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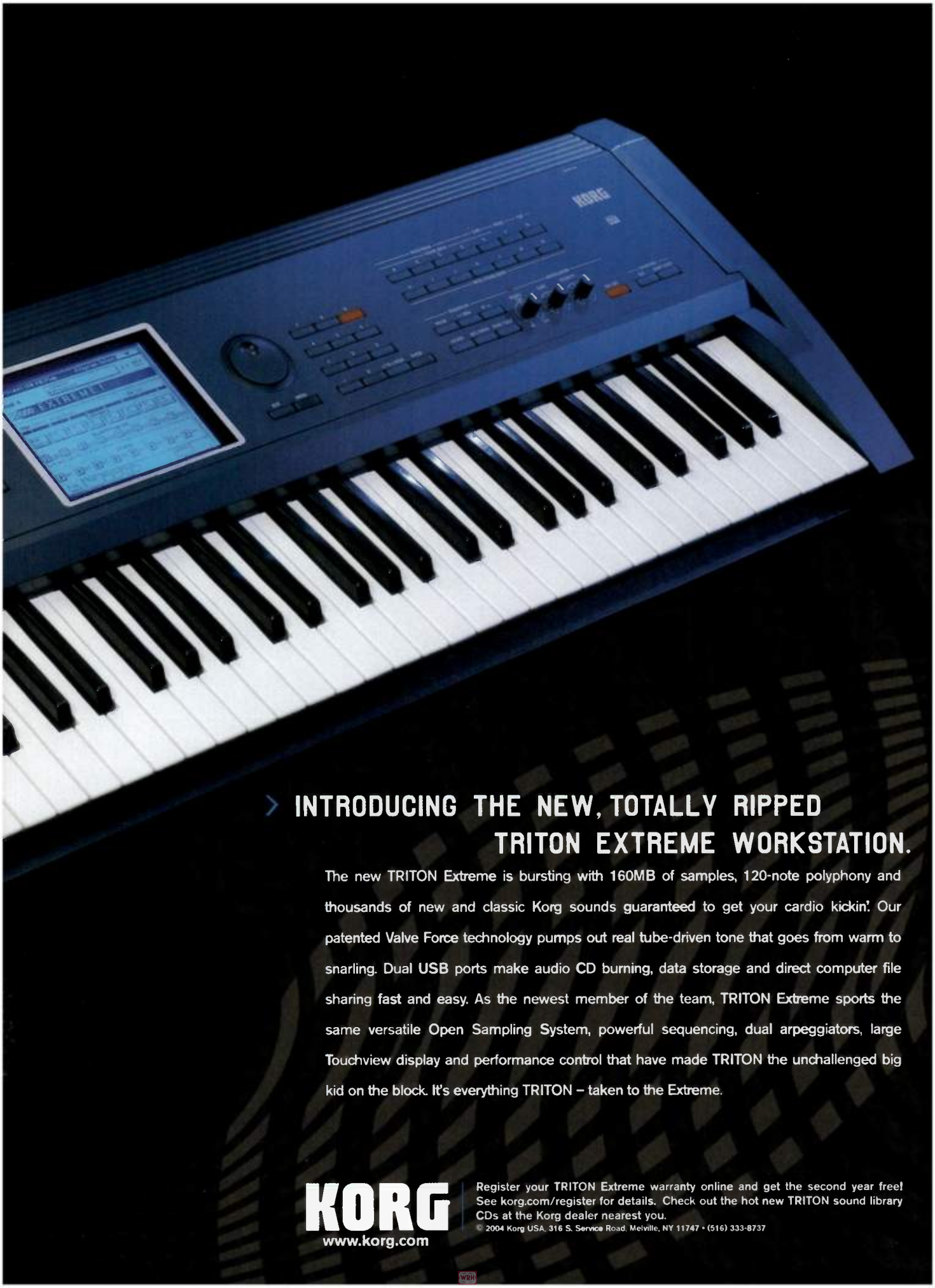

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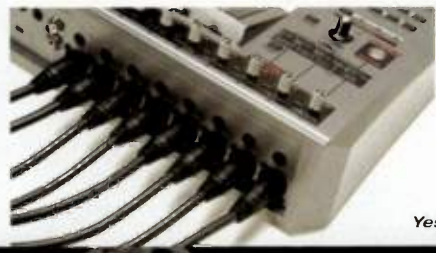

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Mackie's Onyx Development Team was comprised of veteran analog and digital engineers from the great Pacific Northwest (aka Woodinville, Wash). As a side note, this is the first time the elusive Viking Santa has been captured on film.

All Onyx mixers have a built-in Talkback section with onboard mic.

Engineering Director Chris Jubien puts an early Onyx prototype through its paces. Chris is from Canada, where he often tests audio gear in igloos.

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The Onyx 1220, 1620 and 1640 start with Mackie's new Onyx mic preamps, which meet or surpass any standalone mic preamp on the market in terms of pure fidelity and headroom (and yes, we have the specs to prove it). Next up is our completely new Perkins EQ circuitry, a "neo classic" 3- and 4-band design which gives you the sweet musicality of British EQ with greater filter control and minimum phase shift (plus a true hardware EQ bypass).

Two months to design a knob? Yep. Onyx knobs feature a more comfortable contoured shape with tapered top, smoother grip and a high-profile beak so you can almost mix blind.

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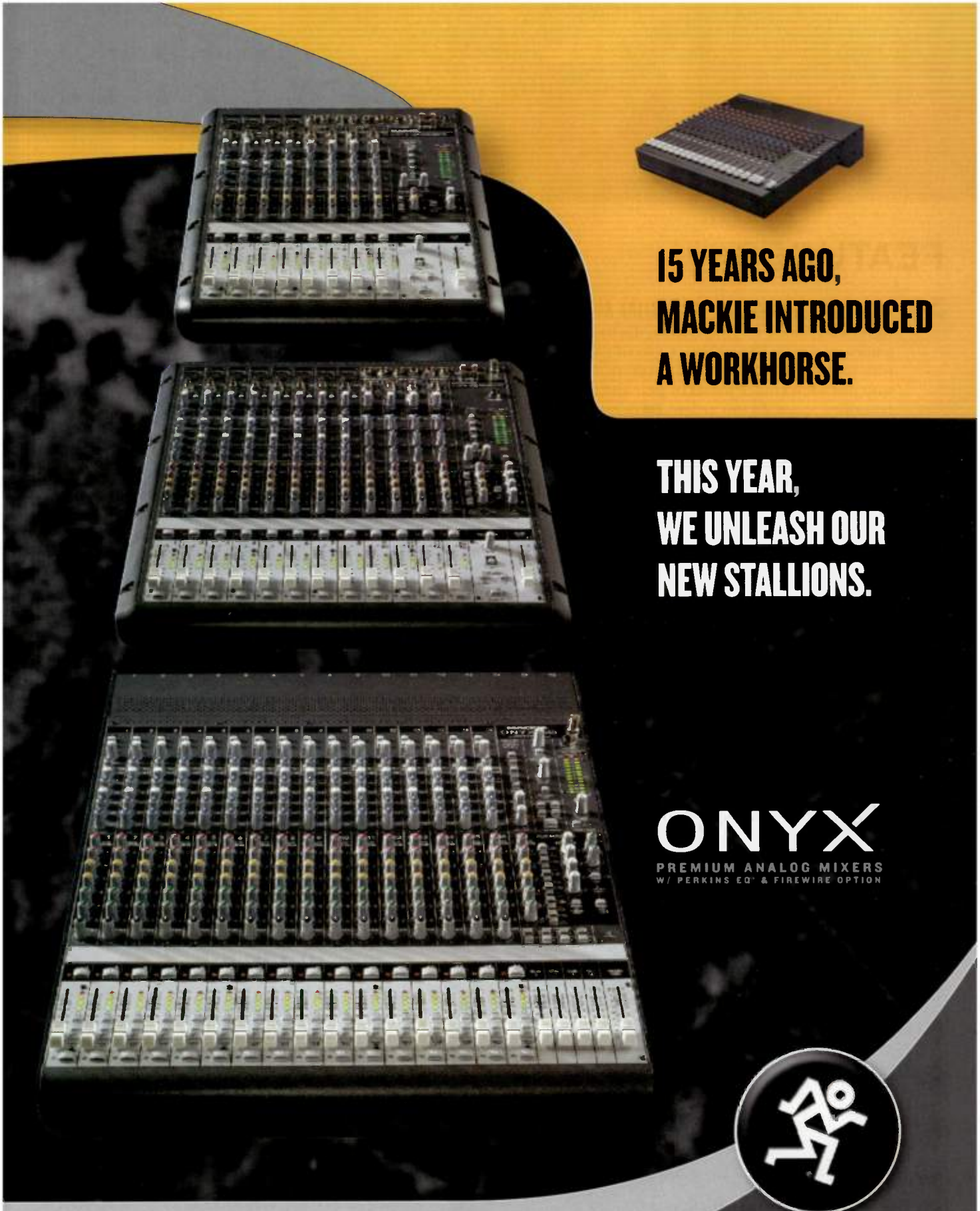
In addition to balanced direct outputs on every channel via dual DB-25 connectors (a first for any small-format analog mixer), you can outfit each Onyx mixer with an optional FireWire I/O card, which lets you stream either 12 or 16 independent channels of 24-bit/96k audio, plus a stereo mix, directly to your Mac or PC. You can also monitor a stereo mix via FireWire from your computer, making our Onyx Series mixers superb audio interfaces to your digital world... or any other world you plan to rule.



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I N S I

FEATURES

Cover photo by Dave Bishop

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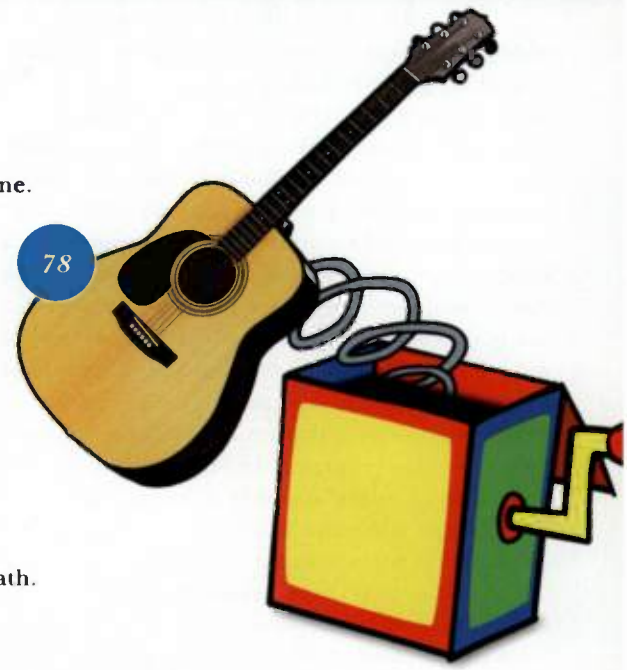
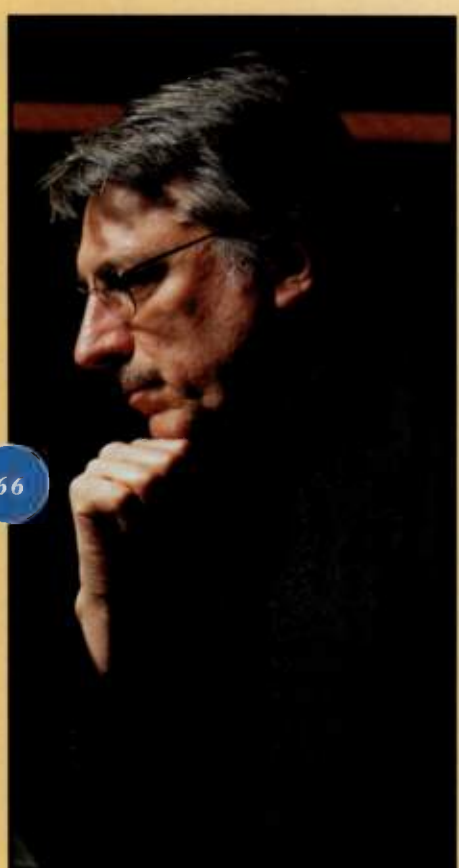
COLUMNS





REVIEWS

Guiding the Blind


DAN PHILLIPS

In many cultures, music making is a widely recognized form of employment for the blind. I don't imagine that blind people are any more musically talented than sighted people, but because their other options have traditionally been more limited, those who had musical talent have more often been encouraged. Today, the rise of computer technology and the personal recording studio appears to be a double-edged sword for blind musicians. On the one hand, computer technology and the Internet offer the possibility of increased independence. On the other hand, most music software relies on a graphical user interface, and if you can't see the screen, navigating a GUI can be a nightmare.

The good news is that on the Windows platform, there are at least two good screen-reading programs. Among other things, these screen readers use a speech synthesizer to "speak" onscreen text in any standard Windows application, providing blind people with much more access. At least some Windows music applications can use screen readers to make their products more accessible to blind users.

The bad news is that no comparable screen reader has been released for Mac OS X. And even where basic enabling technologies for the blind exist, few music-software developers are actively pursuing it. That could and should change.

Let's start with the positive. JAWS, from Dancing Dots (www.dancingdots.com), is a screen-reading program for Windows. That includes reading pop-up text labels triggered by a mouse rollover, for instance, to identify tools. Synapse Adaptive's Windows Eyes (www.synapseadaptive.com) is another screen reader for Windows.

If a program has a text-driven interface, a blind person equipped with a screen-reading program can hear the text and navigate the program. For example, Sony Pictures Digital Networks' Sound Forge offers an invisible text window that screen readers can "see" but sighted users can't. Keystroke equivalents let you navigate the program without mousing, a boon for blind folks as well as sighted users. As a result, blind musicians can get around Sound Forge without the developer having to alter the GUI.

Fortunately, JAWS's scripting language allows you to customize it to work with nonstandard and proprietary software, so if a music program has the right hooks (such as standard code identifiers for controls and windows), you can script JAWS to create a voice interface for it. David Pinto's CakeTalk for Sonar and Sibelius Speaking are sophisticated JAWS scripts that provide access to Cakewalk's digital audio sequencer and Sibelius's music-notation program, two applications that are particularly accessible to the blind. Both scripts are available from Dancing Dots.

So a few Windows developers have made a good start. Following their lead, more signal-processing plug-ins, soft instruments, looping tools, and all sorts of educational and entry-level music programs should also be made accessible to the blind. Apple should find or develop a capable screen reader for Mac OS X comparable to JAWS for Windows. Blind musicians have long been noteworthy members of the music scene; let's give them every chance to contribute.

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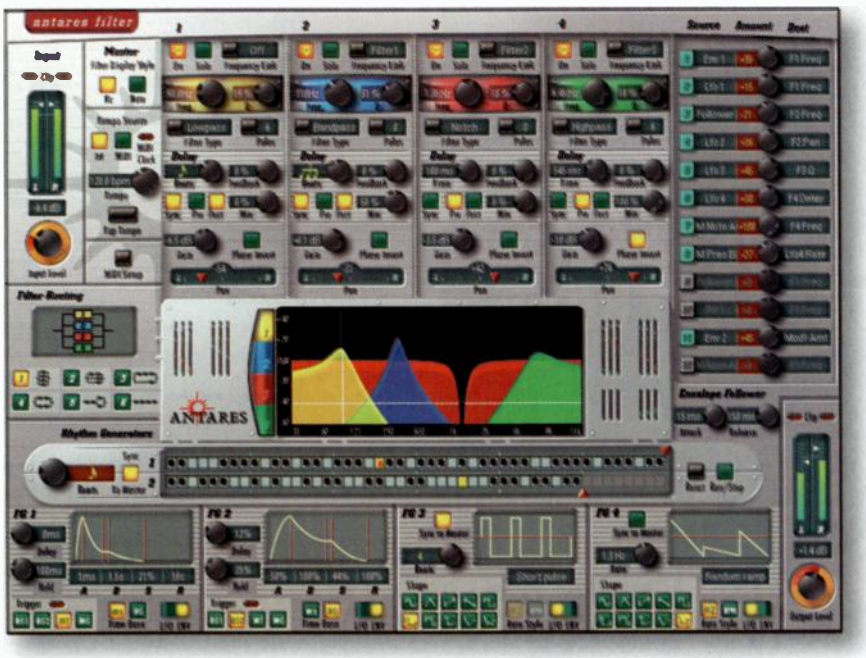

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- >> Envelope Follower
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- >> Powerful Modulation Matrix
- >> Extensive MIDI control
- >> Every time-based parameter syncable to internal master tempo or MIDI Clock
- >> Tracks (or entire mixes) that pulse with life and character

Think of it as the audio equivalent of a brilliant surgeon — who's a **really** good dancer.

