. Versatile controls. Ample drum and effects library. Index options for CD burning.

CONS: Low-contrast LCD. Minor delays on some Knob Matrix functions.

FEATURES	■■■■■
EASE OF USE	■■■■■
AUDIO QUALITY	■■■■■
VALUE	■■■■□

MANUFACTURER
Korg
www.korg.com

Mix Master

The D3200 has an extensive set of editing and mixing features. You get standard recording features like auto and manual punch-in and -out; editing functions such as copy, erase, reverse, normalize, and fade; and features like time expansion and compression, "learning" noise reduction, 16 levels of undo and redo, and 4 locate points per song.

The D3200's mixer has 12 submixer channels in addition to the 32 recorder channels. Of those, 24 are equipped with a 4-band parametric EQ with selectable high or low shelving. Channels 25 to 32 provide a 2-band shelving EQ, as does each of the 12 submixer channels. The master track has a 4-band fully parametric EQ. Recording-input channels and track-playback channels use separate EQs.

The unit has a total of 12 buses, including 2 effects sends. Fader, EQ, pan, and effect settings can be stored in a Scene, and as many as 100 Scenes can be saved and automatically recalled for each song. You can automate fader, pan, and other mixer events internally, and you can back up and restore mixer data, switch scenes, and control transport functions through MIDI.

A built-in CD-RW drive lets you back up and restore songs and user data, or import WAV files for integration into songs. The D3200 has an Album CD Project function, which is essentially CD-burning software, that specifies the song order and length of gaps between songs. The internal hard drive can be partitioned so you can work on multiple album-CD projects at once. The CD-RW can also be internally routed to the mixer inputs and used as a CD player.

A portion of the D3200's internal hard disk is automatically allocated as a PC (USB) drive. That lets you import or export WAV data between the D3200 and a computer DAW program using the unit's rear-panel USB port (see **Fig. 2**) and can also be used to store system data. High-speed USB 2.0 is supported, so even large amounts of data can be transferred smoothly. You can also load backup data created by most other Korg D-series models.

On the Job

The D3200 I received for review was easy to work with and a welcome change from a CPU-and-software environment. Setup was effortless, which is an expected advantage of portable digital studios. I connected my 2.1 powered monitor system to the D3200's monitor out jacks, and I was able to deconstruct the funky R&B track that Korg includes as an onboard demo.

The song loaded quickly; the transport controls were solid, and their effect was instantaneous. Mute and solo functions allowed me to strip away the layers of production quickly and check out the sound of the unit. The audio quality of the demo, which was recorded at 16-bit, 48 kHz, was excellent. Automated events sounded smooth. Guitars and keyboards were mixed well, and the D3200 effects used showed a mix of sonic character. They didn't sound as though they came from the same unit, which is good.

Eager to try out the D3200 as a composing tool, I put it into Session Drums mode and began searching for beats. The nice assortment of traditional drums made it easy to find a quick loop for a Latin-jazz instrumental I had in mind. The Knob Matrix helped me to audition a variety of feels and sounds quickly. Because I could also edit the sounds by clicking on the Kit and Kit Change buttons in the LCD, changing textures was a breeze.

I was not completely satisfied with the LCD screen; the grayscale interface looked somewhat washed out, no matter how the screen was tilted or the contrast pot was turned. But the display was always readable, and the knobs, wheel, and ClickPoint joystick let me advance through tasks so fast that I soon forgot about the display aesthetics.

Full Effects

With my drum track programmed, I put down a foundation track with guitar. Plugging the guitar into the unit's high-impedance input was easy, but applying the D3200's effects on input required a trip to the manual, which is excellent. Korg has taken pains to put together an instruction book that makes every function clear, with plenty of numbered illustrations and cross-references to other sections of the book. It was an essential aid in helping me find any function I needed—from the first instruments I plugged in to the final CD I burned.

After applying a compressor from the unit's Dynamics and Filter section to my guitar input as an insert effect, I wanted to see what else I could do to the sound. The Multi section gave me a nice choice of amp models (Tweed 1 × 12, US HiGain, and so on). I wound up soloing over my drum loop (see **Web Clip 2**) for 20 minutes before thinking about recording again.

When I resumed my project, I was able to smoothly proceed from track to track, using all the unit's features, including virtual tracks, effects, aux sends, and buses, to create a 24-track demo and final CD mix that I could play for anybody. I was consistently impressed with the D3200's wealth of solutions for any session challenge I faced. Although a few minor inconveniences popped up, such as delays when new songs, functions, or sounds were loaded into memory, I never experienced a hang-up that slowed a session down.

Tenacious D

With recording quality becoming standardized in units like the D3200, and the difference in component quality less of an issue, factors such as ease of use, functionality, feature set, and price are what set one unit apart from the next. The D3200 excels in all these categories. It's an excellent composing and recording demo box on the road or at home, and it can serve as the backbone for a small studio creating demos for songwriters. With its disc-burning features and its ability to import audio from CDs or USB, it has the flexibility to prepare a mix for the mastering house or burn multiple copies for the clubs.

Anyone who wants a fully functional recording studio and doesn't want to invest several thousand dollars into a computer, a high-end audio application, CD-burning software, and outboard gear should put the D3200 high on his or her shopping list. EM WEB CLIPS

Rusty Cutchin is an associate editor of EM. He can be reached at rcutchin@comcast.net.

FIG. 1: Along with all the other parameters, MX4's Main page supports a beautiful LFO window that resembles a camera lens. The lens graphically illustrates changes you make to the LFO parameters using the surrounding slider knobs.

MOTU MX4 2.0 (Mac)

A plug-in that combines analog modeling with wavetable synthesis.

By Marty Cutler

Virtual analog synthesis can take either of two main paths: one emulates classic instruments, and the other blends physically modeled analog-synthesis components, digital waveform generation, and the signal flow and enhanced control afforded by modern computers. MOTU MX4 is a relatively recent addition that forges classic and unique new sounds from a variety of modulation options and a cross-fertilization of vintage digital and analog synthesizer technology.

Installing MX4 couldn't be easier; the provided iLok USB key authorizes it, and you're ready to go in just minutes. MX4 works only under Mac OS X with MAS, RTAS, and Audio Units hosts; it's unfortunate that there's no version for Windows. I ran MX4 on my dual 1.42 GHz Power Mac with OS X 10.3.9 and 2 GB of RAM. Hosts included Ableton Live 5.0, Apple GarageBand 2.0.2, Granted Software Rax 1.2.3, and MOTU Digital Performer 4.6.

The Cook's Tour

The basic complement of synthesis tools includes three oscillators, six LFOs, and four envelope generators (see Fig. 1). At first glance, the synth presents a relatively conventional subtractive synthesizer signal flow from oscillator to mixer, filter, and amp, but there's plenty more. One of MX4's great beauties is the extraordinary depth of programming capability that is neatly tucked just out of sight (see Fig. 2). For example, rather than clutter the screen with nests of virtual patch cords, MX4 takes full advantage of contextual menus invoked with a mouse-click and with the Control and Option keys, which can, for example, link an LFO to an oscillator or define a modulation range. Likewise, other modifier keys combined with mouse-clicks can pull up modulation-routing menus, reset modulation starting points, or create fractional parameter values.

At the top of each MX4 page is a small horizontal strip that holds basic page- and file-navigation options. Starting from the left are menus for bank and patch selection; you can scroll incrementally with plus and minus buttons or pull down the menus for a better overview of choices. Next, the Save button lets you immediately write your patch tweaks to the current location. Extra kudos for the A/B button—I'm used to an edit/compare feature on my hardware synths, and very few virtual instruments have that capability. Revert removes changes from RAM and returns the synth to its original state.

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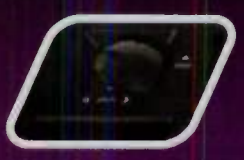

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FIG. 3: The Mods page hosts a variety of rhythm-oriented modulation devices and shapers. The lens in the center displays scaling and inversion shapers.

The next group of buttons in the strip accesses MX4's four pages: Main, Mods, File, and Random. Most of the programming happens on the Main page; the Mods page (added as of version 2) focuses on various rhythm- and tempo-based programming refinements and on additional modulation-shaping tools.

Each oscillator draws from an extremely generous selection of waveforms. Choices range from traditional sine, sawtooth, and rectangular waves to white and pink noise and a batch of unique and interesting wavetables. Version 2 adds band-limited, antialiased versions of all the prior version's waveforms. You select them incrementally with plus and minus buttons or click on the waveform name and use the pull-down menu.

You can wring lots more tonal variety from the synth's FM capabilities. Oscillator 3 modulates 1 and 2, and you control the amount of modulation to each carrier. In conjunction with the rich assortment of wavetables, the timbral variations are practically limitless.

Modulation in All Things

A slider that controls waveform symmetry invokes typical pulse-width modulation effects when you apply it to a rectangle wave. Modulating the symmetry of a sine wave can bring it close to a sawtooth wave's harmonic content. If you select a wavetable rather than a traditional waveform, though, you can use the oscillator's Index slider to adjust its harmonic content based on the slider's position in the wavetable's cycle, or you can sweep it for continuously evolving sounds. Modulating wavetable symmetry is rather unpredictable because the harmonic content depends on the portion of the wavetable's index you're modulating (see **Web Clip 1**).

Using a slider to modulate symmetry is only the tip of the sound-design iceberg. You can also modulate symmetry and the wavetable index with any MIDI Control Change (CC) message to produce powerful sounds with a pulsating inner life (see **Web Clip 2**). What's more, thanks to the Shaper section, you can have modulators modulating other modulators in as complex a daisy chain as your CPU can handle.

The LFO window is a gorgeous piece of design, resembling a frontal view of a camera lens. Sliders surrounding the lens set LFO Rate, fade-in time, delay, starting phase, and waveform symmetry. Changes to those parameters are reflected graphically inside the lens. Tabs at the lens's upper-right quadrant let you select from each of the six LFOs.

MX4's two multimode filters sound glossy and smooth, but they give you plenty of latitude for grit and distortion. Each filter offers highpass and lowpass varieties as well as bandpass and band-reject types, with 6, 12, 18, and 24 dB slopes. You choose a filter configuration in the Topology window, which has a pictographic pop-up menu that illustrates the signal flow of the two filters and the distortion section. If distortion isn't enough, applying FM to the filters adds a bright edge and an animated buzz to the timbre.

Four razor-sharp, responsive envelope generators can modulate any continuous parameter. Although Envelope 1 is hardwired to amplitude, all envelopes can modulate multiple parameters. MX4 adds a delay stage before the attack, and a hold rate between sustain and release. The additional stages furnish more graceful tailoring of attack and release stages than a typical ADSR.

MX4 is a veritable modulation theme park. The key to its extensive modulation capabilities is the Source and Shapers panel at the lower right of the Main page. The combinations you specify there can provide endless permutations within a complex modulation signal flow. A radio button to the left of the Source column lets you select a modulation source. You can shape the mod source by making a selection in the Shaper column. If you Option-click in the center notch of a modulated parameter's slider, the slider will split in two, letting you define high and low modulation ranges. The Information window displays the selected parameter, the base value (the value that the modulation starts from), and the modulation range. The window also displays the number of voices in use.

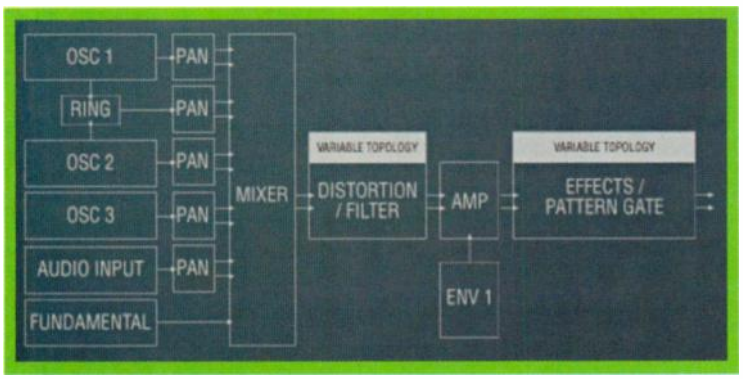

FIG. 2: Although MX4's signal flow is familiar and relatively simple, quite a few additional controls are hidden with key commands, minimizing onscreen clutter.

REVIE

Oscillatin' Rhythm

The majority of the Mods page furnishes rhythm-based modulation components such as a programmable gate, two types of onboard sequencers, and an arpeggiator. As with the filter section, the Mods page offers variable topology for the Pattern Gate and two effects slots; you can feed one element into another in any order you choose by dragging the three modules (see Fig. 3). Mods are not constrained to simple triggering of notes or amplitude; shapers can affect timbral changes or modify other modulators.

Invert and Transform shapers provide a terrific graphic display for scaling modulation; inverting filter envelopes is great for those chirping, resonant synth basses. The Quantizer is a bit more esoteric, but you can shape modulation by pitch; clicking on notes in the keyboard graphics lets you lock modulation values to selected pitches. (Here's your opportunity to shape an LFO in G Mixolydian.) MX4 can process external audio in several ways: you can blend it with the instrument's oscillators, apply filters and effects, or process it with a ring modulator; and—in the Mods section—you can input sound so that its amplitude serves as a control source or shaper with the envelope follower.

Pattern Gate slices MX4's audio output into rhythmic pulses. You can alter rhythms with its step-sequencer-style interface and change the pulse envelopes. A similarly arrayed Trigger Sequencer can force any of MX4's envelope generators into discrete rhythmic steps. Each patch can hold its own programmable arpeggiator and pattern sequencer. All rhythmic modulators sync to the host tempo and let you set a rhythmic value and swing amount.

Animation Festival

Although MX4 can emulate a variety of vintage synthesizers, comparisons to any particular classic instrument are irrelevant.