Antares's new Filter plug-in gives you everything you need to shape your tracks with surgical precision or animate them with an almost limitless variety of tempo-synced rhythmic effects.

A DANCING STURGEON?

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At the core of Filter are four stereo multimode filters that provide all the warmth and smoothness you'd expect from classic analog filters. With low-pass, highpass, bandpass and notch modes, variable cutoff slope, four independent delay sections, and a variety of filter routing options, you'll have the power to create dynamic filter effects that range from extremely subtle to just plain extreme.

CONTROL FREQ

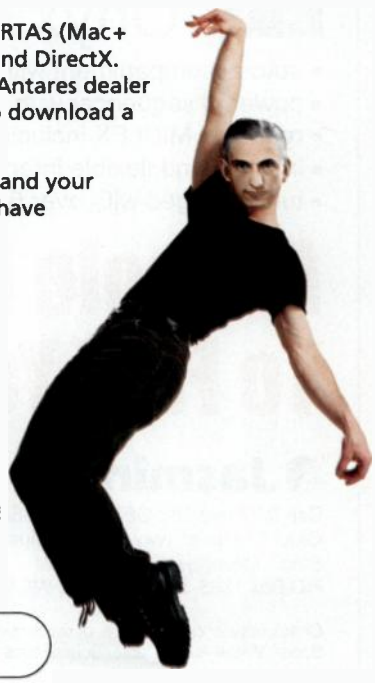
Since filters sound the coolest when they're whizzing around, we've given Filter a modulation section that would make many a full-blown synthesizer jealous. Pretty much everything that matters can be modulated. Control sources include four multi-shape LFOs, four Envelope Generators with delay and hold times, two Rhythm Generators, an envelope follower and a slew of MIDI controllers. All routed through a Mod Matrix with more than enough patches to get yourself into serious trouble.

FEETS, GIT MOVIN'

To really get you dancing, Filter includes two drum-machine style Rhythm Generators for loop-based grooves or complex polyrhythms. And to make sure everybody stays in step, every time-based parameter—from tempos, to delay times to envelope rates—can be locked to Filter's internal master tempo or your host's MIDI clock.

Filter will be available for RTAS (Mac+PC), VST (Mac+PC), MAS and DirectX. Check it out at your local Antares dealer or drop by our web site to download a fully functional demo.

Then get out your scalpel and your tap shoes and prepare to have some serious fun.



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PREACHING TO THE CHOIR

You hit the nail firmly on the head with your December 2003 "Final Mix" ("Pay You Back with Interest"), and it rings clear as a choir! As a musician, recordist, and producer for more than 20 years, I've experienced the spiritual payoff you speak of, as well as the compromises and letting go. Thanks for mixing it so well into such a resonant piece!

John A. Hadden
Resting Lion Studio
Huntington, Vermont

SPLITTING SIGNALS

The article "Creative Looping" in your December 2003 issue says on page 30 that real-time tape flanging was "developed at the Abbey Road Studio." Actually, the 1959 Toni Fisher hit "The Big Hurt" features this type of flanging (using two tape recorders simultaneously), and that was about five or six years before Abbey Road Studios.

James White
via e-mail

LOOPS, SCHMOOPS

I just finished reading Len Sasso's "Creative Looping" article in the December 2003 issue.

I make my own samples by adding real instruments or sounds. I'm a guitar player who would rather play an actual instrument. Don't get me wrong; I use Propellerhead Reason 2.5 and Sonic Foundry Acid 4.0, and I record on a PC-based DAW running Steinberg Cubase SX. I am by no means a technophobe, although I still prefer analog to digital.

In this digital age, what's being created is less music, and more noise. All the quantize and swing buttons in the world can't compare to a real drummer who has feeling and style. I mean no disrespect to Len Sasso, but I'm going to plug my Fender Telecaster into my Marshall amp and play some real music.

Steve Nemeth
Dual Impact Studios

RESTORATION ON A BUDGET

Although "Noise Busters" (July 2003) is an informative article on cleaning up audio, you should have considered mentioning GoldWave audio-editing software for Windows, as well. I don't have a lot of money, so I use Coyote Electronics' Groove Mechanic for quick cleanup; GoldWave for more detailed, manual cleanup; and Jens Fangmeier's Feurio for most of my CD burning.

Depending on the time one wants to spend on restoration, you can obtain very good results using "low end" software, as defined in your article.

Greg M.
via e-mail

WHAT'S THE FREQUENCY?

I restore LPs (et al.) on the side, and your review of shareware and freeware applications for multiple platforms ("Noise Busters," July 2003) is second to none. Keep up the good work!

I have two questions about RIAA frequency compensation on LPs. It seems the playback and recording response curves are nearly linear and inverse on recording and playback. First, are there any leftovers on the output from a turntable (through a receiver) that would need to be analyzed or removed? Second, if there are any leftovers, what can be done to use or remove them?

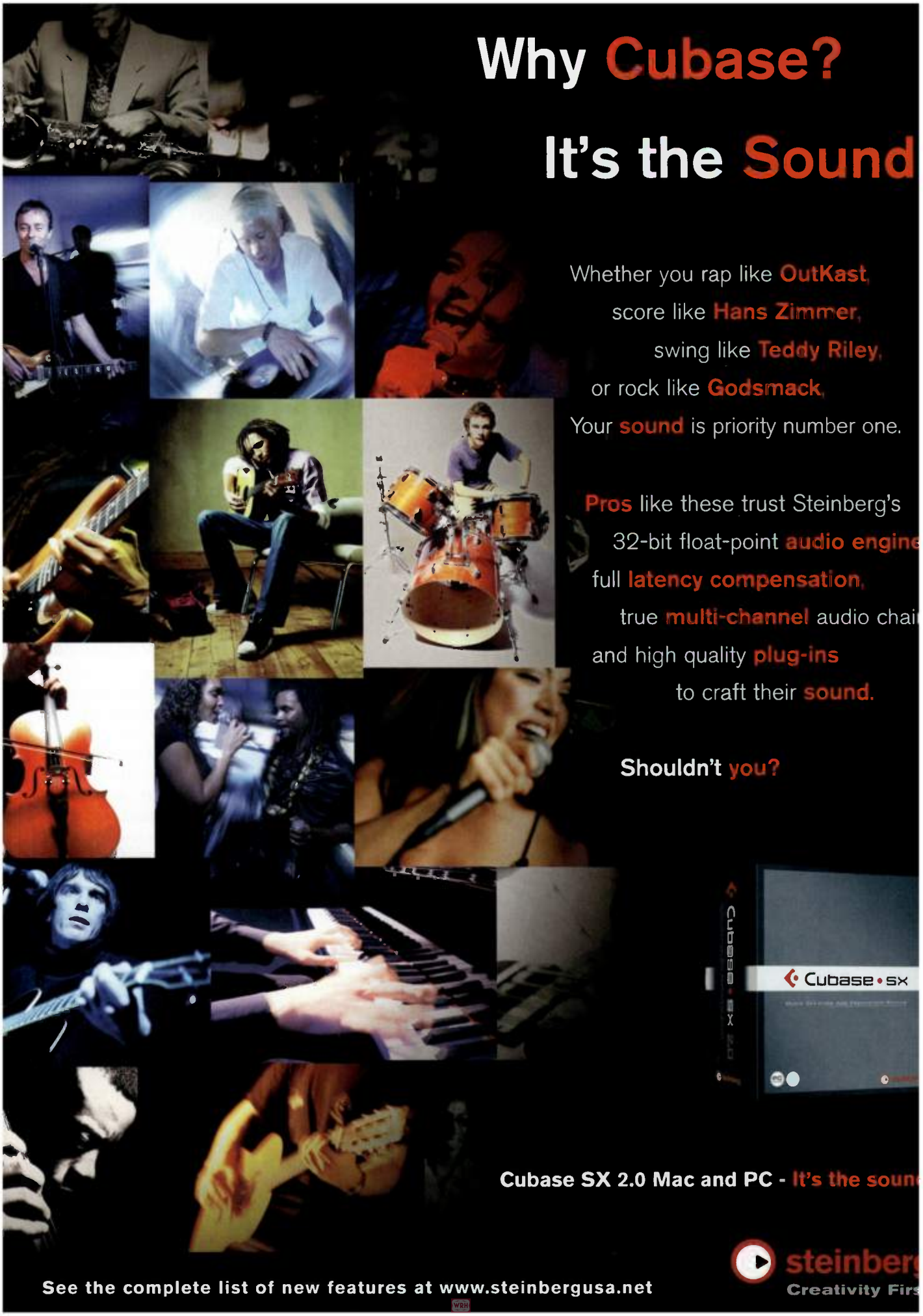
RIAA frequency compensation is discussed online at http://cylindersontheweb.tripod.com/sonic_restoration.htm.

Dave Lees
Fixsonic.com

Author Oliver Masciarotte replies: Dave—The RIAA's emphasis and de-emphasis characteristics are indeed inverse but far from linear in the frequency domain. Linear response would appear as a straight line on a frequency versus amplitude graph.

First, RIAA emphasis and de-emphasis can be applied either in the analog or digital domain. As with any emphasis/de-emphasis stage, it is included to preequalize the signal to better match or compensate for the characteristics of the transmission channel—in this case a phonograph record, which has poor low-frequency headroom and poor high-frequency noise characteristics. Before EQ was used largely for creative purposes, equalizers got their name from this ability to "equalize," or fix, a signal prior to transmission.

Most analog communications standards, including noise reduction, television, radio, and tape, employ emphasis on encode and de-emphasis on decode. Also, as with 15 and 30 ips tape, 33⅓ and 78 rpm records have different emphasis curves tailored to their individual characteristics. Back to your question: if the de-emphasis is performed in the analog domain, there can be a mismatch between the ideal filter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response of the specification and the real-world performance of an RC circuit that makes up the so-called compensation network. Usually, the mismatch is proportional to the cost of the electronics, because cheap, mass-produced parts are often way out of spec while hand-matched, fancy components can very closely match the desired EQ curve. In the digital domain, it's easier to "get it right," since there are fewer variables.

Because you usually don't have access to the original pristine signal, you have no way of knowing how much your result deviates from the original. But, with thoughtful choice of tools, you can do the least damage. Most mastering/restoration folks I know choose to transfer from the phonograph cartridge directly into an A/D, then perform the de-emphasis in the digital domain using a IIR filter built for that purpose.

For analog or digital de-emphasis, careful product designers pay attention to the filter's frequency and phase response to minimize distortion during processing. Better-quality DAWs, like my Sonic Studio HD system, have minimal-phase distortion, RIAA de-emphasis characteristics in their filter setting's menu.

LCD OR CRT?

I need some advice concerning my choice of monitor(s) for a Mac-based home studio. The two questions I'm wrestling with are:

1. LCD versus CRT. I like the lower cost of CRT monitors but am concerned they will contribute noise to my system.

2. What is the consensus concerning the use of two monitors? I know some who are quite happy with one large monitor, but I'd like some expert advice.

David Arnay
via e-mail

Former EM associate editor David Rubin replies: David—Ultimately, the choice between the traditional CRT and the newer LCD monitors must be based on your working style, workplace requirements, and budget.

As you point out, CRTs are often much less expensive than LCDs, when comparing similar viewing areas. But large CRTs (the kind needed for DAWs, sequencers, and other audio software) are enormously heavy. A dual-monitor CRT setup with a 20-inch displays adds some serious weight to your desktop and presents a sizable footprint.

LCDs are lighter, take up less desktop space, and generate much less heat. In my small studio, the heat factor alone was enough to make me switch to LCDs. They're also easier to move around and reposition, which helps when rearranging your system.

But LCDs have their own drawbacks. For starters, LCDs have a narrower viewing angle, both vertically and horizontally, than CRTs. Move too far outside of the sweet spot, and the color balance and contrast suffer appreciably. If you think you'll often be viewing the monitor from way off to the side, consider a CRT, which looks bright and clear from almost any angle. However, this gap has closed somewhat, and many LCDs now offer viewing angles as wide as 160 degrees.

CRTs still offer better contrast (which especially affects how they display shades of black) and better color purity than LCDs. LCD technology has improved significantly, though, so the differences are subtle. Furthermore, LCDs are much brighter than CRTs.

In addition, many people don't realize that LCDs have a "native" resolution (typically the highest resolution), and text and other onscreen characters may look unusually coarse or fuzzy when viewed at lower (even slightly lower) resolutions. I haven't found that to be a problem; I just run the display at the native resolution. Some people, however, might find the resolution issue to be troublesome, especially with certain fonts. If possible, try to view some text at your intended resolution before buying the monitor.

I've found that the newer CRTs are less likely to generate noise than the older ones, but I can't vouch for any specific make or model. CRTs could always pose a potential problem in a highly sensitive system because of magnetic interference, whereas LCDs don't present this problem. Fortunately, most monitor speakers are now shielded to eliminate magnetic interference, but if you are playing electric guitar close to a CRT, you could get some noise from the magnetic pickups.

In general, I highly recommend LCDs for studio use. A dual-LCD system with a digital-video feed is a space-saving, energy-efficient, low-heat, flicker-free, razor-sharp thing of beauty, and it offers onscreen real estate galore.

The choice between using one or two monitors is largely a matter of taste and budget. My dual-display setup gives me about as much viewing area as the largest of Apple's displays, but it cost me about $1,000 less. Of course, there is a separation where the monitor frames meet in the middle, but I haven't found that to be troubling.

What's more, my dual-monitor system is more flexible than a single display. I can split the monitors between two computers, and can have, say, one large Mac display or a split Mac/PC setup. Whichever monitor you shop for, carefully compare some samples of small text (not the color graphics typically shown in stores), and choose the sharpest display you can afford. Your eyes will thank you for it.

WE WELCOME YOUR FEEDBACK.

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- Integrated per-channel EQ and assignable FX controls
- Universal Bus Architecture
- Smoother audio engine with gapless effects patching
- ACID™ loop & MIDI Groove Clip support
- DX, DXi, ReWire, VST, VSTi support
- Integrate MIDI-compatible control surfaces
- Import/export OMFI & Broadcast WAV
- VSampler 3.0 DXi multi-format digital sampler
- Ultrafunk Sonitus:fx audio effects suite
- Lexicon Pantheon Reverb
- ASIO & WDM compatibility
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E-MU's Digital Audio Systems deliver everything you need to produce audio on your PC with professional results – 24-bit/192kHz converters, (the same A/D converters used in Digidesign's flagship ProTools™ HD 192 I/O interface), hardware-accelerated effects and mixing, and seamless compatibility with your favorite PC audio and sequencing software for less than $500. These Digital Audio Systems stand toe-to-toe with the most advanced and expensive audio systems in the world, and will change your expectations of desktop recording forever.

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- MTC Out

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- Turntable Input (w/ground lug and hardware RIAA preamp)
- 24-bit/192kHz ADAT In/Out (switchable to S/PDIF)
- 24-bit/96kHz coaxial S/PDIF In/Out (switchable to AES/EBU)
- 24-bit/96kHz optical S/PDIF Out (switchable to AES/EBU)
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*estimated street price

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Sync Daughter Card

Word Clock Sync In and Out

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Quick access to your most-used functions (e.g. Show/Hide, Session and effects management, etc.)