PRODUCT SUMMARY

MOTU MX4 2.0

soft synth plug-in
$295

PROS: Capable of lots of timbral motion. Elegant and extremely flexible modulation scheme. Excellent time- and tempo-based controls. Can use audio input as oscillator and control source. Terrific documentation and tutorials.

CONS: No Windows version. No standalone version.

FEATURES	
EASE OF USE	
AUDIO QUALITY	
VALUE	

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The plug-in can dish out sounds ranging from raucous, in-your-face leads and chunky brass ensembles to thick or gauzy pads—all animated with tons of timbral and harmonic motion.

Among my favorite factory patches are Held=Pad, Fast=Pluck, which sounds like the result of some unholy congress between a didgeridoo, a dentist's drill, a vacuum cleaner, and a vocoder-processed drum machine. A patch called MS Windows 2058 for Washing Machines superimposes astonishingly eerie, bagpipes-from-hell melodies over a pulsing drone (see **Web Clip 3**). Other standouts include the distant, distorted cloudiness of We're Through and the drunken, single-finger melodies of Happy Hour for Chips.

MX4 is somewhat processor intensive, but it provides facilities to reduce the toll on your CPU, such as toggling the effects on and off and limiting the number of voices per patch. If you don't need a heavy-duty multiple-filter configuration, simpler filter arrays and coarser slopes can help ease the drain.

Documentation is more than thorough. My copy arrived with a lucid manual and a separate booklet covering the multitude of features in version 2. The manuals are rife with terrific programming tips. A section in the update pamphlet gives an overview of five important concepts for programming MX4. Many of the tutorials reference a bank of example patches.

Love Notes

I am thoroughly enamored of MX4. You can delve deeply into its first-rate modulation capabilities, harness its intelligently arranged Randomization page, or simply tweak a few parameters and create something new and interesting. A few items for future development are on my wish list, however. As a MIDI guitarist, I wish that it had some multitimbral capabilities, or at least the ability to allocate a limited number of voices in a single instantiation over multiple MIDI channels. If your CPU can handle it, though, you can run multiple instantiations assigned to different MIDI channels. Hopefully, MX4 will become a standalone instrument, too.

This review barely scratches the surface of what is available in MX4. MOTU's Web site holds a generous number of MX4 audio demos. Better yet, you can download a 30-day, fully functional copy of the synth. Don't forget to come up for air and food.

EM WEB CLIPS

Former EM assistant editor Marty Cutler fights bad software manuals in New Jersey so you don't have to fight them at home.

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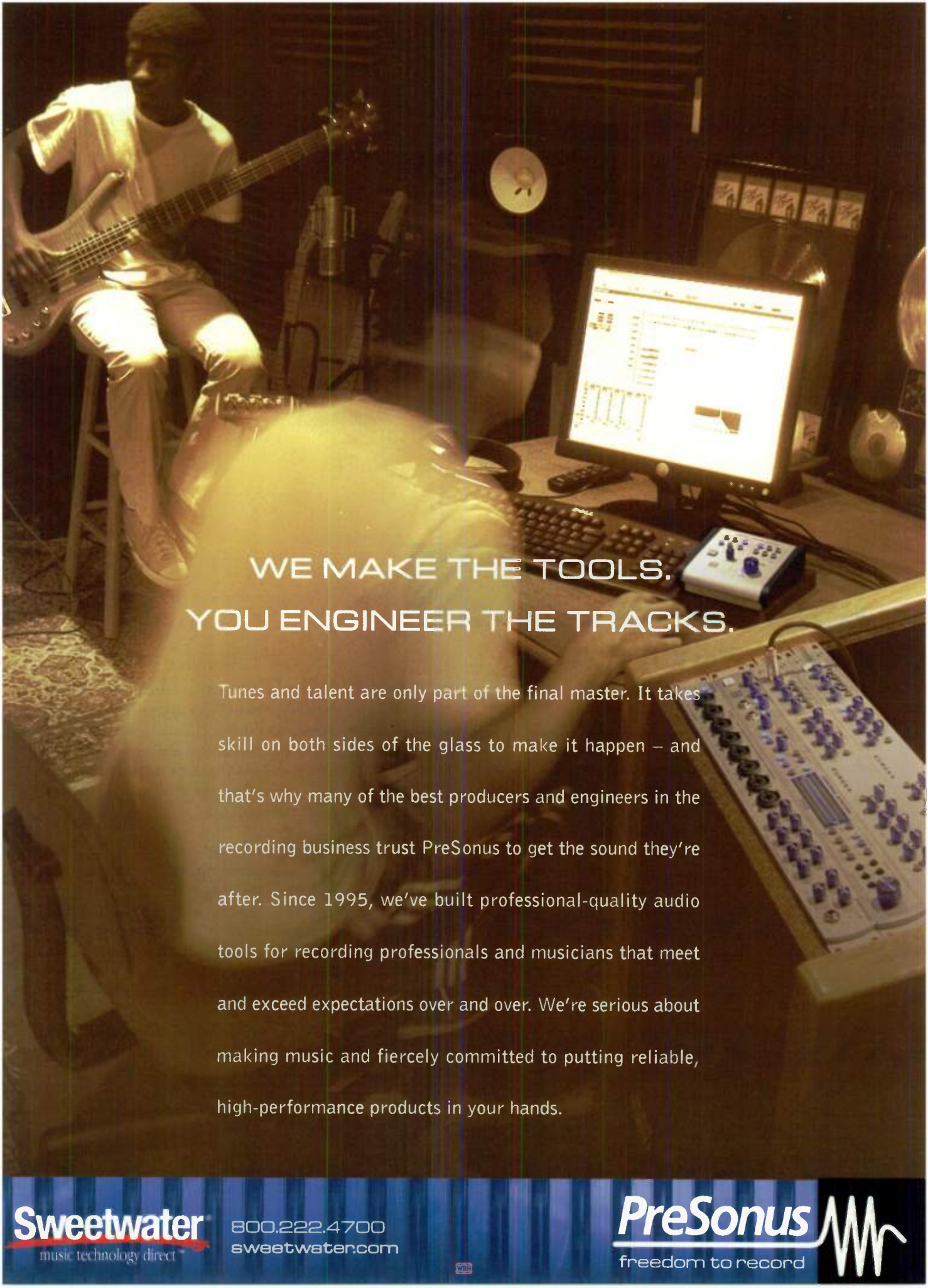
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FIG. 1: The DK25 DrumKit System comes in two flavors: the DK25/R (left), which is intended for studio applications, and the DK25/L (right), which features cardioid overheads and is designed primarily for live drum miking.

EARTHWORKS DK25

A simple and effective drum-miking system.

By Rob Shrock

Capturing a stellar recording of a drum kit is rarely simple. Typically, it involves multiple mics, multiple tracks, and lots of processing afterward. The Earthworks DK25 DrumKit System (see **Fig. 1**) offers an alternative method for kit recording that is simple and effective: two microphones serve as overheads, and a third microphone is set up to record the kick drum.

The manufacturer's claim is that this system provides a more solid, focused drum sound than the array of 7 to 12 microphones commonly used in professional studio and stage applications. The idea is that by reducing the phase cancellation issues that arise as the number of microphones increases, the kit will have more punch and will sound like one cohesive instrument rather than a collection of individual drums.

That is not a new approach to recording the drum kit. A large part of John Bonham's drums with Led Zeppelin were recorded with two overhead mics and a mic on the kick drum. Those drum sounds have influenced a generation of rock drummers, engineers, and producers, and they still sound great today. It's interesting to note that many great recordings of the past treated the drums as a single instrument, rather than segregating the kick, the snare, and the cymbals (as is now commonplace).

New and Noteworthy

Earthworks has not simply repackaged old techniques into a new product. The DK25 system uses a proprietary microphone design. The mics feature a small diaphragm for exceptionally fast transient response but maintain the extended and consistent frequency response for which Earthworks mics are known. A new twist is a proprietary (and patent-pending) device called the KickPad (see **Fig. 2**), which is inserted between the kick drum mic and the mic pre. The KickPad customizes the kick drum mic's signal to a near-perfect sound. The result is a dynamic, detailed, and open drum sound.

Earthworks has not simply repackaged old techniques.

The DK25 is available in two versions (both of which I received for this review): the DK25/R (Recording) and the DK25/L (Live). The DK25/R comes with two TC25

omni condensers for overheads and a single SR25 cardioid condenser for the kick. The DK25/L, which is the same price as the DK25/R, features three SR25 cardioids to minimize unwanted stage sounds. Both systems use the KickPad on an SR25 cardioid for the kick drum.

FIG. 2: Both incarnations of the DK25 come with the KickPad, a proprietary device that plugs in between the mic preamp and the microphone, and alters the EQ contours of the mic signal to optimize it for kick drums.

Inside the Box

The DK25/L includes a metal carrying case with interior-fitted foam for the three microphones and the KickPad. The DK25/R comes in a nice wood box that also has interior-fitted foam. Both sets include a small windscreen for the kick drum mic to help minimize airburst problems. Each microphone comes with a clip that holds it securely in place. All the mics are condensers, so they require phantom power.

The specifications indicate that the frequency response for the omni and the cardioid mics goes as high as 25 kHz, with the cardioid going down to 50 Hz and the omni going down to 9 Hz. The mics can handle a maximum of 145 dB SPL, so there is no harm in using them in loud settings. The diagrams in the manual and in the promotional material claim that the polar patterns are uniform. A layman's test backed up that claim: the omni sounded virtually the same in all directions.

The KickPad appears to do its sonic treatment with some kind of tailored EQ. Earthworks would not reveal exactly what was going on inside it other than to confirm that there was some frequency manipulation. According to Earthworks, the KickPad can be used with any cardioid mic, not just the SR25. Even if you don't want to purchase the entire DK25 kit, you can buy just the KickPad for $124 and use it with a cardioid that you already own.

Whatever kind of voodoo is going on inside the KickPad, it helps make for a deep, great-sounding kick drum when paired with the SR25 (see **Web Clip 1**). If it doesn't suit your taste, you can easily shape the sound further with EQ, compression, or both.

In the Studio

I was able to use the DK25/R set for a rhythm date I produced at Radio Recorders in Los Angeles. Session drummer Kevin Stevens was on the date, so I knew I had the makings for a good drum sound.

When I arrived and informed the engineer we would

be using Earthworks microphones on the drums, he immediately smiled. Only a few weeks earlier he had cut a new album for Lucinda Williams in the same room using the same TC25 microphones for drum overheads. He was pleased, and he told me how easy it was to get a great drum sound with them.

Because we were cutting five tunes that day, I set up additional microphones on the drums so that I'd have the sonic versatility to vary the sounds from track to track. In addition to employing the DK25/R system, we added an AKG D 112 on the kick; a mono ribbon overhead; traditional mics on the snare, toms, and hi-hat; and two AKG C 12s set up as room mics. In other words, I would have the tracks from the Earthworks 3-mic system and a traditional 11-piece miking array to choose from in the mix (see **Web Clip 2**).

No More Overheads Needed

I had originally intended to add a second pair of overheads to the session as a safety measure, but as soon as I heard the TC25s by themselves in the control room, I dropped that idea. If I had to, I could have lived with the sound from just those two mics. I then asked the engineer to bring up the SR25 on the kick drum (with the KickPad), and there it was: a complete drum sound.

The sound of the kit was detailed and focused. I could hear every nuance. The kick was deep and full, the snare and toms maintained a great balance of top end and bottom end, and the cymbals were open and airy. A touch of high-end EQ was added to give the sound some shimmer.

We then started adding some of the other drum mics, and the sound changed. In some ways it got bigger as the subtle delay and phasing effects of more microphones were introduced, but in other ways it got cloudier and more tonally segregated between individual drums. It's not a bad sound, just a common one. The "great drummer, great kit, great room" vibe was starting to disappear.

Without question, the DK25/R system provides excellent drum sound that works well if you're in a great-sounding room. If you're in a studio that is small or doesn't sound so great, however, I would opt to try the DK25/L cardioid microphones as overheads to cut down on the room sound. In fact, if you are in a small or mediocre room, I would argue that the 3-mic system sounds better than a multi-mic array, because the latter will introduce more of the bad room and increase the phasing issues.

As I write this, I am still working on the five songs we recorded that day, and I have not decided which combination of microphones best serves each song. I have, however, decided to use only the DK25/R system without any of the other mics for at least one tune—that drum sound fits the song's vibe perfectly.

One technique that I am also considering is using the additional snare and tom microphones to feed a reverb unit without bringing up their nonprocessed signals in the mix. That way I can use discrete processing, one of the advantages of individual drum miking, while still maintaining the focused, punchy, and direct drum sound provided by the DK25 mics.

Of course, the beauty of the DK25 DrumKit System is that it gives you three killer microphones you can also use for other applications. The mics came in handy in

PRODUCT SUMMARY

EARTHWORKS DK25

drum-mic set
$2,100

PROS: Easy to get a great drum sound. Excellent transient and frequency response. Uniform polar pattern. Microphones are great for recording other sources. KickPad provides tailored EQ specific for kick drum. KickPad works with other cardioid microphones.

CONS: Relatively pricey.

FEATURES	■■■■■
EASE OF USE	■■■■■
AUDIO QUALITY	■■■■■
VALUE	■■■□□

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DK25 SPECIFICATIONS

TC25	
Frequency Response	9 Hz–25 kHz (+1/–3 dB)
Polar Pattern	omnidirectional
Sensitivity	8 mV/Pa (–42 dBV/Pa)
Power Requirements	48V phantom, 10 mA
Peak Input Level	145 dB SPL
Output	XLR (pin 2 hot)
Minimum Output Load	600Ω between pins 2 and 3
Noise	27 dB (A-weighted)
Dimensions	6.5" (L) × 0.86" (D)
Weight	0.35 lbs.
SR25	
Frequency Response	50 Hz–25 kHz (+/–2 dB)
Polar Pattern	cardioid
Sensitivity	10 mV/Pa (–40 dBV/Pa)
Power Requirements	48V phantom, 10 mA
Peak Input Level	145 dB SPL
Output	XLR (pin 2 hot)
Minimum Output Load	600Ω between pins 2 and 3
Noise	22 dB (A-weighted)
Dimensions	6.5" (L) × 0.86" (D)
Weight	0.35 lbs.

the studio that day, because I didn't like the acoustic piano sound we were getting, even though we were using a very popular set of studio microphones.