Viewer Window (3 Modes)

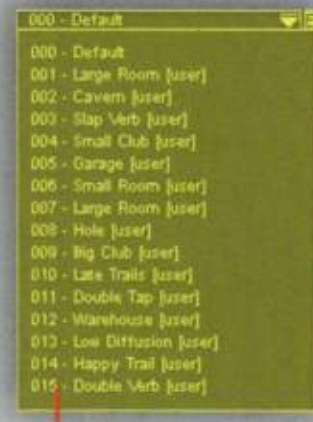
- Effects Editor lets you view all effects parameters for quick editing
- Input/Output Patchbays give you an instant overview of your I/O configurations at a glance

Audio Source

A Patchstrip can be fed by any external analog or digital input, as well as internal host sources from your audio application via ASIO™ or DirectSound®

Insert Chain

The Insert Chain allows you to easily route a channel of audio to any effects plug-in, physical output, or virtual output (ASIO) in any order (i.e. monitor your recording with effects but record dry). You can also send and return audio to/from your outboard gear like a traditional insert



Effects Presets

Save your favorite effects parameter settings as presets within the effects plug-in

Main & Auxiliary Busses

Offer you the same flexibility as the Insert Chain above to route your Aux Busses and Main Buss to effects plug-ins, physical outputs and virtual outputs (ASIO)

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Choose from Internal Clock, External ADAT, or External S/PDIF sync to synchronize your entire digital studio

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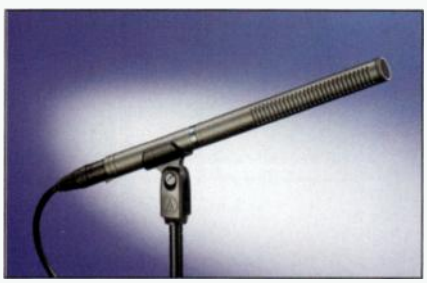
WHAT'S NEW

By Geary Yelton

AUDIO-TECHNICA AT897

A shotgun microphone is to the audio recordist what a telephoto lens is to the photographer. If you record dialog for video, provide sound reinforcement for live theater, or make field recordings in the wild, you probably know how hard it is to find an inexpensive shotgun mic. To meet that need, Audio-Technica has introduced the AT897 ($369), a 5.1-ounce, 11-inch-long shotgun mic that offers the advantages of larger models at a reasonable cost.

The manufacturer says the AT897's condenser element has excellent on-axis response, and its Line + Gradient polar pattern offers off-axis rejection from the sides and rear of the mic. A low-frequency rolloff switch is also included. The AT897 uses either one AA battery or 11 to 52V phantom power. It ships with a foam windscreen, a stand adapter, and a soft carrying case. Audio-Technica U.S., Inc.; tel. (330) 686-2600; e-mail pro@atus.com; Web www.audio-technica.com.



E-MU EMULATOR X

E-mu Systems has announced Emulator X (Win, $299.99) and Emulator X Studio (Win, $599.99) Desktop Sampling Systems, which combine software, a PCI card, and an I/O breakout box. E-mu claims that these new computer-based samplers offer the audio quality of its freestanding samplers, but with the advantages of Windows' user interface and file-management capabilities. Emulator X and X Studio support disk streaming and provide WDM, DirectSound, and ASIO drivers for Windows 2000 and XP.

E-mu's waveform editor provides 36 virtual patch cords per voice, more than 50 Z-Plane morphing filters, and DSP tools such as time compression and expansion. Automated sampling capabilities let you quickly turn user samples and WAV files into a ready-to-play preset. Emulator X and X Studio support a variety of sound formats, including EIII, EOS, Akai, and GigaSampler. Both systems feature a PCI-based acceleration processor that, according to the manufacturer, delivers zero-latency mixing and monitoring and at least 16 simultaneous effects without straining your computer's CPU resources.

Emulator X provides balanced ¼-inch stereo analog I/O as well as coaxial S/PDIF, ADAT, FireWire, and MIDI I/O. Emulator X Studio features the same 24-bit, 192 kHz converters used in Digidesign's Pro Tools|HD interface. It also adds a pair of TFPro 48V phantom-powered preamps with Neutrik connectors that accept mic, line, and high-impedance inputs; a daughter card with word-clock I/O, SMPTE I/O, and MTC Out; a turntable input with a ground lug and an RIAA preamp; additional MIDI I/O; and eight dedicated speaker outputs. E-mu Systems; tel. (831) 438-1921; e-mail info@emu.com; Web www.emu.com.

MCDSP CHROME TONE

The Chrome Tone (Mac, $495) multieffects processor provides chorus, delay, distortion, EQ, flanging, gating, reverb, tremolo, and wah in a single TDM plug-in. It can dynamically respond to audio and MIDI signals by adjusting the depth, speed, and shape of a syncable LFO and other modulators. McDSP says Chrome Tone makes efficient use of DSP resources to provide multiple simultaneous effects.

Chrome Tone's amp section models a guitar amplifier, offering five distortion modes with compression and sustain, a noise gate, low-cut filters, and shelving and parametric EQ. It also models guitar cabinets and simulates spring reverb. In the modulation-effects section, you can control chorus, tremolo, and wah with an audio input, sidechain, or MIDI data. The wah effect has three bandpass, three highpass, and three lowpass filter configurations, as well as modes that combine wah with phase-shifting. The chorus effect offers two types of chorus, flange, and tape delay simulation. Stack mode combines the amp, tremolo, wah, and chorus effects. Chrome Tone also includes individual effects plug-ins for the amp, chorus, tremolo, and wah with phaser effects.

Chrome Tone requires TDM hardware and is compatible with Pro Tools|24 Mix, Pro Tools|HD, and Pro Tools|HD Accel systems running in Mac OS 9 or Mac OS X 10.3. An RTAS version is expected soon. McDowell Signal Processing (McDSP); tel. (650) 318-0005; e-mail info@mcdsp.com; Web www.mcdsp.com.

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Differentiating itself from standard computer I/O boxes which are typically based on a patch-bay paradigm, the Omega 8x4x2 USB I/O mixer is based on a mixer paradigm and includes input, output and mixing functions that support a variety of tracking/monitoring applications while requiring no additional mixing hardware. The I/O mixer is packed with professional features such as ultra-transparent, high resolution A/D converters, extremely low-noise mic preamps with 48-volt phantom power and active balanced line level inputs. MIDI and S/PDIF ports allow connection to a variety of digital equipment.

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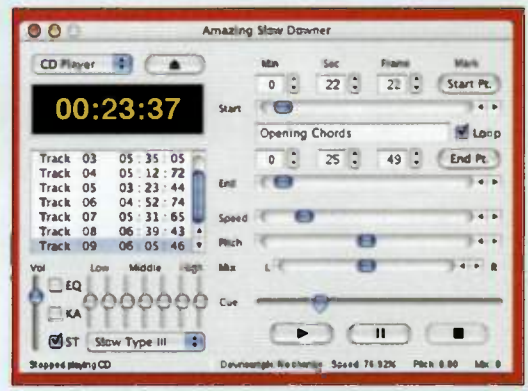
DOWNLOAD OF THE MONTH▲▲▲▲

▶ AMAZING SLOW DOWNER 2.6.3

Amazing Slow Downer (Mac/Win, $39.95) is a standalone application for high-quality, independent manipulation of pitch and playback speed of audio files and CD tracks. Its primary purpose, as its name suggests, is to slow audio playback so you can pick out a musical part and learn it. The author, Rolf Nilsson, is a jazz musician and teacher; his original motivation for Amazing Slow Downer was to provide a practice and transcription aid, but postprocessing save-to-disk options make it a useful DSP tool too. Amazing Slow Downer supports WAV, AIFF, MP3, AAC, and MP4 audio files, and it can convert MP3 files to AIFF format with or without downsampling and bit reduction.

Amazing Slow Downer works with audio CDs and files on your hard drive. Consequently, when you're trying to transcribe a few bars of a song from a CD, you don't need to convert the track to an audio file first. It actually slows down and speeds up CD playback in real time.

Amazing Slow Downer's controls are all on a single screen. On the left side, you select a track or file and view the current playback position in minutes, seconds, and frames (1/75 of a second). At bottom left are controls that enable preprocessing EQ and Karaoke mode (which suppresses material common to both stereo channels), determine whether playback is stereo or mono, and offer a selection of Slow Down algorithms. Loop, speed, and pitch parameter controls are on the right.



You can loop any portion of an audio file or CD track and adjust its pitch in semitones or cents over a two-octave range. Speed can be viewed as Stretch amount (ranging from –50 to 400 percent) or as a multiple of the original speed (ranging from 20 to 200 percent). A Mix control sets the output's stereo balance, which is handy both for homing in on a single instrument and for adjusting the effect of the Karaoke function. You can save all the aforementioned parameters in up to ten presets for each audio file.

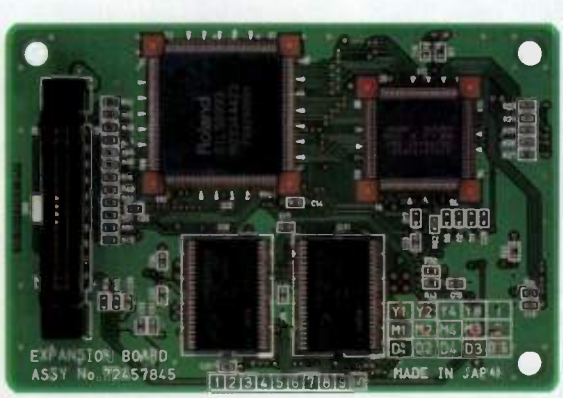
Amazing Slow Downer will import CD tracks and save audio files, with or without processing, to your hard drive. If looping is active, you can choose how many loops to include in the file. The program's import capability is handy for saving clips with altered speed or pitch and for extracting segments of audio CD tracks—for example, from audio sampling CDs.

Whether you do a lot of audio transcription or simply want a convenient utility for pitch and speed change, Amazing Slow Downer is worth a look. You can download a playback-limited demo of Amazing Slow Downer from Roni Music's Web site (www.ronimusic.com).

—Len Sasso

▼ ROLAND VS8F-3

Roland recently released the VS8F-3 ($395), an expansion board that hosts plug-ins for its V-Studio digital audio workstation (DAW) platform. What makes the VS8F-3 especially significant is that it accepts plug-ins written for the V-Studio by third-party developers. Roland DAW users will gain many of the effects-processing capabilities enjoyed by computer-based DAW users, while still retaining the advantages of a portable, self-contained recording system.

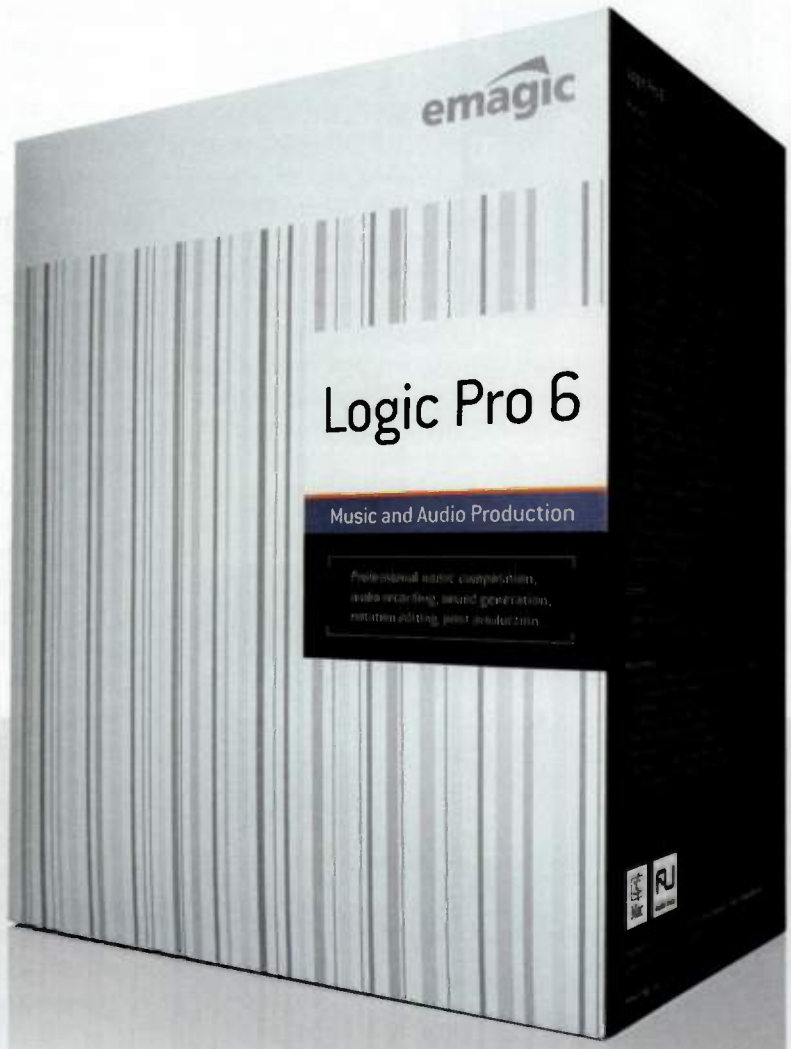


The VS8F-3 offers up to 56-bit processing and sampling rates as high as 96 kHz, depending on which V-Studio is the host. Each expansion board runs two plug-ins, and five Roland plug-ins are included: Mastering Took Kit, Pre-Amp Modeler, Stereo Reverb, Tempo Mapping Effect, and Vocal Channel Strip. At least eight developers have plug-ins available for the VS8F-3, including Antares Auto-Tune, IK Multimedia T-Racks, Massenburg DesignWorks High-Resolution Parametric EQ, McDSP Chrome Tone Amp, and SoundToys SoundBlender. Cakewalk has ported its SoundStage and fx:reverb plug-ins to the Roland platform, and TC Electronic is offering its TCR 3000 reverb. Universal Audio's software LA-2A Leveling Amplifier and 1176LN Limiting Amplifier emulations are also VS8F-3-ready. Additional plug-ins will soon follow these initial offerings.

The VS8F-3 initially supports the VS-2480 series, VS-2400CD, and VS-2000CD. Roland plans support for the VS-1680, VS-1880, and VS-1824 models in the second quarter of 2004. Roland Corporation U.S.; tel. (323) 890-3701; Web www.rolandus.com.

emagic

Completely Amazing

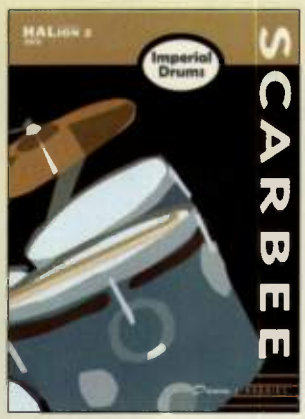


Introducing the new Logic Pro 6, the most flexible, most advanced and best equipped music production software for the Mac. At an impressively low price you get all the highly praised professional software instruments and plug-ins from Emagic. Also included is the renowned EXS24 mkII software sampler. Plus you'll get incredibly realistic sounding vintage instruments such as the EVP88 electric piano, EVB3 organ and EVD6 clavinet, and a range of software synthesizers that provide amazing options for creative sound design. Even the new convolution reverb – Space Designer – is included. While Logic Pro 6 includes an outstanding array of impressive capabilities and options, if you're just looking to get started, check out Logic

www.emagic.de

SOUND ADVICE

SCARBEE



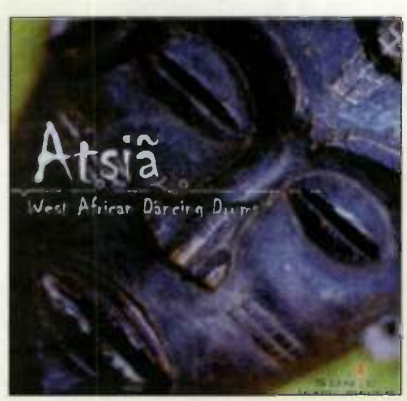
The newest sample library from Scarbee is *Scarbee Imperial Drums*, or S.I.D. ($399), a 12 GB collection of drum samples on three DVDs for the HALion 2 format. Nearly 30,000 individual samples were recorded in 24-bit, 88 kHz audio and downsampled to 44.1 kHz for release. The entire library focuses on a single DW drum set with Paiste cymbals, along with additional snares and a second DW kick drum, providing a sampled kit with an unprecedented level of detail, according to its developer.

For realism and versatility, each drum was recorded from several mic positions, with and without bleed. A single S.I.D. kit features over 420 kick-drum samples in 75 Velocity layers and over 3,000 samples dedicated to one 14-inch hi-hat. You can access more than 2,200 samples of a single snare drum in over 130 Velocity layers by using a combination of one key and the mod wheel, preventing the usual repetition of a single hit that betrays sampled drum parts. S.I.D. also supplies streamlined kits for users with modest computer resources. Scarbee/EastWest (distributor); tel. (800) 969-9449 or (718) 932-6328; e-mail info@soundsonline.com; Web www.scarbee.com.

SONIC IMPLANTS

If your music could use some African percussion, Sonic Implants has a sample disc for you: *Atsiã: West African Dancing Drums* ($229.95). *Atsiã,* named for one of the oldest dance styles in West Africa, contains over 400 MB of sampled instruments traditional to Ghana's Ewe tribe. The collection provides handmade drums with names like *apentema* and *sogo,* as well as the more familiar talking drums, log drums, and *surdu.* Also included are bells and shakers such as *axatse* and *toke* bells. Most of the instruments have never been available in a sample library before.

In addition to individual hits, *Atsiã* features loops of individual instruments in 4/4 and 6/8 time. All sounds were multisampled in stereo with up to four Velocity levels for each type of hit. Atsiã is available in GigaSampler, Kontakt, and SoundFont formats. Sonic Network/Sonic Implants; tel. (888) 769-3788 or (617) 718-0202; e-mail info@sonicimplants.com; Web www.sonicimplants.com.



SONY PICTURES DIGITAL

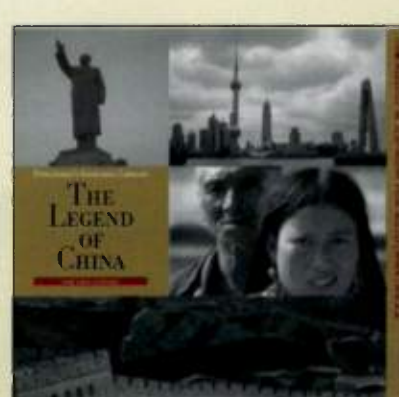
As part of its Loops for Acid series, Sony is shipping *Numina II* ($59.95), produced by ambient recording artist Jesse Sola. *Numina II* combines atmospheric textures, electro-acoustic percussion, synth arpeggios, and other cinematic timbres in a collection of loops and audio events designed to enhance film, video, and audio recording projects.

Sounds are divided into four folders of Acidized WAV files labeled Ambience, Drones & Themes, Percussion, and SFX & One-Shots. Most are clearly the products of skillful synth programming, effective processing, and clever sound design. Some sounds are simple, others complex, and all are quite fresh, with varying degrees of tension ranging from tranquil to near panic-stricken. Sony Pictures Digital; tel. (800) 577-6642 or (608) 204-7680; Web http://mediasoftware.sonypictures.com.

DISCOVERY FIRM

Discovery Firm has released *The Legend of China: The New Edition* ($55), a revised version of its acclaimed sample library. The original came on two audio discs (see our October 2000 review, online at www.emusician.com); the new set is on one audio CD with 339 recordings and one CD-ROM with 387 Acidized WAV and REX2 files, all recorded in China. In addition to multisampled percussion, strings, and wind instruments playing musical phrases and individual notes, the collection provides vocal phrases and words and various sound effects such as Tibetan prayers and subway ambience.

In each Acidized WAV folder are two folders labeled Loops and One-Shot Phases & Samples. Additional WAV files are archived in a zipped Bonus Tracks file. Instruments range from *yang qin* to *bian zhong*; vocal samples are courtesy of Peking Opera performers. Discovery Firm; e-mail overseas@discoveryfirm.com; Web www.discoverysound.com.



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STARR LABS ZTAR D3X

Starr Labs has introduced its new flagship MIDI guitar controller, the Ztar D3X ($5,995), an instrument that incorporates virtually every option the company offers. Although it has no actual strings, the D3X features 6 String Triggers, 6 Key Triggers, 12 expression pads, a ribbon controller, a breath controller, and a palm-mounted joystick. In addition, the D3X has a NeckSensor pressure strip mounted on the fingerboard, an embedded Wavetable Sound Card with a headphone output, a USB port, and ¼-inch inputs for the included sustain and volume pedals.

An onboard computer controls the D3X, providing features that other controllers don't offer. Its software is capable of an unlimited number of tunings and can split the fingerboard into zones. A looping sequencer is also included.

To simplify fretting, Multi-Harp mode lets you assign a chord to each key on the fingerboard as you pick normally with your right hand. You can play different chords at the same time and hammer-on from one chord to another.

The D3X has a diamond-shaped body and neotribal graphics on a black-sparkle finish. The V3X, a Ztar with a Strat-style body and the same feature set as the D3X, is also available. A hard-shell case is included with both models. Starr Labs; tel. (619) 233-6715; e-mail harvey@starrlabs.com; Web www.starrlabs.com.

PSPAUDIOWARE PLUG-INS

Three new downloadable plug-ins for Macintosh and Windows are available from PSPaudioware: PSP EasyVerb ($69), PSP MasterQ ($149), and PSP Nitro ($149). PSP EasyVerb provides nine reverb algorithms that simulate physical spaces (such as arenas and cathedrals) and studio processors (such as plates and spring reverbs) at sampling rates as high as 96 kHz.

PSP MasterQ is a parametric equalizer for mixing or mastering that consists of 12- or 24 dB-per-octave low- and high-cut filters; low- and high-shelf filters; and low-mid, mid, and high-mid peaking filters—each with adjustable frequency and Q. PSP MasterQ displays each filter's individual curve and a high-resolution graph of the overall EQ curve. It supports sampling rates from 44.1 to 192 kHz.

PSP Nitro is a multimode filter plug-in that emulates several types of analog filters and other effects, including phaser, bit-crusher, waveshaper, and delay. Virtual cables connect internal modulation sources (including two syncable LFOs and an envelope detector) and route MIDI signals to control processing parameters.

All three plug-ins include a library of presets and currently support VST and DirectX for Windows and VST for Mac OS X. (Audio Units and RTAS versions should be available around press time.) System requirements vary with each plug-in, but they all need a minimum Pentium III with Windows 98 or a Mac G4 with OS X 10.1. PSPaudioware.com s.c.; tel. 48-60-196-3173; e-mail contact@pspaudioware.com; Web www.pspaudioware.com.

DASH SIGNATURE EVE

The newest plug-in from Dash Signature, developer of VST instruments for Windows, is Electronic Vintage Ensemble, or EVE (Win, $99.95). EVE features a playback engine for a multisampled library of classic keyboard sounds and electronic percussion, including the Yamaha CP70B, Hohner D6, Solina 1, and Vox Continental C. It combines those samples with sawtooth and square waves from a Korg MS20 and a Minimoog to produce what its creator calls Semi Additive Synthesis (SAS).

EVE is 3-part multitimbral, and each part has a lowpass filter, three LFOs, 3-band EQ, and an ADSR envelope you can assign to control filter cutoff, amplitude, and pitch in any combination. Each part also has send and return controls for EVE's two effects processors, Modulator (phaser, chorus, flanger, and rotary) and Echo (a tape-echo simulation). A MIDI Learn function lets you use a MIDI controller or sequencer automation to control more than 100 parameters in real time.

When you buy EVE, you receive two versions: one version with a stereo pair of outputs and another with five stereo pairs. A free upgrade supplies additional waveforms and brings the number of factory presets to 256. EVE requires at least a Pentium III/700 MHz with 132 MB of RAM running Windows 98SE, ME, NT, or XP and a VST-compatible host. EVE has increased polyphony and performance on computers with SSE-enabled processors. You can download a demo from the company's Web site. Dash Signature; e-mail info@dashsignature.com; Web www.dashsignature.com.

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KEY CHANGES

After 2004, Summer **NAMM** will no longer take place in Nashville, Tennessee, because it has outgrown Music City's convention space. For the foreseeable future, it will alternate between Indianapolis, Indiana (beginning in 2005), and Austin, Texas (beginning in 2006) . . . **Audio Ease** (www.audioease.com) is shipping VST Wrapper 4, which enables Digital Performer 4 users to take advantage of VST plug-ins for Mac OS X . . . **PSPaudioware** (www.pspaudioware.com) is shipping MixPack 1.7, an update to its bundle of four audio-enhancement plug-ins. It is now compatible with DirectX and VST for Windows, MAS and VST for Mac OS 9, and VST for Mac OS X. Audio Units and RTAS versions for OS X will be shipping soon . . . **Dave Smith Instruments** (www.davesmithinstruments.com) has updated the Evolver to version 2.0. A free download adds a new bank of 128 programs and provides additional modulation destinations, longer envelopes, tap tempo, and other features . . . **Wave Arts** (www.wavearts.com) has released version 4 of its Power Couple bundle (which includes TrackPlug and MasterPlug), featuring a new user interface, a cross-platform Preset Manager, Undo and A/B buffers, and Mac OS X 10.3–compatibility . . . **Native Instruments** (www.nativeinstruments.de) has finished updating its virtual instruments for Mac OS X compatibility, supporting VST, RTAS, and Audio Units. Recent updates include Kontakt, Kompakt, Pro-53, B4, and Spektral Delay . . . **McDSP** (www.mcdsp.com) now offers Mac OS X 10.3– and Pro Tools|HD Accel–compatible versions of its entire product line, including Analog Channel, CompressorBank, Chrome Tone, FilterBank, MC2000, and Synthesizer One.

APOGEE MINI-MP

Apogee's most recent addition to its Mini series is the Mini-MP ($995), a compact 2-channel mic and instrument preamp designed for studio and mobile recording. It features two low-distortion preamps that are capable of 75 dB of gain. You can adjust levels with a single knob for each channel, eliminating the need for input-gain switches. A stereo master output-gain control makes the Mini-MP a good match for any audio equipment, balanced or unbalanced.

The Mini-MP seems to offer plenty of headroom, with mic inputs that handle levels as high as +18 dBu. The Neutrik combo jacks also accept automatically selected high-impedance instrument and line-level inputs from a variety of sources. The electronically balanced line drivers simulate transformer behavior to provide perfect symmetry in balanced mode and no increased distortion in unbalanced mode. A switch enables a Mid-Side (M-S) matrix decoder, and the Mini-MP can operate as two independent preamps. Other features include 48V phantom power, a phase switch, and an 80 Hz highpass filter with 18 dB-per-octave rolloff.

To maintain low noise and distortion, gas-filled relays are used for all switching, and capacitors in the audio chain are used only to separate phantom power from the active circuitry. The Mini-MP accepts power sources from 6 to 14V for optimum versatility. Apogee Electronics Corp.; tel. (310) 915-1000; e-mail info@apogeedigital.com; Web www.apogeedigital.com.

APPLE GARAGEBAND AND JAM PACK

Apple Computer is shipping an updated version of its creative software suite, iLife '04 (Mac, $49). In addition to bundling new versions of iPhoto, iMovie, and iDVD with the latest iTunes, iLife '04 features a new music-production application called GarageBand. iLife's other components have made creating and managing various types of media simpler and more affordable; GarageBand aims to do the same with music production.

GarageBand is a multitrack digital audio and MIDI sequencer that comes with more than 50 software instruments, over 1,000 Apple Loops, and 16 effects processors with more than 200 presets, including half a dozen modeled guitar amps. The loop library provides backing tracks, and a graphic database helps keep the loops organized. You can mix tracks to stereo and export your mixdowns to iTunes so you can burn them to CD, encode them as MP3 or AAC files, or transfer them to an iPod.

To extend GarageBand's capabilities, Apple also offers an optional add-on package called Jam Pack ($99). Jam Pack adds more than 2,000 additional loops, 100 sampled instruments, 100 effects presets, and 15 additional amp presets.

iLife '04 is bundled with all new Apple computers. To use GarageBand and its software instruments, you'll need at least a Mac G4/600 MHz with 256 MB of RAM running Mac OS X 10.2.6 or later. Apple Computer; tel. (408) 996-1010; Web www.apple.com. ●

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By Scott Wilkinson

Getting Found

A new program helps consumers find your music online.

Music technology has progressed so much that anyone with talent and time can record and produce music that rivals almost anything done in a commercial studio. But once you have that final mix in your computer, how do you get people to buy it? There are several Web sites, such as CD Baby (www.cdbaby.com), that allow independent artists to sell their music online, but it isn't so easy for consumers to find your stuff among the myriad offerings.

A company called Predixis (www.predixis.com) has developed an interesting answer to those concerns and incorporated it into their new digital-music management software, MusicMagic Mixer. The program is now available for Windows, with Mac OS X and Linux versions in beta testing.

MusicMagic Mixer is based on algorithms that analyze the audio information in a digital-music file and generate values for 35 parameters that correspond to the musical content. This helps the program find other files with similar content by comparing parameter values. For example, the tempo parameter reflects the overall sense of speed, not just beats per minute. The tempo algorithm looks for peak events and repeated patterns within certain frequency spectra, while filtering out patterns that don't correspond to overall speed. Other parameters include how "noisy" the piece is (distortion vs. clear harmonics) and overall volume level. The time required to analyze a piece in this way is about 80 percent of the file's real-time duration.

Predixis maintains a server with a database of recorded music and the parameter values for each piece. As of this writing, it includes about half a million titles. The database interacts with the MusicMagic Mixer client software, which users can download from the company's Web site.

After the software is installed on your computer, you point it to your music collection, and it scans the files for metadata, such as title and artist. It also performs a quick analysis of the first 30 or 40 seconds of the file to create a "fingerprint" that serves as a backup in case the metadata is inaccurate (which is all too often the case). Currently supported file types include MP3, WMA, OGG, and FLAC. (The last two are open-source compression formats; OGG is much like MP3, and FLAC is a lossless format.)

Next, the program contacts the Predixis server and compares the metadata and fingerprints of your collection with its database. If a piece you have is already in the database, its parameter values are uploaded to your computer. If not, the software performs an analysis of the unknown selection and sends the resulting parameter values to the server, which adds them to the database.

Once it has the parameter values for your collection, MusicMagic Mixer uses them to automatically generate playlists of selections that are similar in genre, mood, and so on. Then the software sends the playlist to one of several supported software players, such as Musicmatch Jukebox, Windows Media Player, or AOL Winamp. You can also download it to a hardware MP3 player or play it through the Turtle Beach Audiotron, which sends the audio through a home network (Ethernet or phone line) to any networked audio system.

COURTESY PREDIXIS

Predixis's MusicMagic Mixer automatically generates playlists of digital-music files and lets users look for similar music on CD Baby and other sites by pressing the CD Ideas button.

How does all this lead consumers to your music on a site like CD Baby? When users press the onscreen CD Ideas button, the software looks for music on the site with parameter values that are similar to whatever they've selected from their library; if your music has similar values, they'll find it right away. Currently, Predixis has agreements with CD Baby and Amazon.com, and they're planning to develop relationships with many music e-tailers. If this catches on, anyone who likes the sort of music you record will be able to find you easily, and nothing is more valuable than that to an independent artist.