Because I had the SR25s with me, we set them up in addition to the piano mics already in place and ultimately recorded two stereo pairs. As with the drum set, I am still undecided as to which combination of mics I'll end up using for each song—either the two sets blended together or the Earthworks mics by themselves. If I do end up using a blend, it will probably be at least 60/40 in favor of the SR25s, if not more.

After that session, I used the SR25s for recording acoustic guitar and achieved an excellent sound. I even tried one out on vocals and it sounded great. Remember, you can use these microphones for just about any source—they aren't exclusively drum microphones. Just don't forget that the TC25s are omni mics, which will pick up sound in all directions and may not be appropriate for some sources and environments.

The Works

I loved the approach to recording drums and the quality of sound provided by the DK25 DrumKit System. It is easy to get a killer sound, especially when using the KickPad. (Earthworks will send you a free demo CD with information and sonic examples of the DrumKit System. See the company's Web site for details.) Even if you can't afford the complete system, there is no reason not to get a KickPad for use with your own mic when recording a kick drum. It will save you time.

The transient and frequency responses of Earthworks microphones are legendary, and they have become staples in test-measurement and creative-recording circles. Drums and percussion arguably benefit from those traits most, and it is satisfying to shift the paradigm back to listening to a drum kit as a single instrument. The DK25 DrumKit System forces you to think that way again, and I am finding that good drummers love it. If you provide the great drummer, drum kit, and room, Earthworks will take it home from there.

EM WEB CLIPS

Rob Shrock is an EM contributing editor who played keyboards on the new Burt Bacharach release, At This Time, *and has worked with a who's who of artists.*

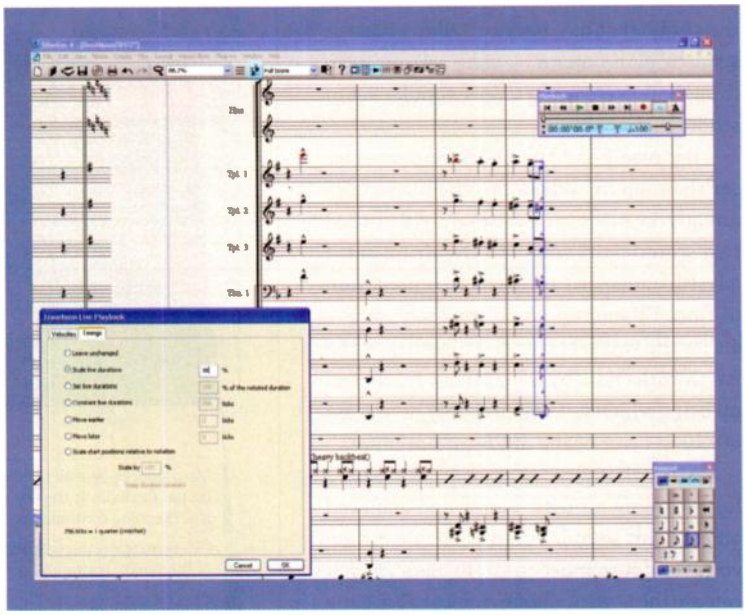

FIG. 1: Articulations and other markings in Sibelius carry MIDI data that gives them proper playback characteristics. Redefining text markings is quite easy, but redefining articulations is more involved.

SIBELIUS Sibelius 4 (Mac/Win)

User-friendly notation software with new and powerful features.

By Brian Smithers

Sibelius 4 is the latest incarnation of Sibelius Software's well-respected notation program. This upgrade retains the comprehensive feature set and intuitive interface that have placed previous versions among the preeminent score writers available, and also adds at least two major new features and a number of smaller enhancements. Because EM has previously reviewed versions 2.1 (in September 2003) and 3 (in June 2004), I'll recap Sibelius's main characteristics and then focus primarily on the new features.

Sibelius is one of those programs that prove that professional features and a steep learning curve do not have to go hand in hand. When I first auditioned Sibelius in its original Mac/Win version (it had existed on the Acorn platform for years prior), I was literally up and running in minutes. At the time, its intelligent layout features managed to provide readable and good-looking parts and scores with minimal effort, but in many cases they did not offer enough flexibility for the user to overrule the program's decisions. With each subsequent revision, however, the program has become more amenable to user preferences.

Its Flexi-time Record mode tracks your tempo as you play parts, taking real-time note entry to a new level of efficiency. Play easy parts quickly and difficult parts slowly, and Sibelius understands. Playback is equally intelligent, recognizing articulations and other markings. You can redefine the playback parameters of score markings and define new markings, but redefining articulations is a cumbersome process, as I discovered when trying to get a marcato to play back as a short jazz accent instead of a long orchestral accent (see **Fig. 1**).

Sibelius includes a modest but good-sounding sample library that is well suited to orchestral scores. Using a custom version of Native Instruments Kontakt for playback, the bundled Silver sound set offers 20 different instruments but lets you play only 8 at a time. If you upgrade to Kontakt Player Gold, then you get more sounds and can use as many as 32 simultaneously. You can also assign each staff to any hardware or stand-

prophet-5

The Analog Prophet 5 - The Best of Both worlds - The Digital Prophet VS

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alone software synth in your arsenal, but Sibelius does not support plug-in instruments without a third-party plug-in host.

Picture This

New to Sibelius 4 is the Video window. You can now import a movie file directly into the program and score directly to picture. This is a great step forward for film composers. Instead of composing in your favorite sequencer (which has probably supported video for some time) and then exporting a MIDI file and importing it into Sibelius to create a score and parts, you can choose to work directly in the score from the start. It's the next-best thing to having a video window in your score paper.

Because Sibelius leverages your system's video player for the Video window, anything you can play in Windows Media Player on the PC or QuickTime on the Mac or PC can be played in Sibelius. The window is resizable and can even display full screen on a second monitor, subject to the quirks of your media player. Playback through a FireWire device is not currently supported.

Most common frame rates are supported, with the noteworthy exception of 29.97 drop frame. This means that, when working with standard NTSC color video, you'll have to use 29.97 nondrop, which will make the tempos in your score slightly inaccurate. Then you'll have to recalculate them before setting up the click track when you reach the scoring stage. Fortunately, a fix should be available by the time you read this. The Timecode and Duration dialog box lets you specify offsets, allowing you to start playback at a specific point in the score, or from a point other than the beginning of the movie.

You can create Hit Points, which are markers in the score or parts for visual reference of important cues (see **Fig. 2**). Change a tempo, and the Hit Points move accordingly. Unfortunately, you can't yet use Hit Points as locate points, nor can you jump directly to a specific frame. In fact, frame-accurate positioning is not currently possible. You can scrub the video only to a resolution of 0.2 second, or approximately six frames. According to Sibelius Software, though, the 4.1 update will feature frame-by-frame advance.

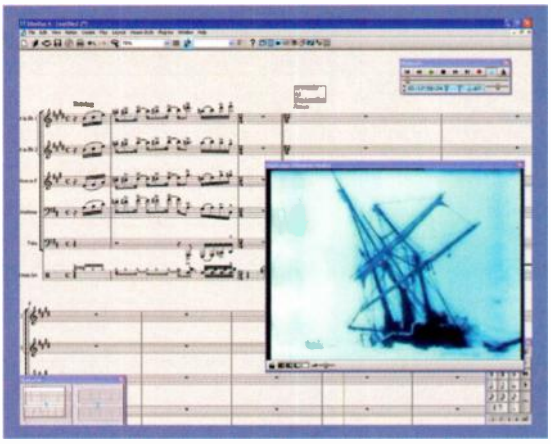


FIG. 2: Sibelius 4's support for video includes a resizable playback window and the ability to place markers, called Hit Points, in the score and parts to designate important cues.

Although Sibelius's newfound support for video is a great enhancement, I was disappointed to find that I couldn't attach a specific frame to a specific bar and beat and have the previous tempo scaled accordingly. Instead, to make a Hit Point that heralds the first frame of the kiss at the first note of the violin cue, you must break out your calculator, old-school-style. The point of using a computer for a creative task is letting *it* do this sort of left-brain number crunching so that you can focus on the creative side. Sibelius hopes to add this functionality in a future version.

Sum of the Parts

To paraphrase Zeno's paradox (the one regarding the racecourse), the closer you get to the end of a project, the more you feel as though you'll never finish. This is due to the myriad details that must be addressed before you're actually done: expression markings, articulations, dynamics, large- and small-scale layout, proofreading, and of course the biggest obstacle—extracting parts. This last step is a ritual so fraught with exasperating pitfalls that some of my colleagues (who use another major score writer) actually bypass it in favor of copying and pasting from the score.

Sibelius 4's Dynamic Parts make part extraction a thing of the past. All parts coexist with the score from the beginning; if you add a staff to the score, a corresponding part is created. Every edit you make in the score is reflected immediately in the parts. Alternatively, you can enter notes in the parts, and the score will be updated.

Notes are of course inextricably linked between parts and score, but the positions of dynamics and

PRODUCT SUMMARY

SIBELIUS SOFTWARE Sibelius 4

notation software
$599

PROS: Excellent output quality. Exceptional ease of use combined with exhaustive professional feature set. Video support. Dynamically linked score and parts. Intelligent playback of markings. Flexi-time entry follows tempo of user input.

CONS: Kontakt Player supports only eight simultaneous sounds. It's difficult to customize playback. Video support is not frame accurate in version 4.0

FEATURES
EASE OF USE
DOCUMENTATION
VALUE

MANUFACTURER
Sibelius Software
www.sibelius.com

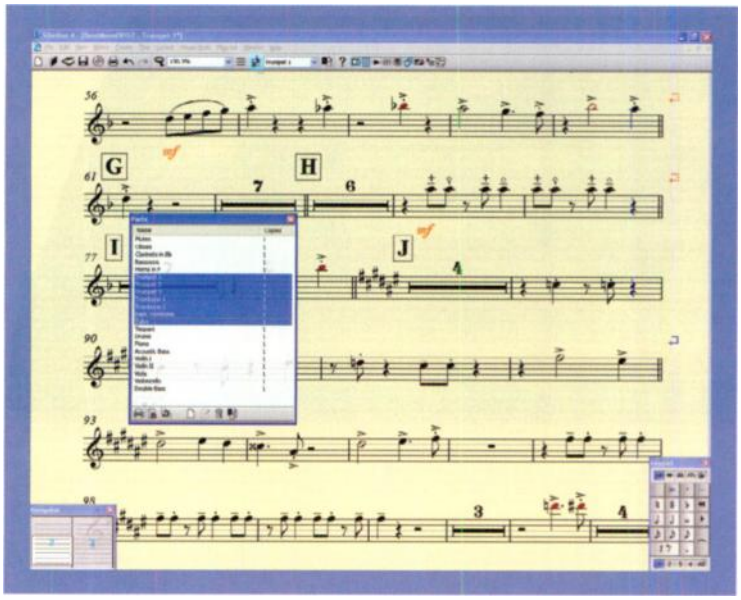


FIG. 3: Dynamic parts coexist with the score from the start without having to be extracted. Note entry can take place in either score or parts, and other elements can be repositioned in the parts without affecting the score.

other markings can be fine-tuned within the parts without affecting the score. Once you've adjusted something in a part, it turns orange in the part to indicate its tweaked status (see Fig. 3).

The Parts window offers some welcome enhancements for dealing with part layout, most notably a function called Copy Part Layout. As the name suggests, Copy Part Layout lets you adjust the layout of one trumpet part, for example, and then apply those adjustments to the remaining trumpet parts. No longer must you endure the tedium of making the same tweaks repeatedly. Similarly, Multiple Part Layout allows you to individualize settings from the Document Setup, Layout, and House Style dialog boxes for subsets of parts.

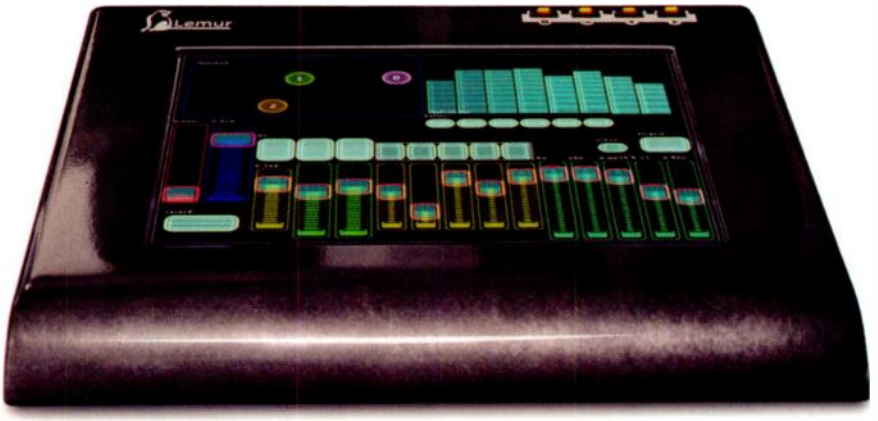
Dynamic parts are contained within the same file as the score, making file management simple. In some circumstances, however, it may be necessary or desirable to have individual files for parts, so the traditional Extract Parts command is still available, although it has been relocated to the Parts window. I was at first concerned that dynamic parts would bog down my system, but even with a large orchestral score, I saw no performance drain.

Al Coda

Version 4 is rounded out by a number of additional enhancements, including a Worksheet Creator wizard, almost 500 new pieces of music, and easy copy and paste to Microsoft Word and Excel. Among the new plug-ins is a powerful but oddly mouse- and dialog-boxs-intensive process for copying and pasting articulations and slurs. A new font called Helsinki does a good job of giving scores an old-school classical look.

If you're already a Sibelius user, version 4 gives you good reasons to upgrade, regardless of whether you use it in a production or education environment. If you're not yet a Sibelius user, check out the demo and see if the interface appeals to you as much as it does to me. You will find no shortage of professional features, and you will find nothing lacking in terms of quality output and powerful interface.

Brian Smithers is a musician, composer, engineer, and educator in Orlando, Florida.



DMITRY PANICH

JazzMutant's Lemur is a touch-screen controller that allows you to create multiple highly customizable interfaces. The many available interface elements can model the physical properties of natural forces such as friction and gravity.

JAZZMUTANT Lemur

The future of sound control.

By Jonathan E. Segel

Using a mouse to manipulate the many buttons and knobs in today's DAWs can get tiresome and painful. Though most external control surfaces give you a lot of controls to work with, and perhaps even some programmability, JazzMutant's new Lemur multitouch control surface takes the concept of remote control to a new level.

The Lemur is a touch-screen controller with a programmable interface that interacts with the destination software via Open Sound Control commands (OSC; see www.opensoundcontrol.org/about). The included JazzEditor software (Mac and Windows, with Linux "on the way") allows the user to create a colorful, configurable interface using various graphical objects. The software is also used to set up the parameters for communication with the Lemur itself. The result is an extremely versatile piece of hardware that is way beyond your average knobs-and-dials MIDI controller, though at an equally beyond-average price: at $2,495, the unit probably costs the same as the entire computer system you might use it with.

At Your Fingertips

The Lemur is a flat device that has a 12-inch multitouch-sensitive LCD screen and an overall footprint the size of a large laptop. It contains its own processor for graphics and mathematics, so it uses no host CPU power. It has four lighted buttons at the top right corner: the Settings

LEMUR SPECIFICATIONS

Display Size	12"
Display Resolution	800 × 600 pixels
Display Type	LCD
Computer Interface	Ethernet (100Base-T)
Power Requirements	12 VAC external, 100–240 VAC
Dimensions	14.5" (W) × 1.2" (H) × 11.6" (D)
Weight	6.5 lbs.

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button accesses communication settings, the Interface List button displays thumbnails of all available interfaces, and the forward and back buttons scroll through the interfaces. The front side is curved for comfortable manipulation, and around back are the power switch, DC input, and an Ethernet port, through which the Lemur communicates with the outside world.

OSC has several advantages over MIDI, most notably 32-bit precision and extremely low-to-nonexistent latency. (Some speculate that OSC will one day replace MIDI.) Only a few commercial programs currently employ OSC directly, the most ubiquitous being Reaktor (and a few other Native Instruments applications) and Cycling '74's Max/MSP and Jitter. Several popular noncommercial programs also use OSC, including SuperCollider, AudioMulch, and Miller Puckette's Pd.

The Lemur can also internally map its own OSC messages to MIDI, so that nearly any program can use it as a controller. For example, you could use the Lemur's Pad object to send Note Ons to trigger cells in a non-OSC-compatible application like Ableton Live. When the output is mapped to MIDI, the Lemur shows up as a virtual MIDI input port in the target software. Of course, the output resolution is then limited to the depth of the MIDI Control Change message you're using.

Getting the Lemur to communicate with your software is trickier than making a USB or FireWire connection, but once the idea of using Ethernet cables and IP addresses sets in, you're halfway there. Fortunately, the documentation contains step-by-step instructions for many routing configurations and is thorough and easy to read.

Making Contact

When you plug the unit into your computer's Ethernet port, the Lemur will assign itself a network address. The Settings button at the top of the front panel accesses the Lemur's Setup window in case you want to assign a specific address to it. The corresponding button at the top right of the JazzEditor software shows you the Lemur's address as well as your host computer's IP address, which you then have to make sure is the target address. Next, you specify the TCP/UDP port, usually port 8000 in, and you're on your way.

Initially I had frustrating problems with Lemur-to-computer communication. The main problem was the Lemur's inability to gain a network address dynamically.

MAKING CONNECTIONS

Parsing the OSC messages in Max/MSP, SuperCollider, or Reaktor is, for the most part, fairly easy. In Max you use the provided udpreceive object, then route the messages by name and send their respective values to the corresponding object in the patch. Max and SuperCollider will look for OSC activity by port, but in Reaktor you must set the remote controller's information (name, IP address, and port number) in the global OSC settings panel. Sometimes, however, I noticed that Reaktor couldn't detect the Lemur, especially when starting a new patch. Relaunching Reaktor and the JazzEditor software usually solved that problem.

Simpler Reaktor modules have OSC input listed in their Properties menu already (knobs and faders, for example). For others, you have to get the message in and route it somehow. In attempting this I encountered a few snags. For example, whenever an object is created in the Lemur interface, its name shows up in the JazzEditor's Project browser window with a little checkbox that is not enabled. In order for the object's data to be sent, you must remember to enable the object in the Project window. Also, every time I created a Lemur object, I needed to bring Reaktor into focus and move the object on the Lemur to get Lemur to show up in the list of available inputs for OSC control in the desired Reaktor module's properties window.

A bigger problem was getting Reaktor to parse OSC messages with multiple arguments. In Max/MSP or Jitter, this is no problem: a message comes in by name, followed by any number of arguments. For example, if you had a MultiBall object with five balls, the *x*-axis movement (horizontal) would send "/multiball.x," followed by five numbers corresponding to the balls' positions. It would also send a set of messages for the corresponding *y*-axis (vertical) and *z*-axis (contact-point size) movements.

In Reaktor, however, it is extremely easy to route an OSC message with one argument, but messages with multiple arguments proved extremely difficult. Even a set of two radio buttons weren't understood by a Reaktor panel button when set in radio mode. Trying to send Lemur MultiBall or MultiSlider info was impossible; an attempt to use Reaktor's OSCReceive module to parse multiple-argument messages crashed the application. A few emails with Native Instruments' tech support have confirmed that its OSCReceive module is faulty, at least in the Mac OS.

One way around this is to assign each individual multislider bar to a MIDI controller within the Lemur, which gets the control information to the application, though at a much-reduced resolution. MIDI mapping to multiple controllers is done automatically; when you set the controller number and assign its out value to be a multiple-argument object, the Lemur maps each argument to the next available controller number.

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Independent Producer/Engineer
Nashville, Tennessee

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But this was fixed within a week with updated firmware and helpful suggestions from the manufacturer. With the new firmware installed, I plugged the Lemur into my router; it acquired its own network address and then became accessible to any networked computer in my studio. Because the Lemur appears as a networked device, many Lemur controllers could exist on the same network, or one Lemur could conceivably control many computers.

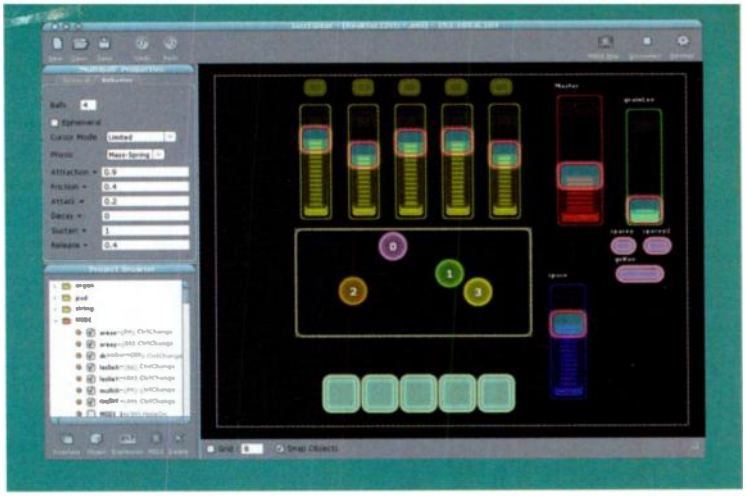

FIG. 1: Lemur's multitouch capabilities let you control many different parameters of your software at once. Its internal clock feature allows controller data to be transmitted in sync with real time or with the tempo of your music.

No Objections

The real work with the Lemur starts with making your own graphical interfaces. This process involves choosing from eight interface objects: Fader, Monitor, MultiBall, MultiSlider, Pads, RingArea, SignalScope, and Switches (other interface elements are in the works, according to the manufacturer). You can freely position objects anywhere on the screen and easily change their sizes, and you can have up to 1,024 objects and 1,024 mathematical expressions (for scaling or converting data, for example) in one project (see **Fig. 1**). You can make either horizontal or vertical faders, and you can configure the Monitor object to monitor other objects' values (for example, the values of MIDI notes being sent from an application). You can also monitor any variables you might be using. The Lemur comes with one variable already set: Time, which is based on its built-in clock. The Time variable allows the Lemur to display and transmit timecode, and you could map the values of some variable to Time so that the values would increment or decrement in sync with the clock.

MultiBall is an *x*-*y* area controller that can have up to ten controllable balls. The balls can remain permanently visible or show up only when you touch the pad, and the Lemur can discretely recognize up to all ten spots at once. When you place your fingers on the MultiBall object area, the ball nearest the spot you touch becomes the one that you control. Imagine controlling the frequencies and volumes of ten separate synthesizer voices with your ten fingers simultaneously!

A RingArea has one ball and an attraction point, and you can set a value for the attraction point that the ball will always move back to. For example, the attraction point could be set to play back a sample at normal speed, and you could then use the movement of the ball to "scratch" the sample like a turntable. You can arrange the Pads and Switches into multiple rows and columns, and each control can transmit its range of values under enveloped control.

Beyond the ability to make your own custom interfaces, what's really amazing is the type of control you have over the interface objects' visual and output behavior. Objects can be set to mimic various physical properties, such as the behavior of a mass spring. You can apply the properties of friction and attraction to an object's behavior, or use the amount of area that your finger covers (that is, contact-point size or *z*-axis control) to make touching the screen feel more pressure sensitive. Adding a small amount of physical behavior can make a world of difference, not only in tactile perception, but also in the way a parameter responds: smooth interpolation of a control makes volume or frequency feel much more organic.

The addition of mass-spring behavior adds previously unheard-of possibilities for, say, controlling filter cutoff frequency. Imagine that instead of simply moving a knob or fader to set filter resonance and cutoff frequency, you have a ball in a two-dimensional grid, where one axis is set to frequency and the other to resonance level. If you assign a higher attraction value and lower friction value to the ball's response settings, the Lemur will follow your finger on the screen like a living object, circling in on the values you are moving through and transmitting. (Because of its multitouch capabilities, you could control many such objects at the same time.) Translated into an audio response, this makes for extremely natural-sounding results.

PRODUCT SUMMARY

JAZZMUTANT Lemur

control surface
$2,495

PROS: Amazing response. Infinitely configurable. Beautiful interface. Instant visual feedback. It's much better than knobs or sliders.

CONS: Expensive. Software still in development. Not all bugs have been ironed out yet. Few commercial music programs support the Open Sound Control communications protocol.

FEATURES	■■■■■
EASE OF USE	■■■■■
DOCUMENTATION	■■■■■
VALUE	■■■□□

MANUFACTURER
JazzMutant/Cycling '74 (distributor)
www.cycling74.com/products/lemur.html

Because your interface designs are saved as XML files, you can transfer an interface file to a text editor and edit it. A single file can contain multiple interface arrangements, all of which are accessible from the JazzEditor software's Project browser or via the Interface List button at the top right of the Lemur itself. This allows you to easily switch among all the interfaces you have saved.

Double Duty

Keep in mind that for every patch in Max/MSP or Reaktor you want to control, you'll have to spend an additional amount of time creating a corresponding patch for the Lemur. Interfaces are highly individualized to the parameters of a specific patch, so some are going to be easy, and some not. And even after you've finished the design stage, you'll need to spend additional time ensuring that the target software can handle what you're sending it (see the sidebar "Making Connections" for details of communication problems related to specific applications).

The Lemur feels good. Perhaps not fuzzy like a living lemur, but the touch surface has a slight grid pattern, so running your fingers over it provides very, very slight resistance, making it just perceptible. This feeling, combined with the visual feedback from the LCD touch screen, makes it seem as if the objects are really responsive to pressure, despite a lack of actual pressure sensitivity. The visual response of the unit depends on the physical behavior you have set for an object; faders may move exactly with your fingertip or follow it only slightly. (Note that your host audio software will follow the object's motion, not the movement of your finger.)

Using the Lemur as a control surface in live performance feels much more like playing an actual instrument than simply turning knobs or moving faders on a control surface. In the studio, it's easy to make banks of sliders to control a mix and to touch and control many sliders at once. With the added ability to program a softer interpolation between touched points, large adjustments in a mix can be rendered smoothly.

The Lemur is easily my favorite hardware controller, and I hope it becomes popular (especially so that the price will drop). It seems to have a few problems that still need sorting out, including infrequent freezes of both its hardware and its software, and there is certainly a learning curve. But if you're aching to get away from your usual keyboard-and-knobs controller, and you want to take a step into the future, this is a great place to start.

Jonathan E. Segel is a musician and composer of various types of music. Learn more at www.magneticmotorworks.com.

FIG. 1: KRK's RP-6 produces surprisingly strong bass response from its 6-inch driver.

KRK SYSTEMS RP-6

Budget powered monitors with beefy bottom.

By Rusty Cutchin

KRK's pro close-field monitors have traditionally been a welcome relief from the clinical hyperaccuracy of certain studio monitors that can kill your love of music as they help you create a reliable mix. Given a choice between entrenched standard-issue speakers and KRK models, many engineers prefer the latter, especially the company's midline and high-end monitors, which offer a clear sound, ample power, and the courage to be different, despite the speakers' yellow-bellied appearance.