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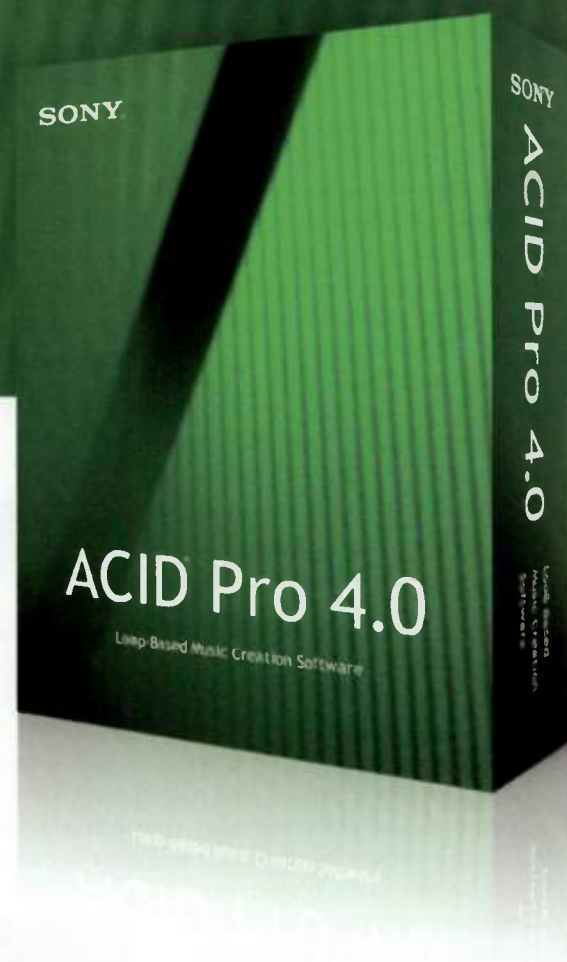

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By Matt Gallagher

Band on the Run

Kinky captures inspired moments on the road.

Hailing from Monterrey, Mexico, Kinky (Gil Cerezo, Carlos Chairez, Omar Gongora, Ulises Lozano, and Cesar Pliego) enjoyed breakout success in 2002. The band toured behind its self-titled debut album, giving an energetic live show that won fans around the world. Kinky's music draws on global influences. "We love techno, house, big beat, pop, acid jazz, Latin jazz, and Latin American folk styles," says Lozano, the band's keyboardist. "We're always writing, and we take the time to record."

Kinky recorded its debut in Monterrey in the band's home studio, which includes a Yamaha 01V digital mixer, a Mac G4/800 MHz running Digidesign Pro Tools 5.1, and a Digidesign Mbox audio interface. "It's a good room that has space for keyboards, guitar amps, drums, and percussion," Lozano says. Lozano plays vintage instruments, including a Fender Rhodes Stage 73 Piano, a Sequential Circuits Prophet-5, a Roland Juno-106, and a Yamaha DX7. He also relies on his Alesis Andromeda A6 for analog timbres. "Each keyboard has a specific sound that I love," he explains.

The band's follow-up effort, *Atlas* (Nettwerk America, 2003), presents an ear-catching clash of pristine and gritty textures as well as musical styles. Lozano says the band's goal for *Atlas* was "to capture the vibe of our live shows, a more organic sound. We work like a collective. We start with a drum sequence, a keyboard arrangement, or a chord progression on the guitar. Sometimes we jam over a loop."

Recording *Atlas* proved to be an adventure that unfolded in several stages. Kinky began writing in Monterrey and then headed for the jungles of Quintana Roo in southern Mexico, setting up their gear in a solar-powered rented house. "The jungle was a good environment for us because it was very intense, so it gave us a good feel for this album," says Lozano.

"We need time to find the right sounds for every song," Lozano adds. "For 'The Headphonist,' we recorded the groove first, in Monterrey, using percussion, bass, and a Hammond [organ] for the cumbia licks. We wanted a dirty sound, so we recorded the snare with an SM57. In Quintana Roo, we programmed a drum loop that contrasted with the dirty sound."

Kinky continued working on *Atlas* during its 2003 tour. Each band member recorded ideas while traveling, using an M-Audio Oxygen8 MIDI controller and a Mac G4/800 MHz PowerBook loaded with Ableton Live, Propellerhead Reason, and Pro Tools software. In one instance, they tracked vocals and acoustic guitar in a hotel room, using a Neumann TLM 193 condenser mic and a pair of Shure SM57s. "We just grabbed the effects rack and put cushions on the walls," Lozano says. "We tried to isolate the room, but having that natural reverb was also part of the idea. It's important to us to keep the vibe."

Kinky completed *Atlas* in Los Angeles using commercial studios. Veteran recording engineer Thom Russo recorded those sessions and mixed *Atlas*. "He had seen our live shows, so he had a perfect idea [for achieving] that sound," Lozano says. "He suggested recording drums, bass, and some guitars to 2-inch analog tape and 24 tracks to get a warmer sound and a little distortion. We kept the drums and the bass on tape and synced the tape with a Pro Tools|HD system.

Atlas/Kinky

"We got interesting sounds by mixing software with analog [devices]," Lozano says. "Once you know the technical rules for doing things, then you can start improvising. Sometimes the way we find sounds is a little bit magical."

For more info, contact Nettwerk America; Web www.sonic360.com or www.kinkytheband.com.

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Recording Dialog for the Digital Arts

Record the spoken word for fun and profit.

By Nick Peck

Project studios have cropped up like weeds in the past decade, and it can be a challenge to find paying gigs to keep them going. Recording dialog is a terrific way to get some work into your project studio. Many industries have a need for human speech recorded well for corporate presentations, video games, documentaries, radio, audio books, Web sites, and independent film, to name only a few applications. You don't need tons of great audio gear—just a few choice pieces and a decent-sounding recording space.

The primary function of dialog is to tell the story. If the audience can't understand what's being said because of bad recording or is unmoved by a poor performance, the enjoyment of the story is greatly diminished. Ideally, narration goes unnoticed by the listeners; they become so captivated by the meaning of the words and the emotional energy with which they are delivered that they become oblivious to the artifice of recording and simply become absorbed in the story.

The engineer or project-studio owner wears many hats when recording voice-over, also known as VO. Your primary responsibility will be to do a good job of recording dialog with the equipment you have available. Depending on the circumstances, you might also need to line up the talent or give them creative direction to elicit the best performances. You will have to edit and organize the best takes, delivering them to the client in their final processed form. And most important, you will want to put on your most professional demeanor to keep the client, the VO artists, and the producer cool and focused during what might be long, arduous hours.

"I Switched"

Who: "Prince Charles" Alexander

Occupation: Music producer, mixer, multi-instrumentalist

Honors: RIAA multi-platinum award-winner; Governor, Grammy Committee Board of Governors

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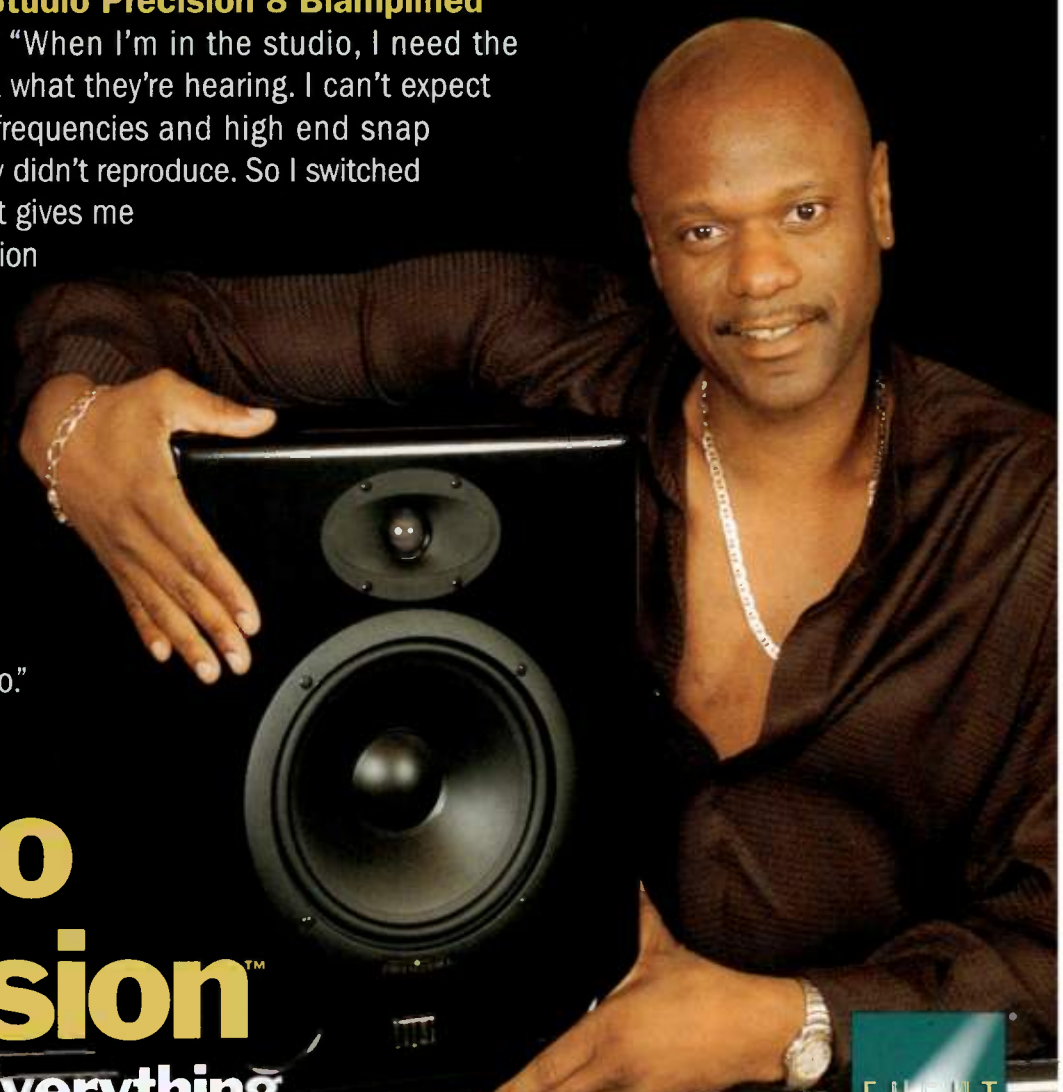
Why He Switched to Studio Precision 8 Biamplified Direct Field Monitors: "When I'm in the studio, I need the artists to be excited about what they're hearing. I can't expect them to imagine the bass frequencies and high end snap that my old monitors simply didn't reproduce. So I switched to the Studio Precision 8. It gives me smooth, full-range reproduction that sounds great and has plenty of low end thump—essential for hip-hop and r&b production. And they're 'real world' too. So I know that when a client plays my Studio Precision mix at home or in their car, they're going to hear just what they heard in the studio."



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EVENT ELECTRONICS

Recording Dialog for the Digital Arts

When you think of digital arts, multimedia might be the first word that comes to mind. Now that most postproduction is done on a computer, however, it's all digital. Thus, digital arts comprise a much broader assortment of audio tasks than they used to.

GEARING UP

You don't need a lot of gear to record human speech well. A small arsenal of carefully chosen, high-quality hardware and software will greatly improve your end product.

The acoustic space. For almost all recording, the most important equipment is a space that sounds good; recording dialog is no different. For VO applications, you generally want to minimize room reverberation and create a dead, direct, intimate sound. You can best accomplish this by building a vocal booth. You can build a temporary booth by using gobos to partition off a portion of your recording area. The floor should be carpeted and the walls covered with a combination of absorbers and diffusers to create a dead, clear sound with minimal resonances.

Inside the vocal booth, place a high stool, a music stand, a cup holder for water, and a mic stand with a pop filter (see **Fig. 1**). I put carpet or a towel on the music stand to deaden the metallic reflections. A reading lamp attached to the top of the mic stand will help your talent to see the script. You'll also need a pair of headphones connected to your talkback system so you can communicate freely with the VO talent.

If you're recording ADR (Automated Dialog Replacement, the process of rerecording an actor's voice against a previously filmed performance) or any other dialog in which the talent needs to time his or her reading to visual elements, you will need a video feed to the recording area. I use an LCD flat-panel computer monitor with an NTSC-to-SVGA converter box, with an S-Video cable running to the monitor from the analog output of my computer's digital-video-capture card. The LCD monitor is much quieter than a standard CRT video monitor, and it takes up a lot less room.

Microphones. Proper microphone choice is a critical aspect of recording VO. You generally want to achieve a rich, creamy vocal sound, but your choice of microphone is tempered somewhat by the dialog application. Fortunately, most mics you would buy for recording dialog work just as well for general musical purposes.

For general-purpose dialog recording, start with a large-diaphragm condenser microphone such as a Neumann TLM 103 or U 87 or an AKG C 414 B. Those mics record plenty of detail across the audible frequency range, so they do well at capturing the human voice's wide dynamic range. You can try some budget large-diaphragm condensers, too, but be aware that many of those mics have a hyped high end that increases sibilance and are missing some of the midrange that accounts for the human voice's warmth.

For a radio-type sound, with a less-pronounced high end and big, beefy low end, look to large-diaphragm dynamic mics. The standard mic for this application is the Electro-Voice RE20, which sounds just great on voice. For an old-time, golden-age-of-radio sound, experiment with ribbon mics such as the Audio Engineering Associates R84. That model is based on the RCA ribbon mics that dominated broadcast for 30 years (see **Fig. 2**).

Finally, if you are recording voice for ADR, in which you need to match the sound of production dialog, you should use the same microphone that was used on the set. Your best bet along those lines is the Sennheiser MKH 416 shotgun mic. Unfortunately, the MKH 416 is not as useful for general music recording as the others, and it is rather pricey. I use mine for field and Foley recording as well as dialog and have never regretted the purchase.

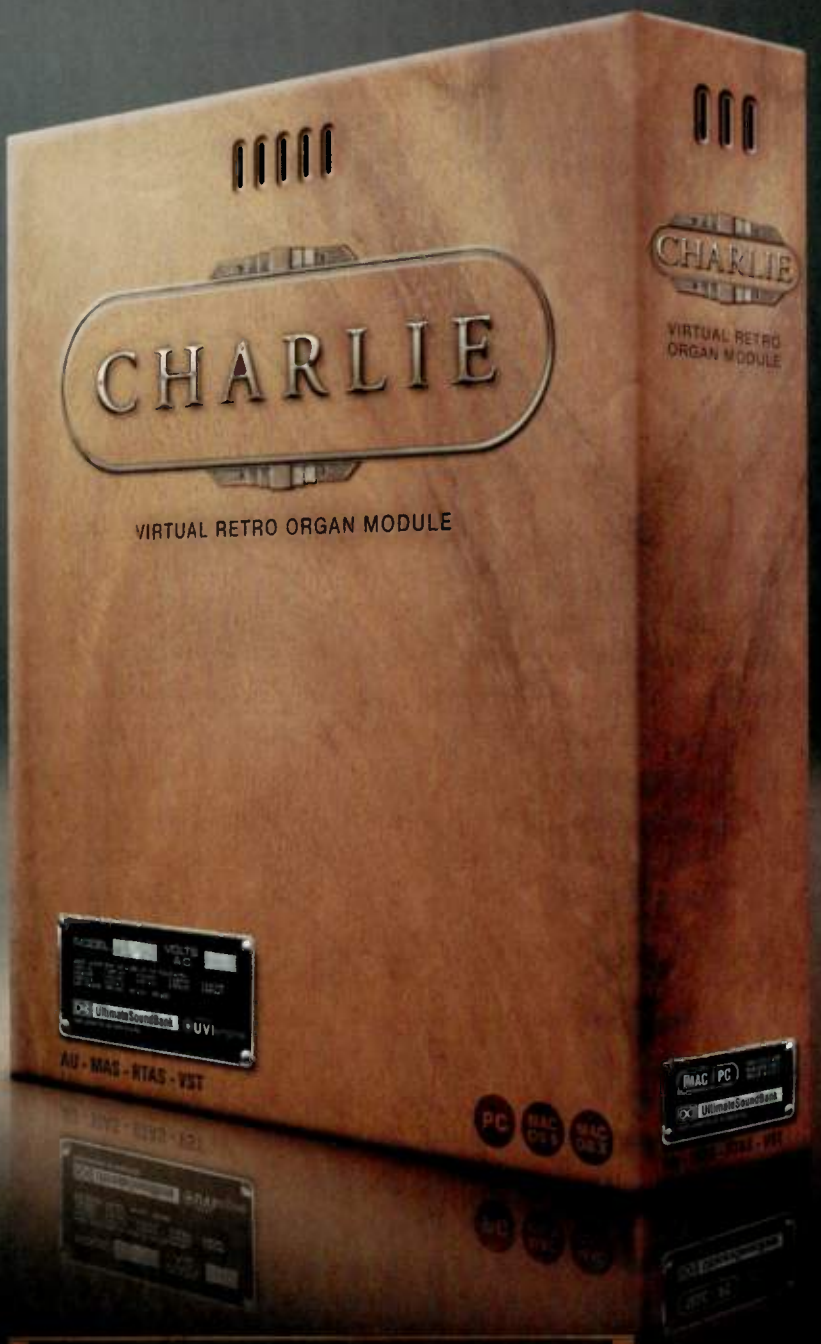
Be sure to use a pop filter in front of all the mics I've mentioned except the RE20. It will tame plosives and keep diaphragms clean and free of debris. A pop filter is particularly critical on ribbon mics because their delicate insides can be ruined by a heavy burst of air.