But despite that consistently good impression, I was stunned by the sound quality of KRK's "entry-level" RP-6, a member of the company's Rokit series of powered monitors. The RP-6 (also known as the Rokit Powered 6) is an amazing performer when configured in a stereo pair. The monitor delivers crisp highs from its soft-dome tweeter and room-filling bass from its 6-inch driver. Teamed with the company's RP-10S subwoofer (see the December 2005 issue of EM or www.emusician.com for a review of KRK's RP-10S subwoofer and V8 monitor), the RP-6 gives you a bone-crushing system for creating big mixes in small rooms. To top it off, the system is available at street prices of less than $700.

Package Perfect

Although much of the RP-6's great value can be attributed to industry developments such as general improvements in monitor design and cost savings from foreign manufacturing, the RP-6 (see **Fig. 1**) fuses several advantages expertly. Its components are housed in a sleek 10½-inch-deep cabinet with rounded corners to cut down on edge diffraction. A slotted front port runs across the bottom of the front panel. The tightly packed 6-inch glass aramid composite woofer and 1-inch ferrofluid-cooled neodymium soft-dome tweeter are housed neatly in a figure-8 rim. The woofer's yellow cone balances nicely with the amber power indicator that's nestled in between the *K* and *I* of the Rokit logo.

The input options on the RP-6's rear panel (see **Fig. 2**) make it ready for any studio situation. You can choose from balanced XLR and TRS inputs as well as unbalanced RCA. A volume pot runs from –30 dB (off) to 6 dB with 0 at the 12 o'clock position. A 4-position high-frequency level adjust pot can be set to –2, -1, 0, or +1 dB to fit your room. A large power toggle switch and power

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REVIEW

cord receptacle round out the rear panel. The RP-6's power section supplies 50W to the woofer and an efficient 18W to the tweeter.

The RP-10S, which was supplied for this review, is an excellent fit for the RP-6s, although it doesn't have TRS outputs. In most cases, you would run your main L-R feed from your board or audio interface directly to the subwoofer, and then pass the high-frequency stereo component on to the RP-6s through the RP-10S's XLR or RCA outputs. The RP-10S's crossover-pot setting is continuously variable from 50 Hz to 130 Hz. The gain pot duplicates the RP-6's range of –30 dB to 6 dB.

RP-6 SPECIFICATIONS

Analog Inputs	(1) balanced XLR; (1) balanced ¼" TRS; (1) unbalanced RCA
Frequency Response	49 Hz–20 kHz, ±1.5 dB
High-Frequency Driver	1" soft dome
Low-Frequency Driver	6" aramid glass fiber
Power Rating	18W (HF); 50W (LF)
Signal-to-Noise Ratio	84 dB (HF); 95 dB (LF)
T.H.D.	0.09% (HF); 0.01% (LF)
Input Impedance	10 kΩ (balanced, unbalanced)
Crossover Frequency	2.6 kHz
Subsonic Filter	40 Hz
Dimensions	8.8" (W) × 12.9" (H) × 10.5" (D)
Weight	23 lbs.

Battle of the Sixes

I wasn't expecting much when I first connected the RP-6s. Although I picked up a pair of KRK's Rokit-brand monitors several years ago and have been happy with them in my home entertainment system, I never considered buying them for serious studio work. The line disappeared for a while as KRK introduced other models, including its recently redesigned V-series monitors. As a result, these new Rokits, with their yellow cones, might have indicated a desperate grab at the low-end market rather than a budget model worthy of the KRK imprimatur.

I knew that the RP-6s were in the latter category within a few seconds of pumping everything from 50 Cent to full-orchestra rerecordings of early-20th-century show scores into them. Highs were smooth and distortion-free without sinking into a pool of shimmering mess. Midrange elements were easily distinguishable, and stereo imaging was acceptable, if not exceptional.

What really surprised me was the RP-6's bass handling when set up as a stereo-only monitor system. I generally prefer to work with 8-inch or larger drivers with a frequency response down to at least 30 Hz or so. That gives me enough bass for accurate mixing without needing to set up a subwoofer for low-end enhancement. A stereo pair with 6-inch drivers almost forces you to wor k with a subwoofer, unless you always work with very low levels.

The RP-6s sound great, while offering good bass response—as ample a bass volume as I've experienced with 6-inch drivers. (The RP-6's low-end frequency response extends to 49 Hz.) The trade-off here, as I expected, was an increase in muddiness in the low mids, typically evidenced by a loss of definition in fingered electric bass parts. Adding the RP-10S subwoofer enabled me to have more control over that trade-off.

Sub Satisfaction

What's more, the RP-10S is a great complement. The subwoofer looks and sounds perfectly matched to the mains, and delivers just the kind of bottom you'd expect; its front-firing 10-inch low-frequency driver puts the punch of a club's worth of sound reinforcement in a small control room, and with the RP-10S's amp delivering 150W of bass energy, the RP-6s are free to pump out crisp mids and highs, which they did with only a slight audible midrange scoop. (A quick trip to 50 Cent's Web site to listen to the groove on "Hustler's Ambition" confirmed my earlier impression that the RP-10S rocks the house on hip-hop material.)

As I have worked with more subs over the past few years, I've become increasingly comfortable with speaker placement and controls, and the simplicity and range of this Rokit configuration was a delight to work with. The RP-6s, which sounded fine alone, suddenly jumped up a notch in effectiveness. With the sub set to handle only the lowest frequencies (below 80 Hz), acoustic piano parts had an enveloping richness rather than the focused directionality they exhibited in the stereo pair. Adjusting the sub's

PRODUCT SUMMARY

KRK SYSTEMS RP-6

powered monitors
$399.99 each

PROS: Excellent sound. Exceptional bass response. Multiple inputs.

CONS: Slight low-end muddiness.

FEATURES
EASE OF USE
AUDIO QUALITY
VALUE

MANUFACTURER
KRK Systems
www.krksys.com

CHUCK DAHMER

FIG. 2: The RP-6 provides a full complement of inputs, including balanced XLR, balanced ¼-inch TRS, and unbalanced RCA.

gain control to near zero imparted that effect across several instrument groups, while raising the gain control made the system jump with energy when kicking a dance track. It's hard to imagine even bleary-eyed club DJs not being impressed by the bass energy churned up by this system.

Win-Win

KRK's Rokit line is best suited to the small composing or project studio, but that doesn't mean that seasoned pros should cross them off their shopping lists. (KRK also sells the RP-5 and RP-8 monitors, which offer 5- and 8-inch drivers, respectively.) It's a measure of the company's faith in the RP-10S, for example, that it was sent as a companion to the midline V8 monitor that EM recently reviewed. Studio owners with 24/7 operation or critical full-time mix assignments will want to check out the higher-end designs that put KRK on the map. The company's offerings consistently have been worth appraising for studios of all sizes.

But the Rokit Powered 6 is a unique animal. It offers surprising bass handling and respectable high-end clarity at a great price. Used with the RP-10S, the RP-6 has all the elements of a great 2.1 system: excellent sound, dependable accuracy, great power handling, an easy fit into any production environment, and a system cost that beats the price of some single studio monitors.

Rusty Cutchin is an associate editor of EM. He can be reached at rcutchin@comcast.net.

FIG. 1: The EastWest/Quantum Leap *Symphonic Choirs* library uses a dedicated version of Native Instruments Kompakt as the front end.

EASTWEST/ QUANTUM LEAP *Symphonic Choirs* (Mac/Win)

A sampled choir library that really has something to say.

By David M. Rubin

The four-volume EastWest/Quantum Leap *Symphonic Orchestra* library broke new ground as the first 24-bit orchestral library with multilayered samples of every note recorded simultaneously from three different mic positions: close-up (C) for individual details; a front-of-stage array (F) for a full stereo sound; and a back-of-the-room setup (S) to capture the natural reverb of the recording space. For *Symphonic Choirs*, producers Doug Rogers and Nick Phoenix have again teamed up with recording engineer Keith O. Johnson. Applying the same recording techniques (with the same equipment) in the same concert hall, they've created a unique, high-end sampled choir library that installs from nine DVDs and weighs in at a hefty 38 GB.

As in the *Symphonic Orchestra* library, the samples from the three mic positions are phase-aligned. You can freely combine them to deliver just the right amount of reverb or to move groups or soloists closer (more C, less F, little S) or farther away (little or no C, a soupçon of F, heavy on the S) in the final mix.

That level of control makes the orchestral and the choir libraries ideally suited to 5.1 surround mixes: record a matched set of patches, and use one of the C patch's channels for the center-front, both sides of the F patch for left- and right-front, and the S patch for the rear surround channels. Working with the room's great-sounding natural reverb eliminates the need (in most projects) for external reverb units or plug-ins. And it enables *Symphonic Choirs* to integrate seamlessly with the *Symphonic Orchestra* collection, yielding a lush, convincingly real-sounding, organic blend that would be difficult to re-create with other libraries. Furthermore, *Symphonic Choirs* includes a powerful application called WordBuilder that further raises the level of realism by enabling the sampled choirs to actually pronounce words and syllables instead of repeating the same sound over and over.

Voice Box

Symphonic Choirs employs a dedicated version of Native Instruments Kompakt as its front end (see **Fig. 1**). Kompakt functions in standalone mode or as a plug-in under a VST 2, a DXi 2, an AU, or an RTAS host. As in the orchestral library, you can load multipatch presets (multis), or you can load individual Instruments (in this case, choir sections or soloists) into any of Kompakt's eight slots. If your computer has the horsepower, you can get as many as 256 notes of polyphony. (For more on using a 3-mic-perspective library and dealing with its

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substantial processing demands, see the EastWest/Quantum Leap *Symphonic Orchestra* review in the December 2004 issue of EM.)

Symphonic Choirs consists of five sections: soprano, alto, tenor, and bass (SATB); and a separate boys' choir. Solo samples are also provided for soprano, alto, and boy vocal parts (although they can't be used with WordBuilder). All of the patches sound fantastic and are wonderfully expressive, with a great sense of presence and smooth transitions from one Velocity layer to the next. I was disappointed, however, that there weren't any solo tenor or bass patches. Those would have come in handy for creating the sound of a small group of soloists, such as a Christmas-carol ensemble or a barbershop quartet.

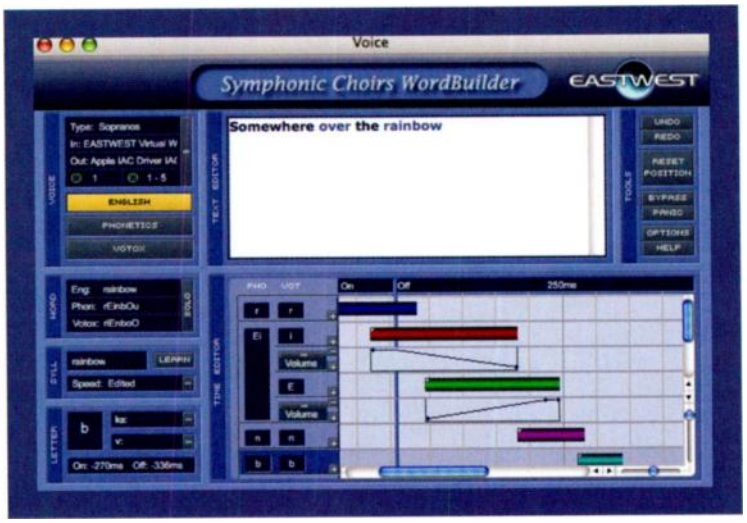

FIG. 2: WordBuilder is a powerful program that converts written text into its phonetic components and then triggers the appropriate samples in the *Symphonic Choirs* library. The colorful graphic display lets you edit the playback of individual vowels and consonants.

The SATB sections are offered with Normal, Legato, Staccato, and Slurred articulations (keyswitching changes from one to another); the boys' choir includes only Normal and Legato options. The mod wheel is used mostly to crossfade between samples with more or less vibrato and with a more-or-less forceful delivery ("hard" as opposed to "soft"). Nonvibrato patches are also provided.

It's easy to create any kind of choir by loading the appropriate section patches into the Instrument slots in Kompakt. Several ready-made, 5-octave Full Chorus Church patches are also included, but with only the S mic perspective. They're great for improvising or for quickly sketching out a part. The mod wheel lets you dial in a light amount of vibrato. The overall sound is warm and rich, and it may be all you need for many projects.

PRODUCT SUMMARY

EASTWEST/QUANTUM LEAP — Symphonic Choirs

sample library
$895

PROS: Top-notch 24-bit recording quality. Full collection of vowel and consonant samples. WordBuilder application enables pronunciation of lyrics. Three-mic approach offers excellent control over natural ambience. Standalone and plug-in options available. Full surround-sound capability.

CONS: High processing demands. Lacks solo tenor and solo bass samples. WordBuilder does not work with the solo patches.

FEATURES	
EASE OF USE	
AUDIO QUALITY	
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MANUFACTURER
EastWest/Quantum Leap
www.soundsonline.com

Wordplay

Most sampled choir libraries consist entirely of a few vowel sounds (such as *ah*, *oo*, or *ee*) sung at different pitches. *Symphonic Choirs*, of course, has all of those sounds, and many musicians buy the library just to use it in the traditional way for backup harmonies and pads. But *Symphonic Choirs* has far more to offer than two or three vowel sounds. In fact, it provides a complete set of vowel sounds, such as *oh*, *ah*, *oo*, *ee*, *uh*, and *ih*, as well as more than two dozen consonant sounds (including a rolled *R*). In theory, you could create words by triggering different vowel and consonant samples from different patches on different MIDI channels, but constructing words from scratch would be an overwhelming task. Fortunately, *Symphonic Choirs* comes with WordBuilder, a program that enables your virtual vocalists to pronounce lyrics as they sing (see **Fig. 2**).

To use WordBuilder, you must first load one of the specially designed multis into Kompakt. These multis consist of five presets that consume a block of five consecutive MIDI channels. Together, the presets supply all of the necessary phonetic sounds for producing words. Type your lyrics into the Text Editor section of the program, and based on its internal 100,000-word pronunciation dictionary, WordBuilder instantly breaks down the words into their phonetic elements. It also simultaneously translates the English words into the standard phonetic alphabet and into the program's own Votox alphabet. Buttons on the left let you choose which kind of text appears onscreen. For example, the phrase *Fly me to the moon* is translated as "flAi mi! tu! d!a mu!n" in the standard phonetic alphabet and as "FlaE mEE TO t!u mOn" in Votox. (Comparison tables and explanations are provided in the documentation.)

Beneath the Text Editor section, the Time Editor shows the text in a sequencer-like graphic display. Each word's phonetic components appear as colored bars along a timeline where they can be shortened, lengthened, crossfaded, time-shifted, made louder, and adjusted in a variety of ways. That's important because you can't just

type in lyrics and have them come out sounding perfect right away. In most cases, you'll have to carefully edit each word to get the performance that you want. Many common words have multiple pronunciations, and most words need to be adjusted to emphasize the right syllable and to properly match the pronunciation to the tempo of the music. WordBuilder's Learn feature makes it easier to adapt the syllable lengths to the note lengths, and the Solo function isolates a word, allowing you to focus on it until you get the desired pronunciation.

For example, when I typed in the word *valley,* the choir seemed to swallow the second syllable a bit on some of the notes. I located the second syllable (it had two elements), stretched it a bit, adjusted its volume curves, and slid it back a tad to reduce the overlap with the first syllable. That did the trick. A little more fine-tuning, and I had the sound that I wanted. Adding a slight gap before consonants and boosting their Velocity offset often helps to emphasize them; adjustable controller envelopes for each phonetic element provide plenty of control in shaping the pronunciation. WordBuilder can also trigger the keyswitches that change the choir's attacks from Normal to Legato, Staccato, or Slurred. (You can learn more about using the program at www.soundsonline-forums.com.)

EastWest recommends that you learn WordBuilder's Votox alphabet. You can then enter exactly what you want without relying on the program's interpretive skills. You'll also have to use Votox (or the standard phonetic alphabet) if you want to enter foreign languages and non-English phrases, such as *be-bop-a-doo-wop*. You can even adjust the English pronunciations to take on a regional accent. WordBuilder's Add dialog box lets you save your new alternate settings for future use. (EastWest has also created a free database of additional Votox words and phrases for downloading.)

If you have Steinberg Cubase SX or Nuendo versions 2 or 3, you can use WordBuilder as a VST Module Architecture MIDI plug-in. In Cakewalk Sonar 4, WordBuilder can run as an MFX plug-in. PC users will have to add a MIDI loopback program to supply the virtual MIDI cables when WordBuilder isn't used in plug-in mode.

Well Spoken

Based strictly on its own merits, *Symphonic Choirs* is undeniably a first-rate library. In fact, it's one of the nicest and most flexible sampled choir libraries I've used. The 24-bit recording quality is top-notch all the way, and the chromatically sampled, multilayered patches are consistently responsive and realistically expressive. I also enjoyed experimenting with some of the nonstandard patches, including clusters, crescendos, falls, and especially the slightly creepy whispered words.

Adding WordBuilder to the package raises everything to a whole new level. If you're willing to invest the time and energy in shaping your performance, the results can be rewarding. You do, however, have to dive in and muck about quite a bit with phonemes, diphthongs, and other phonotactic elements of speech to get a performance that is natural-sounding and reasonably intelligible. You may never get the level of clarity and enunciation that a good live choir with an experienced choir director can deliver. But you can definitely create a virtual choir that sings words instead of just repeating vowel sounds, and that's pretty amazing. For some great MP3 demos of *Symphonic Choirs* in combination with *Symphonic Orchestra,* go to the EastWest Web site (www.soundsonline.com) and click on the EW/QL Choirs link.

David M. Rubin lives and works in the foothills outside of Los Angeles. He is the author of Power Tools for Peak Pro *(Backbeat Books, 2005).*

QUICK PICKS

UNIVERSAL AUDIO

Plate 140 (Mac/Win)

By Eli Crews

EM ★★★★★ hot pick

As the name suggests, the Universal Audio Plate 140 reverb plug-in ($149), designed for use with the UAD-1 accelerator card, is modeled after the German-made EMT Plate 140. A plate reverb is a large sheet of metal that vibrates when you pass audio through it. To shorten the decay time, an asbestos damping pad is moved close to the plate using a remote control. This technology, as esoteric as it may sound, was one of the main methods of obtaining artificial ambience prior to the advent of digital reverb.

Universal Audio Plate 140 offers three different plate reverb settings, each of which can have its own amount of damping.

To use Plate 140, you need a UAD-1 PCI card and either a computer with a PCI slot or a card cage. The plug-in supports VST, Audio Units, and RTAS hosts. (Pro Tools users will need to use the included VST-to-RTAS wrapper from FXpansion.)

Plate Tectonics

Plate 140 was modeled after three unique plates at the Plant Studios in Sausalito, California, all of which have completely different sonic fingerprints. The plug-in has just a handful of parameters. But once you hear this effect in action, you'll realize that those are the only controls you will need to obtain stunning reverb sounds.

The front panel's chicken-head knob selects one of the three plate models. Each model has its own damping control, and you can set the reverb time between 0.5 and 5.5 seconds. The on/off switch lets you disengage the unit without using your DAW's bypass, which is essential given the latency issues (more on that in a moment). Controls on the Universal Audio Plate 140 that aren't found on the original are High and Low Shelving EQ, Pre-Delay, Stereo Width, and Wet/Dry. All of these add functionality while keeping the interface simple.

Hard vs. Soft

For the review, I compared the Plate 140 plug-in with its hardware namesake, as well as with the reverb plug-ins I normally rely on. Because reverb plates are expensive and weigh about 425 pounds each, you won't find them in the average project studio. However, I located some EMT Plate 140s at two Bay Area studios—Tiny Telephone in San Francisco and Fantasy Studios in Berkeley—and both studios were kind enough to allow me to use their reverbs for this review.

I brought in a Pro Tools session and ran the lead vocal, snare drum, and overall mix through the plates, with various decay times. It was difficult comparing the plates, because each of the hardware ones has a unique sound. After hearing the hardware and software plates in an A/B comparison, however, I realized that I like the sound quality of Universal Audio Plate 140 just as much as that of its metal counterpart: the plug-in held its own in terms of depth and resonance, despite the varied sonic characteristics of the hardware plates (see **Web Clip 1**).

Then I compared Plate 140 with my two main reverb plug-ins. Plate 140 blew the plate presets of my standard reverb plug-in out of the water. That plug-in is more versatile, of course, with numerous algorithms and controls. But when comparing apples with apples, Plate 140's sound was far superior.

When I compared it with a popular sampling reverb that uses impulse responses of many of the world's finest hardware plate reverbs, Plate 140 stood up nicely. There was no clear winner here—I'd call it a draw, with a very slight edge given to the convolution reverb for its ability to offer more plate choices.

Because the UAD-1 plug-ins run off a PCI card, noticeable latency is introduced. Most hosts that support the UAD-1 have automatic latency compensation, although even in Digital Performer 4.6 I still experienced slight latency problems. Pro Tools LE doesn't have latency compensation, so I had to use aux tracks and dummy plug-ins to sync everything up. (The UAD-1 manual describes that process thoroughly.) The latency is a slight annoyance, but a very small price to pay for reaping the aural benefits of the UAD-1 plug-ins.

A Clean Plate

The simplicity and sonic fidelity of Plate 140 make it my new "go-to" reverb, and it has excelled in almost every situation I have used it in. If you are looking for lots of knobs to tweak, look elsewhere. But if you're looking for an easy-to-use and excellent-sounding reverb, and you need an excuse to get the UAD-1 card, look no further.

EM WEB CLIPS

Overall Rating (1 through 5): 5
Universal Audio
www.uaudio.com

DIGITECH

DF-7 Distortion Factory

By Rob Shrock

The DigiTech DF-7 Distortion Factory ($149.95) is a stompbox that models seven popular guitar distortion pedals: the Ibanez TS-9 Tube Screamer, the DOD Overdrive/Preamp 250, the Boss DS-1, the ProCo Rat, the Boss MT-2 Metal Zone, the DigiTech Metal Master, and the Electro-Harmonix Big Muff π. Not only does the DF-7 emulate the range of each pedal remarkably well (see **Web Clip 1**), but it also provides controls

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that in some cases allow tonal shaping beyond the capabilities of the originals.

Knob Central

The DF-7 has three dual-concentric pots and a single model-selector knob. The controls are tightly situated next to each other, but I had no problem tweaking them. The unit is powered by a 9V battery or by the optional PS200R adapter ($24.95). I highly recommend purchasing the adapter, because the battery will run out after about four hours of use.

The controls take some getting used to because the names of the knobs vary from the original pedals. For example, the High control on the DF-7 acts as a basic treble EQ for the DOD 250 (which has no EQ control), MT-2, and Metal Master; as the original Tone control for the TS-9, DS-1, and Big Muff π; and as the Filter control for the Rat.

All of the controls on the originals are available on the DF-7. Running the unit's Low, Mid, and High knobs straight up properly matches the flat settings of the modeled stompboxes. For the most part, the Gain and Level knobs are the same as on the originals and respond similarly. If you're the "twist-until-it-sounds-good" type, you won't be disappointed. There is plenty of room for tonal shaping.

The DigiTech DF-7 gives you accurate models of seven classic distortion pedals, and it does cabinet simulations, too.

I own several of the pedals modeled by the DF-7, so I was able to make direct comparisons. The DF-7 accurately emulated the edgy buzz of my ProCo Rat and the bottomy *fizz* of the Big Muff π. I actually preferred the modeled Big Muff π because it was cleaner sounding than the original. The DF-7 was a tad brighter than my DS-1 and had more pronounced upper midrange response than my original DOD 250, but a little EQ shaping on the DF-7 corrected the differences.

Comparing the DF-7's TS-9 emulation was a little more involved because I have a customized boutique Ibanez TS-9DX, which has more bottom end and produces slightly less distortion than the original. I was able to shape the DF-7's tone to get it very close to my TS-9DX by backing down the gain and adding more low and high EQ.

I'm not a big fan of metal pedals, but I must admit that the DF-7 did a good job of impersonating the MT-2 and the Metal Master. Both emulations provided plenty of demonic tone.

New Tricks

The DF-7 gives you two outputs: Amp and Mixer. The Amp out is the regular instrument-level stompbox signal; the Mixer out is designed to go into a full-range system and uses a cabinet simulator. Each distortion model has a fixed, specific cabinet model designed to match with the basic tone of the original pedal. The cabinet simulator is quite useful for DI recording of guitar tones. While the DF-7's tone is not as sophisticated as a Line 6 Pod's, for instance, I can imagine many instances in which it would work just fine in a recording.

The DF-7's flexible output modes give you three options: mode 1, the default, gives you a mono signal for an amp and a cabinet-simulated mono signal for a mixer. Mode 2 gives you a cabinet-simulated left and right output to feed a stereo pair of mixer channels. Mode 3 provides left and right outputs, with no simulator, for connection to a pair of amps. In modes 2 and 3, you can choose between six different stereo spectrums, from narrow to ultrawide. That's a pretty impressive feature for a stompbox.

Pedal Tones

Guitar pedals have long been favorites of music producers for processing other instruments. With its built-in cabinet simulation and its tweakable stereo spectrum, the DF-7 is primed for that role as well. Try it on keyboards, loops, or vocals, and chances are you'll love the results.

I'm not typically a fan of digital stompboxes, because I tend to go for a lo-fi, analog-funk sound. In the case of the DF-7, however, I'm willing to make an exception. It provides plenty of power in a single effects pedal, is reasonably priced, and sounds great on just about anything. EM WEB CLIPS

Overall Rating (1 through 5): 4
DigiTech
www.digitech.com

PSP AUDIOWARE

PSP 608 MultiDelay (Mac/Win)

By Len Sasso

PSP Audioware has come up with another top-notch delay-based plug-in. The PSP 608 MultiDelay ($149) is an 8-tap delay line with the great sound quality, ease of operation, and full complement of bells and whistles for which PSP has become renowned. Each tap has its own multimode-filter and tape-saturation (called Drive) modules, as well as individual modulation and reverb routings. Add two operation modes (called Multitap and Multidelay), MIDI remote and host automation, and a large, interactive LCD-style display, and you're in for a lot of bouncing around.

The PSP 608 comes in VST, Audio Units, RTAS, and HDTM formats for Mac OS X 10.3 Panther and later, and it comes in VST, DirectX, and RTAS for Windows 98 and later. It will work at all popular bit depths and sampling rates, including 24-bit, 192 kHz. Delivery is by download and includes all formats, as well as a 47-page PDF manual. The manual is an essential read if you want to get the most out of the PSP 608, but you can go a long way with the simple, one-page Quick Start instructions.

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On the Right Path

Each of the eight delay taps has the same signal path, which starts with a delay buffer having a range of 1 ms to 8 seconds. The delayed signal is then run through an input gain stage, followed by stereo width and balance controls. The filter and drive stages follow, both of which can be bypassed. Finally, there's a send bus to the global reverb stage. Feedback can be routed back to the input either directly from the delay buffer (pre) or from the output of the drive stage (post).

Controls for each of the PSP 608's taps are identical, and the LCD-style display at the top annotates the settings when the mouse passes over a control.

From an operational standpoint, setting up the PSP 608 taps is as simple as turning them on (you can use any number of them), deciding whether to apply feedback and how much, and setting the delay-time, gain, width, balance, filter, drive, and reverb-send controls.