Mic preamps. A clean, colorless, high-quality mic preamp is the right choice for VO recording. Nine times out of ten, I reach for my Millennia HV-3D, which I can recommend as a quiet, high-gain, solid-state mic pre. If I were intentionally going for an old-time radio type of effect, I might use a tube mic pre in conjunction with a ribbon mic. In general, though, I want the detail and quiet amplification that a good solid-state mic pre can provide.

A high-quality channel-strip unit could serve well for dialog recording, too. Look for units that are as transparent as possible. One decent choice is the Langevin Dual Vocal Combo, which

NICK PECK

FIG. 1: Make sure the voice-over talent has everything he or she needs within easy reach.



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HL10

offers two channels of quiet preamplification, simple tone-control EQ, and smooth optical compression.

Compressors. The human voice is a highly dynamic instrument, and compression is routinely used to get dialog to sit properly in a mix. The amount of compression you should use will vary depending on the application, but dialog that must be intelligible over loud background sources, such as in radio ads and video games, tends to get squashed pretty hard. I would never use heavy compression while recording, preferring to adjust it on the way out instead. I do use a Universal Audio 1176 frequently for light peak limiting while recording, however.

EQ and de-essing. I like to think that the best EQ is proper microphone choice and placement for a given voice.

FIG. 2: For voice-over applications, the Audio Engineering Associates R84 looks and sounds a lot like a vintage RCA ribbon microphone.

Sometimes, though, a bit of corrective equalization is necessary to enhance intelligibility or to create a more pleasing sound. Be careful not to overdo it; we humans are quite sensitive to voice, and extreme processing of the spoken word will just end up sounding wrong. Typical EQ applications for VO include a high-pass filter at 60 Hz to reduce rumble, adding a gentle lift in the 200-to-500 Hz range to increase warmth, or using a high shelf at 6 or 8 kHz to add a bit of sparkle and intelligibility. If I am doing just a bit of gentle corrective EQ and feel confident about it, I will apply the EQ as I am tracking. If my needs are more complex, as when I'm matching previously recorded voice tracks, I will wait until the mixdown stage to apply EQ.

FIG. 3: A de-esser is a frequency-dependent compressor that can tame excess sibilance. Pictured here is the Waves DeEsser plug-in.

A de-esser is a tool that lives in every VO engineer's arsenal (see Fig. 3). It does exactly what it is named for: it lowers the sibilant *ess* sounds in speech, which can often sound sizzly and unpleasant. More specifically, a de-esser is a frequency-dependent compressor that lowers the gain of the input signal when high frequencies, such as those generated by *s* and *t* sounds, are present above the signal's threshold amplitude. De-essers come in hardware and plug-in flavors, and both work well. By the way, you can turn any hardware compressor with a sidechain input into a de-esser. Simply route an equalized version of the voice that emphasizes the center frequency you most want to remove into the sidechain input, while passing the unfiltered voice into the audio input.

Recorders. Just about any computer-based workstation can be used for voice-over recording. A simple 2-track package is usually fine, so long as it supports robust editing. I would steer clear of tape-based systems, simply because editing is such an important part of the process. But Nuendo, Sonar, Vegas, Pro Tools, Audition, and other packages will do the job for you. A laptop computer with a 2-channel USB input device could make a fine portable system for recording and editing VOs on location.

THE PEOPLE

The client. The client is, of course, the person who is paying for all the fun, and ultimately the person you need to please the most. The client may also be the producer or director, in which case he or she will often take creative control during the recording process. When working with the client, ask lots of questions to find out in advance what is wanted. Ask what types of sound the client has liked in the past, and whether he or she has examples of similar material.

You need to hammer out exactly what your final deliverables are. Those would typically be short individual sound files for interactive work, or longer edited performances for projects such as documentaries or books on tape. Establish what the final delivery format should be; CD-R is most common, but DAT and multitrack digital tape are also frequently used. Determine whether you are working on a time-and-materials basis, or if you are supplying an *all-in* bid. If the latter, be sure to include your

time for editing, processing, bouncing, and transfer of materials.

The producer or director. Many projects will have a producer or director, whose job is to supervise and creatively control the performance process. That person will usually be involved in casting (finding and hiring the right voice for the project), script development, and directing the VO talent's actual read.

The talent. Finding the appropriate voice for the job is often the most difficult part of the dialog-recording process. The budget and circumstances of the project often dictate the casting process. More often than not, a director or client will have a talent already lined up before contacting you to record a session. Sometimes, however, you are tasked with finding the VO talent yourself.

The first step is to determine whether you want to use union or nonunion talent (see **Fig. 4**). Most professional voice-over talents are card-carrying members of SAG (the Screen Actor's Guild) or AFTRA (the American Federation of Television and Radio Actors). Such actors cost a bit more than nonunion talent and have to be sponsored by a union signator in order to work a particular job. Typically, the producer is the union signator, which means that the producer agrees to use only union voice talent for the project. Strict rules must be followed concerning breaks, number of different voices a talent can perform in a given session, and so on.

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Leave a decent amount of headroom so that the talent can emote.

So why consider using union instead of nonunion talent? Because they are often more experienced, which can net you better results in less time. In addition, many projects, and some recording studios, require union talent. Darragh O'Farrell, director of the voice department at LucasArts Entertainment Company, has directed countless dialog-recording sessions, working primarily in video games and animated television. According to O'Farrell, "In general, union talent is more polished, more prepared to walk into the studio and perform. Things like mic technique are already there. There's never the need to go through a teaching process during the session."

Nonunion talent can be a viable option for many reasons: your budget may not allow for union talent, you may not be in a locale with a strong union presence, or you may not want to hassle with the paperwork. O'Farrell has occasionally brought nonunion "diamonds in the rough" into the union to make use of their specific talents. To be sure, many nonunion talents can do your job justice, but your casting search is also likely to come up with candidates that are more unqualified or inexperienced. People with less

experience will make more mistakes and have a narrower expressive range than the real hotshots; listen to their demo tapes and screen them carefully before moving forward.

If you're looking for the right union talent, your best bet is to contact a local casting or talent agency. Tell them your requirements and they can narrow the field, leading you to a number of qualified candidates. If looking for nonunion talent, you can contact theater groups or post on computer bulletin boards in the film and radio jobs area. Alternatively, if an acting or voice-over school is in your area, contact someone there. Schools are always happy to find paying work for their students. In any case, insist that the prospective talent send you a demo tape or point you to a URL with samples of their voice. If the prospect hasn't gotten far enough to create a demo tape yet, move on to someone who has.

THE PROCESS

The script. Scripts for voice-over sessions often follow the standard format of film or television scripts. Sometimes, though, they are simply a series of paragraphs of text. Either way, it's best to get a copy of the script to look over before the recording session whenever possible; familiarizing yourself with the material could make the session run more smoothly. Scripts usually have wide margins to allow you to make notes and write down the numbers of the best takes. Voice directors scribble liberally on the scripts, underlining words or phrases they want the talent to punch up or emphasize. I usually do that with a red pencil.

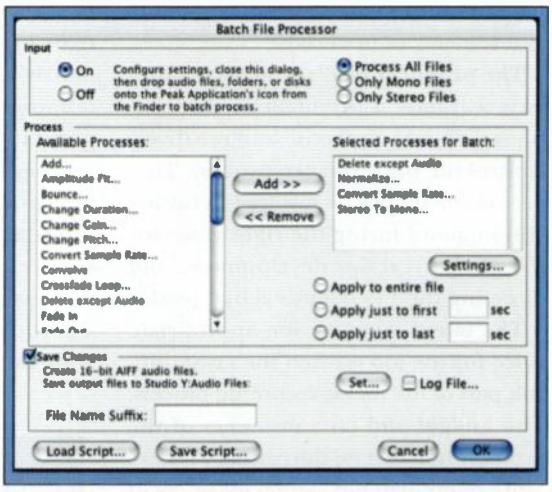

FIG. 5: Some digital audio editors, including BIAS Peak (shown here), provide batch processing capabilities that allow you to prepare a folder full of files simultaneously for final delivery to your client.

Correctly formatting a script in advance of the session can save a great deal of time during recording and editing. If recording several individual lines for interactive use, writing the actual file names beside the scripted lines is a good idea. Making your script as cinematic as possible will give the actors the best insight into the environment in which their words will be heard; knowing the context can help elicit a better performance.


DAVID GOGGIN

FIG. 4: If your budget will support it, you're usually better off hiring union talent for voice-overs. Otherwise, you might waste a lot of time with someone less experienced, or you might luck out and discover a talented newcomer by recruiting someone from a school that teaches acting or broadcasting.

The session. The recording day arrives. You have already set up your studio before the talent, client, and director show up. You have attached and tested the most likely microphone you will use, booted the DAW and prepared your recording template, and placed the script on the music stand. I always leave a glass or bottle of cold water with fresh squeezed lemon in the recording area for the talent, too. Why lemon? I have found that it helps reduce mouth noise, such as the little pops and snaps that occur from saliva moving about. All of those clicks and pops will have to be edited out, so making sure the talent's mouth is always wet will save much tedium later.

After everyone arrives, I work on getting a sound, adjusting levels, testing microphones, and tweaking EQ until everyone is happy. I like to record with a pretty healthy level, but always leave a decent amount of headroom so that the talent can emote without undue fear of clipping.

O'Farrell likes to put the talent at ease before the red light turns on. "The first trick to getting a good performance is letting the talent feel

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comfortable," he says. "If I haven't worked with a particular actor before, I will usually schedule 45 minutes at the beginning of the session to simply hang out and chat. Once a good connection is established, everything becomes a lot easier."

Once recording begins, on-the-fly organization becomes crucial. I keep one eye on the levels and one eye on the script, following along as the talent reads. I always get the talent to slate, or call out, the take number of the line or paragraph he or she is reading before reading it (for example, "Roosevelt signed the treaty, take 6," or "Paragraph 23, take 4"). Then, when everyone is happy with the read, I write the number of the selected take in the margin of the script next to the text and circle it. That makes editing a far easier and less ambiguous process later on.

Recording VO has an established studio etiquette and chain of command, which is necessary to keep the production running smoothly. It is important to know your function within the operation, which determines when you should open your mouth and when you should keep it shut. In general, the director is in charge of making the creative decisions, and the engineer is responsible for recording the session. Typically, the engineer would only discuss a performance if there were a technical problem with the recording. Many directors get annoyed at engineers piping up with their opinion about the aesthetics of a particular read; they feel it is an unwelcome distraction to the production process.

Some directors do welcome feedback from the engineer, though. O'Farrell is among them: "I feel that the more heads put together, the better. I'm open to listening to feedback from the engineer. I've always tried to foster team spirit within my sessions, because other people might notice problems while I'm thinking about some other aspect of the script." Unless you have worked with the director and talent before, though, it is safest to err on the side of caution and offer your opinion only if asked.

EDITING AND DELIVERY

After you complete the recording session, it's time to shape the raw takes into finished materials. Exactly what form they will take depends upon the nature of the project. I start by moving through the material in a rough first pass, keeping the takes that I circled on the script and discarding the rest of the material (after making a backup, of course).

During this process, you may have to

create a composite performance from multiple takes. Whenever possible, it is usually preferable to keep entire takes rather than pasting separate performances together, so you can get the performance in one long flow. However, sometimes it is difficult for the actor to maintain consistency for a lengthy scene. O'Farrell says, "Sometimes we'll end up cutting multiple takes together, but from my standpoint that's a last resort for when you can't get it otherwise. Getting the flow in one continuous take is the ultimate goal."

Once the rough cut is complete, I make a polish pass, editing out unwanted mouth noises, gulps, and other funk from the recordings. I do leave breaths in, for the most part; otherwise the performance would sound subtly unnatural. On some projects, the next step might be to lay the dialog over

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The best EQ is proper microphone choice and placement.

music or sound effects; on others, the voice is all that is required. Either way, your next step is to add any EQ or compression, as discussed earlier.

At this point, if you're recording dialog for media such as film, radio, or books on tape, you are almost finished. Bounce your finished tracks down to the agreed-upon format, record to DAT or burn to CD-R, and celebrate a job well done.

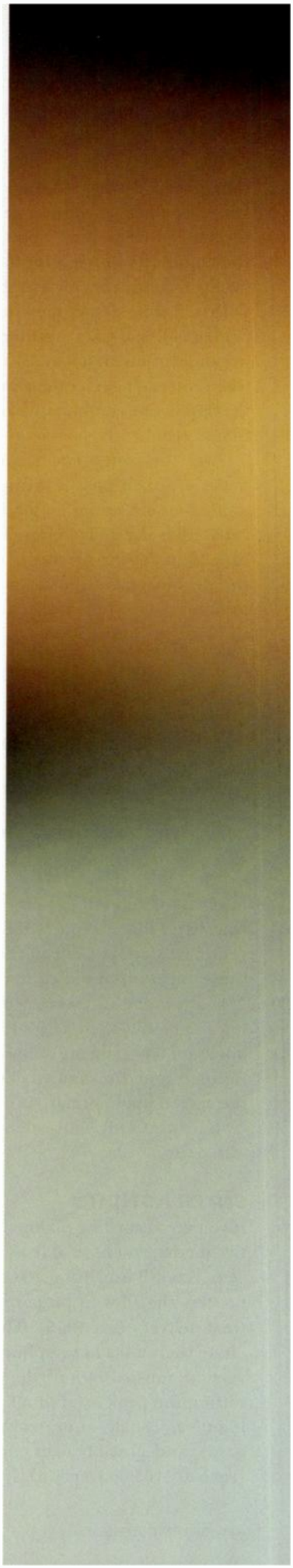
ENTER ACTIVITY

If you are recording dialog for interactive media, you have at least one more step. You will usually be asked to batch-process the files to prepare them for final delivery (see **Fig. 5**). That will include such tasks as sampling-rate conversion, normalization (the process of setting the peak level of all files to be identical, usually at or near maximum gain), and possibly data compression (such as MP3 conversion). In addition, files are often converted from one file format to another, such as from AIFF to Broadcast WAV. Tools such as Sound Forge, WaveLab, Peak, and BarbaBatch will help you accomplish batch conversion. I always deliver both the high-resolution original files and the low-resolution, batch-converted files to the client.

Armed with a little gear, a bit of chutzpah, and this article, you are ready to go record the next Orson Welles, James Earl Jones, or Meryl Streep. Best of luck, and never forget the magic words: "That was great, but unfortunately we had a technical problem. Can we take it again?"

Nick Peck *is a composer, keyboardist, and sound designer-engineer. His day gig is Sound Supervisor at LucasArts Entertainment Company. Special thanks to Darragh O'Farrell for his ideas and insights into the dialog-recording process.*

Smokin' CONDENSERS



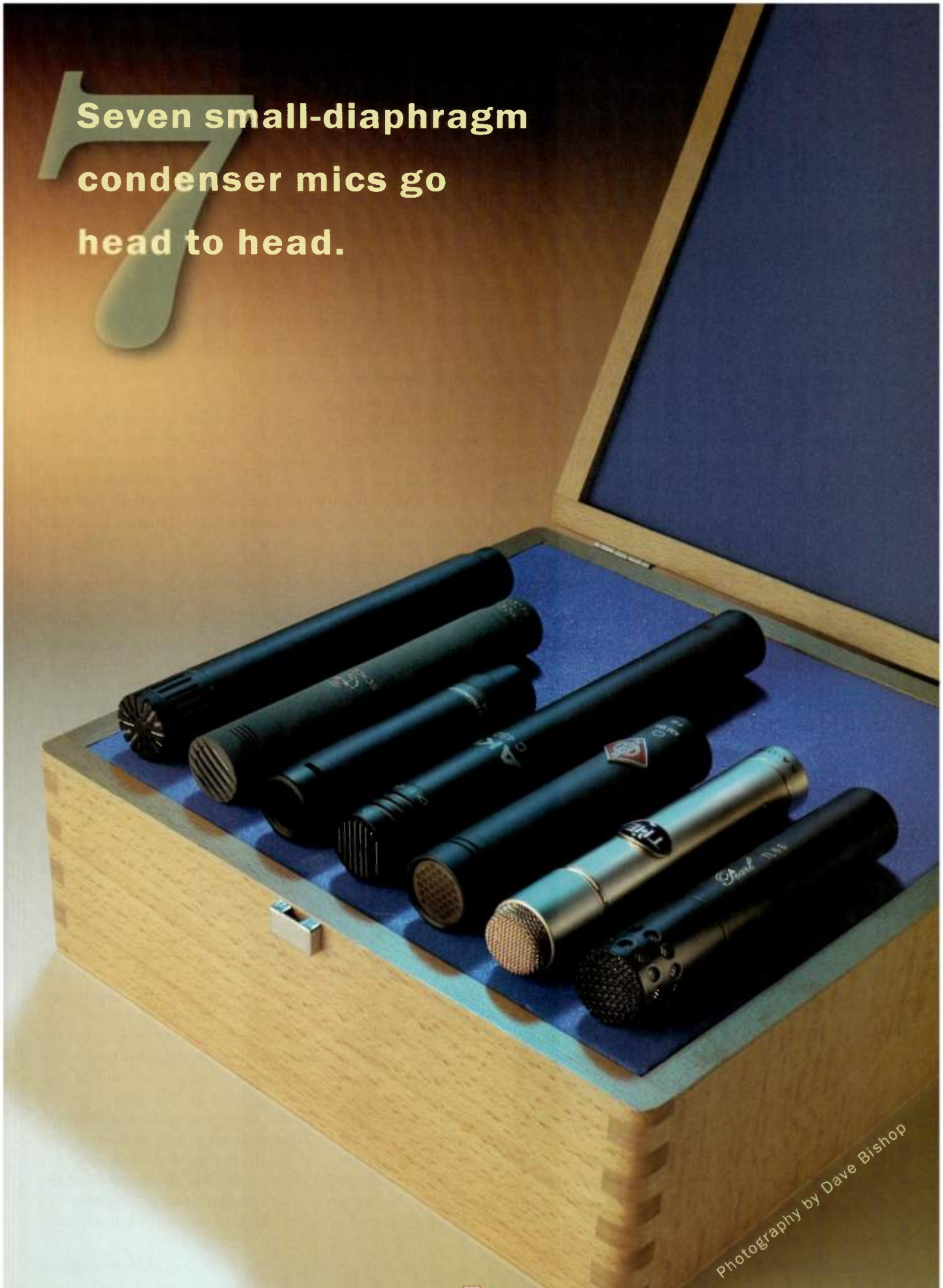
Small-diaphragm condensers—the bread-and-butter mics of the studio—are frequently used in situations that call for bright or accurate high-end pickup, fast transient response, and relatively uncolored fidelity. As such, they are often used singly to record hand percussion, plucked stringed instruments such as acoustic guitar, and bowed stringed instruments. They are used in pairs on drum kits (as overhead mics), piano, mallet percussion, and chamber or string ensembles. In addition, small-diaphragm mics are perfect for medium-distance spot miking of large ensembles and of operatic vocalists (as seen on the Three Tenors broadcasts). However, small-diaphragm mics are rarely used for close-miking vocalists, because they tend to be sensitive to popping and lack the enhanced warmth of large-diaphragm models.

In the classical recording world, highly accurate small-diaphragm mics—especially those with a relatively flat frequency response—are standard equipment for concert-hall recordings of symphonies, choirs, and chamber groups. Matched stereo pairs or specially designed surround configurations, in which all the mics are tested and have identical responses within close tolerances, are preferred by audio purists for that kind of work.

For this article, I compared an international collection of cardioid condensers priced from $480 to $1,100 each. All are end-address microphones with straight, cylindrical bodies—often referred to as pencil condensers—and diaphragms measuring approximately half an inch in diameter. Each mic is a solid-state, transformerless, externally biased "true" condenser using 48V phantom power. With the exception of the AKG C 480 B/CK 61, which is 6.75 inches long, all of the mics are 4 to 6 inches in length.

By Myles Boisen

7 Seven small-diaphragm condenser mics go head to head.



Photography by Dave Bishop

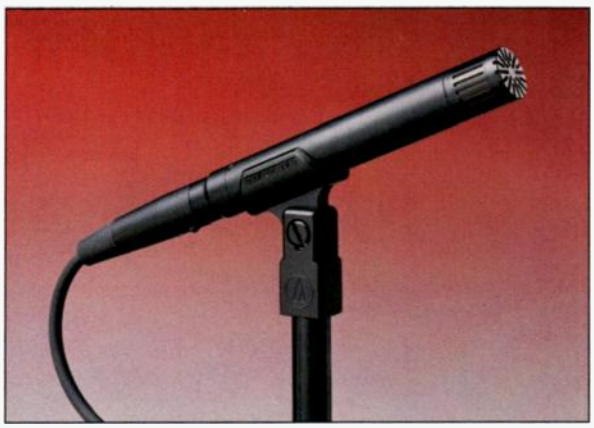
Smokin' CONDENSERS

Many of the mics featured here are part of modular systems, made up of a preamp body and an array of removable and interchangeable capsules (see **Fig. 1**). Generally, modular mics represent the top of a manufacturer's product line, in terms of both electronic sophistication and price. Most manufacturers of modular systems offer omnidirectional, hypercardioid, and other unidirectional pickup-pattern capsules. Some also make specialized modular capsules such as binaural spheres, large diaphragms, and pressure-zone plates. For this article, I tested the microphones with the recommended standard cardioid capsule only.

This article is not meant to be an all-inclusive roundup of pencil condensers in this price range. To narrow the list of transducers to a manageable size, I've left out mics that appeared in Brian Knave's March 2000 article on small-diaphragm condensers, "To Tell the Truth"—which featured eight small-diaphragm condenser mics priced under $600—as well as mics recently reviewed in **EM** (the text of articles mentioned here is available online at www.emusician.com).

THE MATCH GAME

The term *matched pair* means different things to different people, microphone manufacturers included (see the sidebar "Match, Anyone?"). Unfortunately, there is no industry standard for microphone matching, but the companies in this review represent a range of common approaches and procedures for mic pairing.



Audio-Technica AT4051a

Of the seven manufacturers, three—Audio-Technica, Neumann, and Pearl—sent consecutively or closely numbered mic pairs without documentation of any matching program. AKG sent two mics, made months apart, that have divergent numbering, but the company included documentation of the mics' closely matched frequency response. Only Schoeps and the two smallest companies here, Josephson and T.H.E., have in-house programs for assembling and rigorously testing matched pairs.

SETTING THE COURSE

All of the listening tests were conducted at my studio, Guerrilla Recording, in Oakland, California. For the tests using live musicians, I recorded two mic pairs in each pass through two stereo FMR RNP8380 mic preamps to a 1-inch, analog, 16-track tape machine. The musicians played to recorded guide tracks so that the multiple performance passes would be of approximately equal intensity and would sync up easily for comparison.

For auditioning, fader levels were calibrated using meters and by ear to establish equivalent listening levels and to compensate for slight variations in gain between mics. The players' observations about their tracks were solicited in separate listening sessions after all the recording was complete; I've included an assortment of their comments to round out my own impressions in the following discussion of the individual mics.

With the help of drummer and **EM** contributing author Karen Stackpole, I recorded an Ayotte travel kit in my live drum room. That setup let me hear how the mics reacted to both fast and slow rock drumming in a room with plenty of ambience. For these tests, mic pairs on stereo bars were set in XY configuration with an angle of about 90 degrees between them, on boom stands placed at eye level about seven feet in front of the kit. Stackpole's listening evaluation took place a few weeks after the session and was done as a blind test with no visual indication of which mic pair she was listening to at any given time.

I performed the acoustic guitar tracks myself in the relatively dead and unflattering acoustics of a large vocal booth. For this test, I split up the mic pairs, using one as a close mic about a foot from the instrument's 12th fret, and the other as a distant room mic. Leaving the vocal booth's double doors open, I positioned the distant mics about eight feet away, again at eye level, on a boom stand pointing at the guitar from an adjacent carpeted area of the studio. Stylistically these performances ran the gamut from chunky low-end chords to full strumming to intricate moving lines.

For the string-duo recordings, I employed cellist Marika Hughes and violinist Carla Kihlstedt of the group Two Foot Yard (you can read more about

ROB FORLIN

FIG. 1: The T.H.E. microphone system lets you combine a KA-04 preamp with a number of large- and small-diaphragm capsules. This set includes cardioid, hypercardioid, diffuse-field omni, and free-field omni small-diaphragm capsules.

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this group's style in my November 2003 article "Tracking in the Unplugged World"). Opting for a naturally roomy, medium-distance recording, I placed the duo in the live room about six feet apart. The mic pairs were split in this case, with one mic a couple of feet in front of the cello bridge, and the other above the violin, pointing down at the bridge from a distance of four feet. On an original composition covering a range of dynamics and techniques, this miking arrangement was intended to capture a natural timbre from both instruments. The setup gave these tracks a healthy blend of room sound, and it resulted in off-axis bleed of each instrument into the other intrument's mic.

To get a slightly more scientific take on the timbre of these mics, I set up a controlled loudspeaker test. In this test, the mics were set up one by one at exactly the same position in a shockmount 2 feet in front of a full-range powered monitor. Each mic then recorded a variety of complex music mixes to a digital 8-track recorder. Test conditions were as close to laboratory measurement standards as I could arrange in my studio, and playback and recording levels were carefully calibrated using standard test tones and the best gear at my disposal.

The loudspeaker test had three phases: an initial test that confirmed stereo matching of five of the pairs to my satisfaction; a second test that compared all seven models with each other under identical conditions; and a final retest of two stereo pairs—the Neumanns and the T.H.E.s—aided by the able ears of Karen Stackpole. Evaluations of the seven-mic test over Tannoy PBM-8 monitors and Grado SR-125 headphones were also performed under blind-test listening conditions.

Along the way, I also evaluated these mics for self-noise, output gain, and susceptibility to low-end rumble. In these categories, all the mics performed similarly and as expected for transducers designed for critical recording uses.

Josephson Series Four C42

AKG C 480 B/CK 61-ULS

Although the Austrian AKG C 480 B/CK 61-ULS modular system ($1,047 each) is one of the most expensive mics tested for this article, its packaging is far from fancy. Each mic comes packed in light foam inside a noncorrugated cardboard box. A zippered mic pouch with the AKG logo, a foam windscreen, a plastic SA 40 swivelmount, and a manual accompany the mic.

AKG included an individual frequency-response chart with each capsule. The charts show that this pair of CK 61-ULS cardioid capsules (serial numbers 36610 and 36759) have a 1 dB difference from each other from 80 Hz to 11 kHz. Below 80 Hz one of the capsules rolls off gently. Both capsules have presence peaks of about 2 dB at 5 kHz and 10 kHz.

The C 480 B preamp body offers a range of switchable options that are rarely seen on mics of this type. The low-cut switch has three positions: flat (marked "lin" for linear), 70 Hz cutoff, and 150 Hz cutoff. The highpass filter has a 12 dB-per-octave slope. A gain switch also features three settings: +6 dB (a boost designed for use with the lower-output CK 69-ULS shotgun capsule), 0 dB, and a –10 dB pad position. Optional capsules for this modular mic system include the CK 62 omni, the CK 63 hypercardioid, and the CK 69-ULS supercardioid shotgun.

Because the right-side mic of the C 480 B/CK 61-ULS pair began malfunctioning during the drum test, I made a separate evaluation of it against the right-side mics of all the other pairs. It struck me as nice and bright whether heard on its own or mixed in with the guide track. In general, the C 480 B/CK 61-ULSs also delivered a good, sharp attack on all the drums, but lacked some fundamental lower-midrange tone on the kick and snare.

Cellist Hughes and I agreed that this mic excelled on arco (bowed) cello and gave the duo recording a high-end sheen that could be perceived as either bright or thin. In fact, our string listening session ended with a long discussion on the merits of the characteristically crisp AKG sound contrasted with the smoother tonality of the Neumann KM 184 pair in this test. The musicians seemed torn between the two, concluding that the C 480 B/CK 61-ULS was more open in the high end, with a pop sound and more condenser-like shimmer than the KM 184. But in terms of fidelity, the C 480 B/CK 61-ULS didn't quite capture a balanced violin tone; as Kihlstedt put it, it was "not bad, but not special."

At both positions in the guitar test, the C 480 B/CK 61-ULS had a mellow tone quite similar to that of the Neumann and Schoeps mics. These three mics were also similar in that their sound was not always as incisive and bright as I would have liked for a sparkly pop mix. All three had a good, solid midrange tone, but in the close-mic position the C 480 B/CK 61-ULS missed some of the low-end tone and chunkiness on my chordal playing.

In loudspeaker tests, the C 480 B/CK 61-ULS delivered a full sound that was similar to the Neumann KM 184 through headphones. Heard on monitors, its midrange was flat—it was close to the still-flatter midrange of the Schoeps, but noticeably crisper. At times the C 480 B/CK 61-ULS brought out some harshness in cymbal sounds.

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Smokin' CONDENSERS

AUDIO-TECHNICA AT4051A

The Japanese-made AT4051a ($595 each) arrives in a classy padded plastic case that snaps closed with a metal button latch. A foam windscreen and durable AT8405 swivelmount with a solid metal base are included in the roomy case. Rectangles cut into the case's dense foam can be removed to make room for additional modular capsules, namely the AT4049a-EL (omnidirectional) and AT4053a-EL (hypercardioid).

A single-sheet manual sent with each mic details specifications and important user information. According to the manufacturer's chart, the frequency response of the AT4051a cardioid capsule featured in these tests is basically flat below 1 kHz, with a mild bass boost around 50 Hz. Its high-frequency response begins gently rising at 2 kHz and peaks at two "humps" centered at 7 and 15 kHz. The preamp body (an AT4900a-48) has a low-cut switch that engages an 80 Hz, 12 dB-per-octave filter. The serial numbers—A02373 and A02374—on the mics sent for review are identical for the preamp and capsule.