The PSP 608's resonant filters have eight modes, including lowpass, bandpass, highpass, peak (single-band EQ), and two varieties of low and high shelving filters. The filter cutoff frequency can be modulated by a mix of a global LFO and a global envelope follower. The envelope follower can be fed by the input signal, the master output signal, or the individual output of any of the taps, allowing you to influence the filter of one tap by the rhythm of another.

The signal path ends in an optional Master Feedback section with its own filter and drive modules. That is where the PSP 608's two modes differ. In Multidelay mode, each tap has its own feedback loop, and one tap is selected to feed the Master Feedback buffer. In Multitap mode, only the Master Feedback buffer is active. In either mode, one of the tap sliders is assigned to set the Master Feedback delay time (see the graphic and **Web Clip 1**).

All Those Knobs

With 101 knobs, 78 buttons, and eight sliders, the PSP 608 control panel at first appears daunting, but once you consider that each tap control is replicated eight times, it all makes a bit more sense. To make things even easier, the large LCD-style display annotates all settings as controls are passed over with the mouse, and many settings can also be made directly in the display. The display even has an alternative mode that shows each tap's position along a timeline. Unfortunately, that part of the display is not interactive, but it is still quite useful.

Most PSP 608 controls (some buttons and elements of the LCD-style display are exceptions) can be controlled remotely by incoming MIDI, and a MIDI Learn function makes it easy to set up. With so many controls, setup can still be tedious, but, conveniently, you can save MIDI-remote setups to disk. Unfortunately, control settings made by MIDI remote are not reflected in the LCD-style display.

For the tweak averse, the PSP 608 comes with a complement of more than 100 presets ranging from mundane multitaps to otherworldly resonator effects, and there are a variety of presets labeled "init" that are useful as jumping-off points. A multitap delay is an important part of every electronic musician's effects palette, and for quality and ease of use, it's hard to think of a better or more economical choice than the PSP 608.

EM WEB CLIPS

Overall Rating (1 through 5): 4
PSP Audioware
www.pspaudioware.com

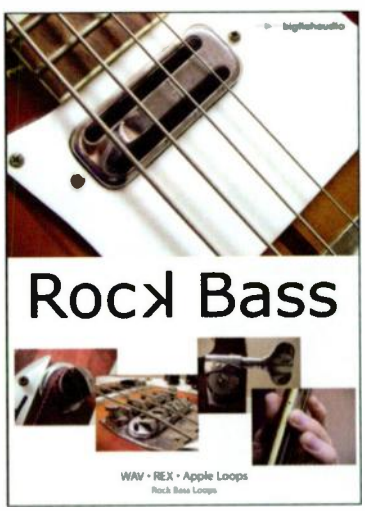
BIG FISH AUDIO

Rock Bass

By Doug Eisengrein

Rock Bass ($49.95) is a genre-specific loop and sample collection of rock- and funk-inspired bass licks. It comes on CD and contains 660 MB of loops and one-shots. Eight folders of loops with tempos ranging from 80 to 120 bpm are arranged by tempo and key, and are provided in WAV, REX2, and Apple Loops (AIFF) format. A ninth folder of one-shot samples rounds out the collection. *Rock Bass* was produced by Pavel Stepanovsky, with editing by Steven Bolar and Mark Rinewalt.

All content is recorded in monophonic, 16-bit resolution at a 44.1 kHz sampling rate, which is appropriate for bass and reduces sample size and load time. The folder- and file-naming scheme makes selecting loops a simple affair. The primary directories are named Apple Loops, REX, and WAV. Each directory houses seven loop folders, named for the loops' original tempos, and an eighth folder of effects-processed loops. The WAV and Apple Loops directories contain an additional folder for the one-shots. Each loop's file name consists of an index number followed by the loop's tempo and key. The one-shots substitute the words "One-Shot" for the tempo.



***Rock Bass* is a varied collection of rock 'n' roll bass loops and one-shots in WAV, REX2, and Apple Loops format.**

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The Long and Short

The number of bars within the loops varies, but the majority of loops are two or four bars long. The one-shots are not always what you might expect—some are single notes, whereas others are licks that haven't been trimmed to loop. Those make good lead-ins, outros, and solo bridges, but, unfortunately, there are only 26 of them (see **Web Clip 1**).

I used Ableton Live 5 and Propellerhead Reason 3.0 to audition and create rough sketches with all of the loops. Live is an ideal environment for trial-and-error loop composition, and Reason has an excellent REX file player.

Although not recorded at the highest resolution, the loops are up-front and provide a good, clean foundation. All samples use electric bass, and the playing appears to be by a single musician. The bass playing is superb, perhaps even a bit too perfect for my taste. A substantial number of the licks are funky, harmonic, and rhythmically complex—not exactly what one might expect in a rock-bass collection. The content probably has more in common with Red Hot Chili Peppers or Tortoise than, say, the bass line of Deep Purple's "Smoke on the Water."

Mix and Match

The good news is that there is a lot of variety. The loops range from high to low octaves, and many are suitable backbones for songs. On the other hand, the loops are wide-ranging in style, and finding loops that match can be a challenge. For genres such as hip-hop and electronic pop, a single bass loop running throughout a song may be fine, but for rock, blues, and funk, that usually doesn't work. Matching time-warped loops of the same key but different tempo is a partial solution.

The files in this collection are well edited and loop seamlessly. Because the rhythm is rock solid, sketching out song ideas over drum loops is a breeze. On the downside, several of the one-shots are recorded with effects built in, which makes them difficult to match with other samples from the collection. I am a strong believer that effects should be added after recording.

Rock Bass is a great collection of useful bass riffs. The sound quality is good and the loop editing is nicely done. I would have liked a little more matching material, especially in the one-shot category, but considering the modest price, the collection is a good value. EM WEB CLIPS

Overall Rating (1 through 5): 3
Big Fish Audio
www.bigfishaudio.com

RAYZOON TECHNOLOGIES

Jamstix (Win)

By rachMiel

Jamstix, a virtual-drummer VST instrument plug-in, is one of the new breed of interactive computer-based instruments that help composers to make music. It requires Windows XP or 2000, a P3 or Athlon 500 MHz CPU (although

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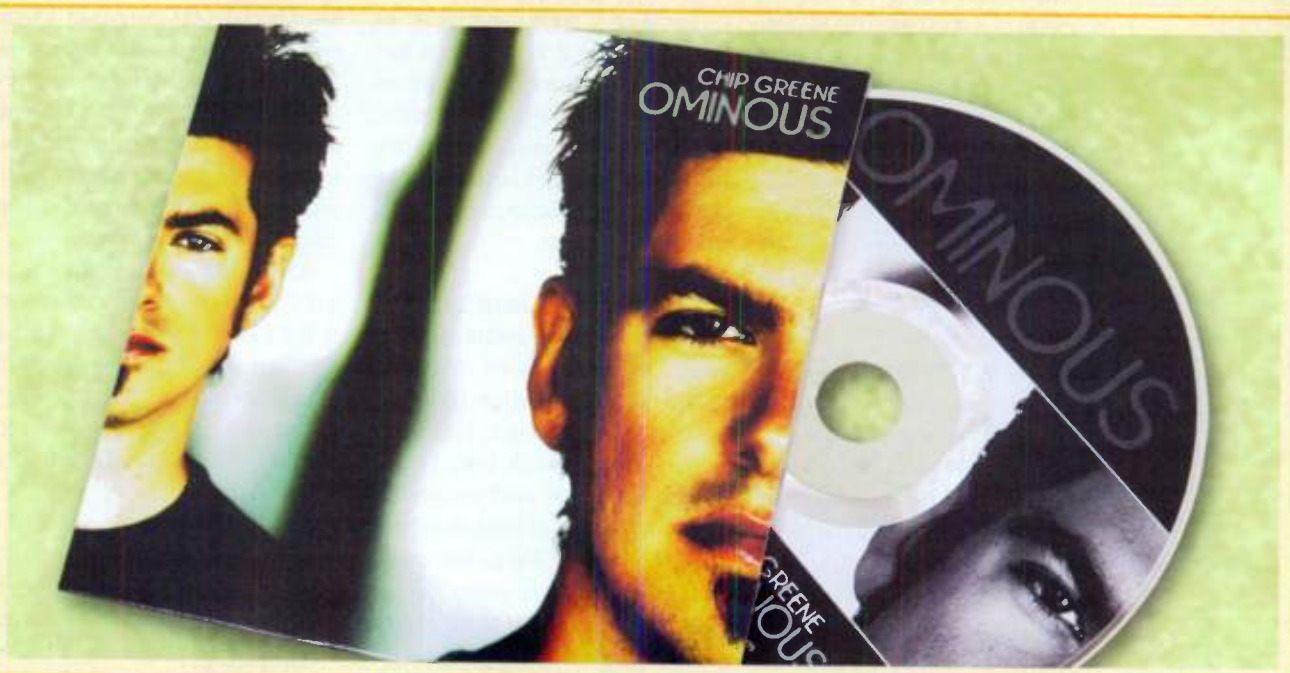

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2 GHz is recommended), 512 MB of RAM, 500 MB of free hard-drive space, and a VSTi 2.0–compliant host. The base model costs $99, but I highly recommend the $129 bundle, which includes two add-on drum packs.

Installation was a breeze: I simply downloaded Jamstix from the Rayzoon Web site (www.rayzoon.com), then unzipped and ran the install executable. I used Sonar 5 as my VSTi host and had no trouble inserting multiple instances of Jamstix and running them on their own using the built-in Jamstix drum brain or running them from accompanying MIDI or audio tracks.

Slammin' Jammin'

Jamstix is really three instruments in one. With its brain turned off, it's a standard drum-kit sound module, which can be driven by MIDI data. With its brain on, it can act as an arranger or jammer. As an arranger, it resembles a sequencer and allows you to create stylistically coherent drum tracks with customized intros, patterns, breaks, fills, and endings (see **Web Clip 1**). Arranging is the right choice if you already have a clear idea of the drum part.

As a jammer, Jamstix generates a drum part in real time to accompany MIDI or audio input, either of which can be provided from a MIDI or audio file or live from a MIDI or audio instrument (see **Web Clips 2** and **3**). You select rhythms from the Jamstix library, edit them as desired, sequence them, and customize the flow by adding an intro, fills, an ending, and MIDI automation.

Jamming is great when you're not sure exactly what kind of drum part you're looking for. You can use a virtual instrument to jam with Jamstix, record your playing while Jamstix automatically records itself, and then switch to manual jam and rearrange patterns, change fills, mute bars, and so on.

Is It Real or Is It Jamstix?

Jamstix's ability to create realistic-sounding drum tracks that jibe stylistically with the other layers in a song is its biggest asset. The realism comes from Rayzoon's Limb Priority Control mechanism, which ensures that the drum part is at all times playable by a human drummer. The stylistic consistency comes from the panoply of customizable style controls in the Jamstix drum brain: Velocity variance, auto cymbals, random accents, probable notes, automatic switch rules, no ghosting on low velocity, and so on. Furthermore, over 50 parameters can be automated from the host.

LiveLoop technology is another great Jamstix feature. It allows you to maintain flexible control over your drums during live performance. You divide an arrangement into sections (called LiveLoops), which are typically verses, bridges, choruses, and so on, then switch between sections with a footswitch or MIDI controller.

The Jam pane of the Jamstix VST window is home base for manual, free, and keyword drum jams.

For the Record

Being a composer more than a technician, I'm a big fan of simple, clear GUIs that practically run themselves. Although Jamstix's GUI isn't the most complex I've seen, it certainly isn't a no-brainer. Much of the complexity is necessary, but the GUI could be a bit more streamlined (see the graphic).

When jamming with Jamstix in Sonar and changing drum-brain parameters in real time, the beat often got out of sync. To resync, I had to stop and restart. According to Jamstix's creator, Ralph Zeuner, that is a known problem with Jamstix in Sonar, but it shouldn't occur in other hosts.

If the Shoe Fits

I found the Jamstix range of styles to be somewhat limited. It covers rock, funk, pop, and blues quite well, but has less to offer for experimental genres such as IDM and electronica. On the other hand, even experimentalists can benefit from the Jamstix groove brain by using it to drive different electronic drum VSTis. Jazz composers should also find Jamstix useful, especially with the recently released BrushPak, which adds a brush kit, sizzle cymbal, swing logic, and snare-brushing control to the mix.

I heartily recommend Jamstix for any PC-based producer of rock, funk, pop, or blues who wants realistic-sounding and -feeling drums in his tracks but does not have the resources to hire a human drummer. Jamstix comes one step closer to the realization of a convincingly human-sounding, all-purpose virtual drummer. Customer support is fabulous. Jamstix creator Ralph Zeuner is a font of Rayzoon wisdom, and wonderfully patient and responsive to his users. And the price is right—at $99, it's practically a steal. EM

EM WEB CLIPS

Overall Rating (1 through 5): 4
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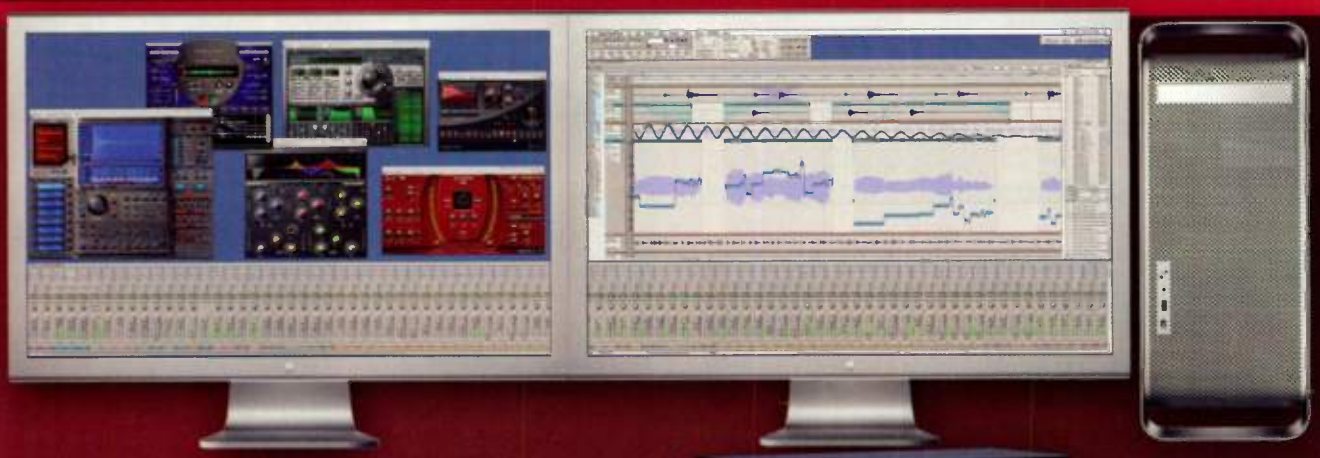

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With two dual-core processors at speeds up to 2.5GHz per core, the Apple Power Mac G5 Quad doubles the punch of its dual-processor predecessor. Do the math: Quad-core processing means four Velocity Engines and eight double-precision floating-point units for blistering performance of up to 76.6 gigaflops. What does that mean for your MOTU Digital Performer studio? Run MachFive, MX4, the Symphonic Instrument and dozens of other virtual instruments, processing plug-ins and disk tracks without even batting an eyelash. This is the wicked-fast Mac that you've always dreamed of. Blaze through your work, deliver ahead of schedule and astound your clients — because this baby really moves. Prices start at $1999.

Large capsule mic

The new RØDE NT2-A can be plugged directly into your 828mkII or Traveler FireWire interface. This professional large capsule (1") studio microphone incorporates three-position pick-up patterns, pad, and high pass filter switches conveniently located on the mic body. At the heart of the NT2-A is the Australian designed and manufactured HF1 dual diaphragm capsule. The frequency and transient response of this new transducer has been voiced to complement today's modern recording techniques, and yet still evoke the silky smooth character of the legendary microphones of the 50's and 60's. These features provide the flexibility and superlative audio characteristics that make the NT2-A one of the most versatile condenser mics available. The NT2-A's variable controls allow switching between Omni, Figure 8, and Cardioid polar patterns. The three position high-pass filter provides a flat response or an 80Hz and 40Hz high pass filter. The microphones Pad can be switched between 0 dB, -5dB and -10dB. The NT2-A comes in a soft pouch with an M2 stand mount.

Stackable MOTU audio I/O

All MOTU FireWire interfaces, including the Traveler, 828mkII and 896HD, are stackable, giving you a cost-effective, expandable system that delivers stunning quality and performance. You can daisy-chain up to four MOTU interfaces to your Mac — even the sleek and portable PowerBook — and record all inputs simultaneously. For example, you could connect four Travelers directly to a PowerBook to record 64 inputs to 64 tracks simultaneously at 48kHz. If you connect four 896HDs, you can record 72 inputs to 72 tracks — all to the internal hard drive. On today's multi-processor G5's, you can expand even further with a PCI FireWire card. With four Travelers connected to the on-board FireWire bus, plus a fifth Traveler connected to the PCI FireWire card, each with 20 inputs, that's a whopping 100 inputs recorded simultaneously to 100 separate tracks. Make no mistake: a MOTU native system with multiple interfaces delivers astonishing performance and value.

Unprecedented Native Studio Power

Wireless transport control.

Looking for transport control of Digital Performer that goes way beyond your mouse or the extended keypad on your Mac keyboard? Try the new Frontier Design Group TranzPort, the world's first wireless DAW remote controller. This convenient, compact unit frees you from your conventional position, sitting in front of your Mac. Now you can control Digital Performer from anywhere in your studio. TranzPort has plug-and-play compatibility with DP, thanks to its dedicated DP control surface plug-in software. In addition to controlling all of DP's transport functions, you can also arm tracks for record, set markers, punch in/out, start loops and more using TranzPort's intuitive interface. You also get real-time feedback on signal levels, timecode position, track names and more via the backlit LCD and LED indicators. Controlling DP has never been more fun, convenient and flexible!

Compact MIDI control.

Looking for the ultimate compact keyboard controller for your MOTU studio? The Alesis Photon X25 Portable 25-key USB MIDI controller/audio interface delivers the revolutionary Alesis Axyz controller dome and ten 360-degree rotary knobs, giving you powerful hands-on MIDI control of your Digital Performer studio and software plug-ins. Advanced features include 24-Bit 44.1/48 kHz USB audio I/O with balanced stereo audio inputs and outputs, 25 key, velocity sensitive keyboard, full-size pitch and modulation wheels, and an LCD screen with dedicated encoder for fast and easy set-up.

On-demand Waves DSP.

For large-scale multitrack recording systems, it is good practice to offload plug-in processing from you host computer. The Waves APA-44M delivers on-demand Waves processing to your MOTU native desktop studio via standard Ethernet. Open your existing Waves plug-ins as usual in Digital Performer via the new Waves Netshell™. But now you can run up to 6 Waves IR-1 Convolution reverbs at 44.1kHz at once, and save your CPU power. Need more Waves processing? Just add another APA-44M with the snap of an RJ45 Ethernet cable. It's that simple. For extreme processing needs, connect up to 8 units to your network. The APA-44M is equally at home connected to a laptop, desktop or both. Just transfer your Waves authorized iLok. You can even share a stack of APA-44M's among several computers across the Waves Netshell network. The APA-44M ushers in a new era of state-of-the-art, distributed-network Waves processing for your MOTU multitrack studio.

5-bay removable storage.

The Glyph GT 205 is a 2U five-bay FireWire enclosure offering many advantages for large-scale multitrack recording, including hot-swap portability and convenience. Specifically designed for applications requiring multiple drives, it can be configured with four FireWire hot-swap GT Key drives up to 500GB each. Its expansion bay offers the option of AIT backup, a SCSI or FireWire hot-swap receiving bay, DVD-R/RW or CD-R/RW. Using Glyph's proprietary Integrity™ hot-swap technology, you can easily shuttle content to other GT Series enclosures. To keep your studio quiet, GT Keys incorporate sound-dampening composite metal technology in their frames. Includes 3-year warranty, plus overnight advance replacement warranty in the first year for GT Keys.

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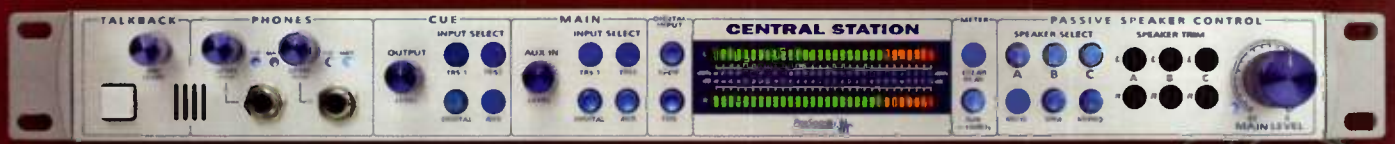
Or, perhaps we should say, it's the perfect finishing touch.



The control room.

The PreSonus Central Station™ is the missing link between your MOTU recording interface, studio monitors, input sources and the artist. Featuring 5 sets of stereo inputs (3 analog and 2 digital with 192kHz D/A conversion), the Central Station allows you to switch between 3 different sets of studio monitor outputs while maintaining a purely passive signal path. The main audio path uses no amplifier stages including op amps, active IC's or chips. This eliminates coloration, noise and distortion, enabling you to hear your mixes more clearly and minimize ear fatigue. In addition, the Central Station features a complete studio communication solution with built-in condenser talkback microphone, MUTE, DIM, two separate headphone outputs plus a cue output to enhance the creative process. A fast-acting 30 segment LED is also supplied for flawless visual metering of levels both in dBu and dBfs mode. Communicate with the artist via talkback. Send a headphone mix to the artist while listening to the main mix in the control room and more. The Central Station brings all of your inputs and outputs together to work in harmony to enhance the creative process and ease mixing and music production.

Purified power.

To get the most out of your MOTU studio gear, you need the cleanest power possible. The negative effects of poorly supplied wall outlet AC power on your gear can be dramatic, without your ever knowing how good your gear can really sound with properly supplied power. Furman Sound introduces the all-new Power Factor Pro with its ground-breaking Clear Tone Technology™, which actually lowers the AC line impedance supplied by your wall outlet while storing energy for peak current demands — over 45 amps of instantaneous current reserve. Additionally, Linear Filtering Technology™ (LiFT) dramatically lowers AC line noise to unprecedented levels in the critical audio frequency band. Also included are Furman's unique Series Multi-Stage Protection Plus (SMP+) surge protection and automatic Extreme Voltage Shutdown (EVS), which protect you from damaging voltage spikes or sustained voltage overload.

Equipped with the same LiFT and SMP+ features, plus EVS Extreme, the Furman Sound IT-20 II ultra-low noise balanced isolation power conditioner is designed for the most critical, ultra-low noise installations. Delivering an astonishing 80dB of common noise reduction from 20Hz-20kHz, you're assured the lowest possible noise floor for all the gear in your MOTU studio. The IT-20 II's toroid transformer design assures a contained magnetic field for complete isolation from sensitive studio components nearby. The ultimate in purified power.

Unprecedented Native Studio Power

Hands-on automated mixing.

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The Mackie HR-Series Active Studio Monitors are considered some of the most loved and trusted nearfield studio monitors of all time, and with good reason. These award-winning bi-amplified monitors offer a performance that rivals monitors costing two or three times their price. Namely, a stereo field that's wide, deep and incredibly detailed. Low frequencies that are no more or less than what you've recorded. High and mid-range frequencies that are clean and articulated. Plus the sweetest of sweet spots. Whether it's the 5-inch HR-624, 8-inch HR-824 or dual 6-inch 626, there's an HR Series monitor that will tell you the truth, the whole truth, and nothing but the truth.

The MOTU system experts.

When it comes to putting together MOTU recording systems, nobody does it better than Sweetwater. Whether you're building a simple portable recording rig with a Traveler and an iBook or a 128-track powerhouse Digital Performer studio centered around the latest Quad Core G5, Sweetwater can help you select the perfect components for your MOTU system, from the specific MOTU audio interface model, to control surfaces and hard drives, to plug-ins and studio monitors. Even better, we can install, configure, test and ship a turnkey system straight to your door — all you'll need to do is plug in the system and start making music.

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Living in the Moment

By Larry the O

Living through a moment of historical significance is an amazing experience and not to be taken lightly. It might be something of global consequence, like the death of the pope, or something of national impact, such as Hurricane Katrina. In the music world, one can name numerous concerts or album releases that changed the course of musical events: the premiere of Igor Stravinsky's *Le Sacre du Printemps* and the release of *Sgt. Pepper's Lonely Hearts Club Band,* to name just two.

Sometimes, however, a moment in time is not a matter of hours, or even days, but a period of time that, viewed through the large lens of history, constitutes only a moment. I believe that the audio and music worlds are in just such a moment right now—in fact, closer to the end of it than to the beginning. I refer to the achievement of the all-digital studio.

Most people do nearly all of their audio work on a computer, but it was only some 20 years ago that I wrote an article for *Mix, Electronic Musician*'s sister publication, pointing out how the combination of three new breakthrough products—the E-mu Emulator II sampler, the Apple Macintosh computer, and Digidesign Sound Designer software—fulfilled many aspects of a digital audio workstation as defined by Stanford's CCRMA computer-music center. It was clear even then that there would come a time when every recording and electronic-music function would be executable in software.

That time is now.

Look at what we can do on our desktops without leaving our screens: record, edit every which way to Sunday, synthesize, play samples or models of acoustic and electronic musical instruments, EQ, compress, add realistic reverberation or create imaginary spaces, add phasing and flanging, and experiment with a jillion other types of sound warping and synthesis that only mathematicians could have conceived of a few short years ago. And, of course, you can mix and master. In short, you can create the whole shooting match on the computer.

The simple fact is that the all-digital studio is no longer a wish-list item.

True, acoustical events are still handled by microphones, loudspeakers, and acoustical treatment, but even those are being worked on: witness "digital" microphones (that is, mics with onboard ADCs) from Neumann, HHB, and other manufacturers.

This is all reality—not just for the shrinking number of fancy-pants high-end studios, but for all of us every single day. Because we experience this constantly, we are tempted to take it for granted. But even if you never knew any other way, it is important to recognize the significance of this point in history, and to realize that every time you boot up your DAW, load a software synth, and throw together a handful of loops, you are part of something powerful and special.

That is not to say we have achieved perfection. There's still lots to be said about computers' frequent misbehavior. Much work is needed on control surfaces and systems, and there's an audible distance between models and the real thing. We still must meet the challenge of injecting human interaction and spirit into a working method that is much more clinical than the old-school ways. I certainly have my concerns about things such as digital microphones' eliminating the variety and character added by mic preamps. But the simple fact is that the all-digital studio is no longer a wish-list item.

So where do we go from here? What is the next historical moment? Interestingly, that is where a converse point must be raised: the quiet developments whose real importance is never recognized, such as resolution of the limitations I've described and others. Reading EM, one quickly loses count of the times when interviewees and writers complain about mixing with the mouse and keeping their eyes glued to the screen, not to mention all the articles that show how to hot-rod your computer for audio (aka "cleaning out the gunk that slows things down"), troubleshoot problems, or work around incompatibilities. The impact of those issues is clearly huge, yet if the problems get worked out, it will never seem as momentous as the achievement of the all-digital studio.

The early part of the year is a good time for reflection. As you move through 2006, I suggest that you contemplate our world of music production in its historical perspective. EM

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