Audio-Technica's small diaphragm elicited a unanimously positive opinion from the participants in the string listening session. The mic was repeatedly described as "smooth." Kihlstedt and I both commented on the pleasant softness of the violin's upper register through the AT4051a. Hughes, who tended to find the other mics suited either for arco or for pizzicato playing, praised the AT4051a's handling of both bowed and plucked lines. She also remarked that it conveyed a "human quality" that was rich in depth and clarity without being too defined. For Kihlstedt and Hughes, who do a wide range of professional work in studios and in symphonies, the AT4051a seemed to strike just the right balance to work well with a variety of musical and production styles.

The AT4051a really impressed me when used as a close-mic on acoustic guitar. It seemed to have all the body of the warmer Neumann and Schoeps models, with the added benefit of a perfectly defined high-end sparkle. At a distance, it took on a smoother and more neutral character not unlike the Schoeps, but retained its characteristic warmth and presence.

On the loudspeaker tests, the AT4051a conveyed a solid and impressive midrange that subtly brought the snare drum, vocals, and other lead instruments to the foreground. The mic's overall airiness and listenability were as pleasing as the Josephson's and the T.H.E.'s. And the AT4051a stood out from the pack in terms of its rich mids and upper bass, big sound, and overall smoothness. Scheduling problems prevented the Audio-Technica AT4051a pair from arriving at my studio in time for inclusion in the stereo drum test.

JOSEPHSON ENGINEERING SERIES FOUR C42

The only American-made microphone in this article comes from the small shop of designer David Josephson in Santa Cruz, California. Against the advertising budgets of the industry's Goliaths, this David doesn't have much of a chance. But a quick glance at Josephson's tech-savvy Web site testifies to the deep theoretical knowledge of precision recording and measurement mics that enables Josephson to stand tall alongside any of the international audio giants. In fact, Josephson has made capsules for a number of better-known mic companies—years ago Groove Tubes and Manley became the first to incorporate his designs into their product lines.

The nonmodular Series Four C42 ($480 each; matched pair with black chrome finish in case, $1,060) is in the entry-level Josephson line. The matched stereo pair I received (serial numbers 2623 and 2624) arrived in what must be the smallest Pelican case made. This hard black-plastic enclosure is 6 inches wide, 2 inches high, and 4.5 inches deep—and, yes, it holds both mics. It locks closed with a sturdy latch and even has an air-pressure valve.

Describing his procedure for mic matching, Josephson says, "Every C42 microphone is tested for frequency response in our anechoic chamber. C42MP matched pairs are selected for best curve fit over the audio range before the production serial numbers are assigned. Regular production C42s match each other very closely anyway, but we are sensitive to users' needs for precise stereo imaging and have offered curve-matched pairs for the past 15 years."

The mic pair ships with two shockmounts. These resemble a standard stage-mic swivelmount except that the mic is held in a thick sleeve attached to the base by what looks like a massive rubber band. No windscreen or documentation was included.

Stackpole picked this pair as her personal favorite for drum recording, summing it up as "tight and tonally well balanced, with good presence, sharp attack, and a sparkly high end." She praised this mic's ability to focus and unify the sound of the kit so that none of its components were washed out or too prominent. To her, this pair sounded rich, but also brighter than the Schoeps and Neumann pairs.

I, too, thought the C42s did something unique to the drum kit, adding immediacy by sharpening the direct transients and separating foreground details from the reflected room ambience. Because of the C42s' superior resolution and high-end detail, I was actually able to discern a flutter echo generated by the kick beater in the drum room. At times the Josephsons might have been too sizzly around

AKG C 480 B/CK-61 ULS

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7 kHz, but the sound was never too bright or thin. These drum tracks sounded crystal clear when mixed in with the guide track at varying levels.

Although Kihlstedt and I both thought the C42 was a bit too bright on the violin, it did, once again, enhance the dimensionality and realism of our live-room recording. The ambience on this track was somehow cleaner and more dimensional, resulting in better stereo imaging and startling fidelity. Hughes interpreted this difference as an increased amount of reverberation on the cello and thought that it complemented her arco playing. She also commented at length on the violin sound, in which she heard an enhanced sense of depth and clarity, as well as nuances of expression that the other mics didn't pick up.

With its winning balance of woody tone, crisp highs, and transient detail, the Josephson C42 helped my inexpensive guitar cut through the guide track mix at differing levels. But even though I had used the Josephson on commercial sessions with the same guitar and loved its tone, from this test position the close mic sounded boxy in the lower midrange. At a distance, it came across as lively and seemed physically closer than the other distance mics because of its inherent sparkle and finely etched resolution.

On headphone evaluation of the loudspeaker tests, the C42s sounded smooth and clear, with a lively quality similar to the T.H.E. They exhibited a big sound with good overall balance through studio monitors, delivering a bit more high end than the Audio-Technica AT4051a and comparably solid lows.

ROB FORLIN

Schoeps U.S. Stereo Set

NEUMANN KM 184

The fixed-cardioid KM 184 ($950 each; SKM stereo set, $1,950) is one of the most widely known small-diaphragm condensers in the recording industry. I wanted to include this popular model from German manufacturer Neumann not only because of its considerable merits as a recording mic, but also because I hoped to provide a well-known point of reference for readers interpreting the results of my tests.

Although the KM 184 capsule is threaded, it is not removable, and the Neumann 180 series is not part of a modular system. Nonetheless, this mic is supported by a full range of accessories and top-of-the-line packaging. The SKM stereo microphone set sent for this article contains two mics (serial numbers 59282 and 59283). Neumann deems the specs and frequency-response curves of these units to be close to each other because of their consecutive numbering, but the company did not specially match them for critical stereo use.

A jewelry-grade wooden presentation box with a metal latch holds both mics. A pair of foam windscreens and interchangeable SG 21/17 swivel holders, which attach to a threaded metal base, are standard equipment for the set. Also included is a comprehensive manual, which shows the KM 184's frequency response as basically flat in the midrange, with a low-end rolloff starting at 200 Hz and sloping gently down to –12 dB at 20 Hz at a distance of one meter. A broad presence boost of about 2 dB extends from 7 kHz to above 10 kHz.

In her blind-test auditioning, Stackpole rated the KM 184 higher than the T.H.E. on cymbal presence, but she gave it a lower rating than the Schoeps for smoothness in the high register. In her assessment, the Neumann performs with "decent low end, good midrange, and good definition in the highs."

In listening to the drum tracks, I also noted that the Neumanns—like the Schoeps pair—picked up lots of midrange and lows from the room, adding power and a desirable "bigness" to the drum sound. Whether soloed or mixed in with the guide track, the KM 184 pair conveyed a strong tonal balance, offered crisp highs that were never overbright, and accurately represented the kit and room ambience.

Both of the string players commented on the realism of the Neumann pair, declaring that "it sounds like us" and describing the sound as honest and familiar. The Neumanns tied the Audio-Technica pair in best representing what Kihlstedt and Hughes are used to hearing acoustically without being too clinical. We all enjoyed the Neumann's richness and open, roomy fidelity, but

Pearl TL 66

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there was some disagreement in the control room about the mics' high-end character. Hughes thought the violin's upper end needed more pizzazz. I preferred the Schoeps's treatment of the violin's upper range, and for my taste, the cello's bow sound was too pronounced on the Neumann tracks.

As a close guitar mic, the KM 184 had a good, solid, middle-of-the-road tone, along the same lines as the AKG and Audio-Technica transducers but with enhanced realism and more satisfying complexity on the midrange. Listening to this close-mic track, I was reminded why the KM 184 is often a first-pick guitar mic for studio engineers working with all kinds of production styles. At a distance, it lost a bit of its characteristic warmth and tested more like the brighter mics without ever crossing over into undesirable thinness.

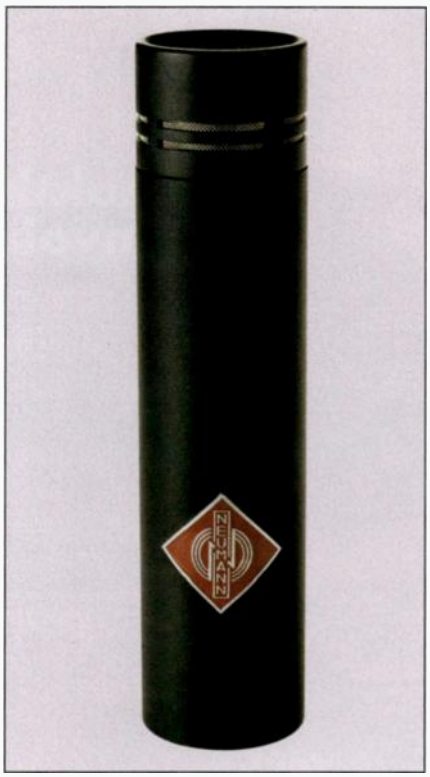
The KM 184 tested similarly to the AKG condenser on the loudspeaker mixes in terms of overall tonal balance and midrange detail. But on headphones and monitors it always sounded a bit warmer and more substantial. High-end pickup was smooth and pleasant, with no evidence of harshness on cymbals.

I decided to retest the KM 184 pair after hearing discrepancies in the stereo matching the first time around. After carefully matching the recording and playback levels of both microphones using a standard 1 kHz tone, I noticed that their channel meter readings did not behave identically with full-range mixes. When listening, I found one mic's treble response to be slightly brighter, with audible differences around 7 kHz (which is the upper range of hi-hat, acoustic guitar, and female vocals).

When I asked Stackpole to listen to the treble response of these mics, she detected no difference, cheerfully declaring their stereo matching to be "close enough for rock 'n' roll." She's right about that, and in Neumann's defense I should reiterate that the company's policy is simply to provide pairs that are sequentially numbered and within specifications, but not matched by ear.

PEARL TL 66

Sweden's Pearl Mikrofonlaboratorium definitely takes the blue ribbon in the Cutest Packaging category. The company's modular TL 66 ($500 each) arrives in a sturdy cardboard tube with removable brown caps at both ends and old-world sepia-print paper wrapped around the exterior. Inside, the microphone is suspended lengthwise within a central foam ring. This doesn't seem like the best possible protection for a premium mic, especially since the mic body can slip up and down a bit within the ring, potentially banging the soft plastic end caps. But this novel package will evidently survive international shipping and is at least crushproof and weatherproof enough for location use.



Neumann KM 184

Although a shockmount and windscreen are available as options, the Pearl TL 66 is the only mic I tested for this article that comes with no accessories. A photocopied sheet showing a universal frequency-response chart for the cardioid and omnidirectional (TL 66k) models was included with one of the mics; the sheet also had handwritten test results showing sensitivity and self-noise ratings for the mic model I received. According to the response chart, the TL 66 has a pronounced low-end rolloff beginning at about 300 Hz and high-end presence boosts at 5 and 10 kHz.

When she first brought up the unmarked faders for this pair of mics on her drums, Stackpole made a face and exclaimed, "Where did the lows go?" I had to agree that, in contrast to the other mics, the Pearls didn't do justice to the beefy tone of her kit's kick, toms,

If I Were a Rich Man

I'd be happy to have any of these mic pairs residing permanently in the Guerrilla Recording vault, and I'm sure I could find uses for all of them. Keeping economic realities in mind, I've assembled a table of my Rich Man and Poor Man picks from this lineup, broken down by recommended usage of these mics.

	Rich Man	Poor Man
Location stereo recording with a matched pair	Schoeps	Josephson
Pop studio use, stereo pair	AKG	Audio-Technica
Pop studio use, single mic	Neumann	T.H.E. or Josephson
String ensemble instruments	Schoeps or Neumann	Audio-Technica
Sampling/Foley/sound effects	AKG or T.H.E.	Pearl or Josephson
Flat response	Schoeps	Audio-Technica
All-purpose modular system and best range of accessories	Schoeps	T.H.E.

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Jason Halbert—programmer; American Idol Tour: Kelly Clarkson, Clay Aiken

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or snare. Stackpole ultimately summarized the TL 66 as delivering plenty of crisp attack but not much punch, and she singled it out as her least favorite mic of the pack. That assessment is valid, though I would add that this mic excels at clarity and could be used to add some zip to the sound of a thuddy, dull kit or a timid drummer.

I suspected that my string players would be favorably impressed by the exciting high-end clarity of the Pearl TL 66 sound. Initially they were pleased with the mic's handling of high harmonics, and I thought that the sound was very natural on restrained passages. But it became clear as the piece progressed toward its robust ending that this mic pair favored the high end too much, sounding harsh on passages that had vigorous bowing. Hughes also noted that her cello sounded muffled in the pizzicato passages, and that compared with the other mics' string tracks, the room sound on this track lacked depth.

When my guitar was mixed in with the guide track, the TL 66 helped it to cut through well and hold its own even at a lower level. For enhanced high-end zip in a bright pop mix, it is certainly the mic to beat. But alone outside the mix, the TL 66 underrepresented the bottom end of the guitar up close and sounded a bit small at a distance.

The Pearl's pickup of recorded mixes in the loudspeaker test was, as the frequency-response chart led me to expect, lean in the bass range and sometimes harsh or unpleasant in its handling of high-end information. Its distinctive frequency curve gives the Pearl an instantly recognizable sound that was always easy to pick out from the rest of the mics under blind testing conditions.

Cosmetically, this mic had a problem that the other models didn't. After routine handling for just the tests described here, the black paint around the rim of both mics' XLR connectors began to chip away.

SCHOEPS U.S. STEREO SET

The deluxe entry comes from Schoeps, like Neumann a highly regarded German microphone manufacturer. The Schoeps CMC64 ST stereo set ($2,165 per mic pair; mics not available individually) is designed around the Collette modular series CMC 6 body and MK 4 cardioid capsule. According to product manager Scott Boland of Redding Audio (Schoeps's U.S. distributor), the U.S. Stereo Set is a special packaging of the standard Colette series created for the American market to simplify ordering modular mic parts and save users some money. The wooden case is a free bonus for U.S. customers only.

The pair I received was sequentially numbered (capsule number 81229 on body number 22813, and capsule number 81230 on body number 22814) and came with a pair of windscreens and two compact suspension shockmounts. Though these mounts are made of plastic, they really do the job, and on location recordings their small size and light weight are actually a plus.

All of these parts are nestled in dense, precision-cut foam inside a gorgeous hinged wooden case. Like the boxes used by Neumann, T.H.E., and a few other manufacturers, this top-quality latching wooden case looks as though it will protect its contents and last for decades. You might even wonder if it's too nice to take out on a rainy night or to the local dive for a live jam session!

Regarding Schoeps's program for matching mics, Boland said, "There are special testing procedures for matched pair capsules, which come in all U.S. Stereo Sets. Schoeps feels that their tolerances are extremely low, and even though there is not a desperate need for exact capsule matching, the company likes to offer it. A matched pair of capsules comes with a special certificate, included with the U.S. Stereo Set. They measure every capsule and match the frequency response and sensitivity. The CMC 6 bodies are discrete Class-A amplifiers that include some handmade and matched components, and we guarantee that the bodies will have sequential serial numbers in the U.S. sets."

No documentation accompanied the Schoeps Stereo Set shipped to me. Apparently Redding Audio keeps the matching certificates for review units on file so that they don't get lost. According to a frequency chart on the manufacturer's Web site, the MK 4 capsule's response is ruler flat from

MATCH, ANYONE?

The term *matched pair* of mics means different things to different people, including the folks who make them. Some manufacturers maintain that any mic they produce of a given model sounds so much like—or "specs" so closely to—others of that model there is no reason to specify matched pairs; essentially, they claim that any two of the mics constitute a match. Other manufacturers offer, often for an additional price, to put their mics through extra testing and then pair those (typically in a single box) that sound the most alike and have the closest specs. Still others offer matched pairs with consecutive serial numbers, the idea being, perhaps, that two mics made immediately one after the other (or at the same time) offer a greater likelihood of being sonically matched as well.

I've tested many mics in recent years, and there were sometimes clearly audible differences between two mics of the same model, whether dynamic or condenser. (For the few ribbon mics I've tested, this was not the case.) Therefore, for critical stereo applications, I prefer pairs that are sonically matched. As for consecutive serial numbers, that seems to me more a cosmetic concern.

—*Brian Knave*

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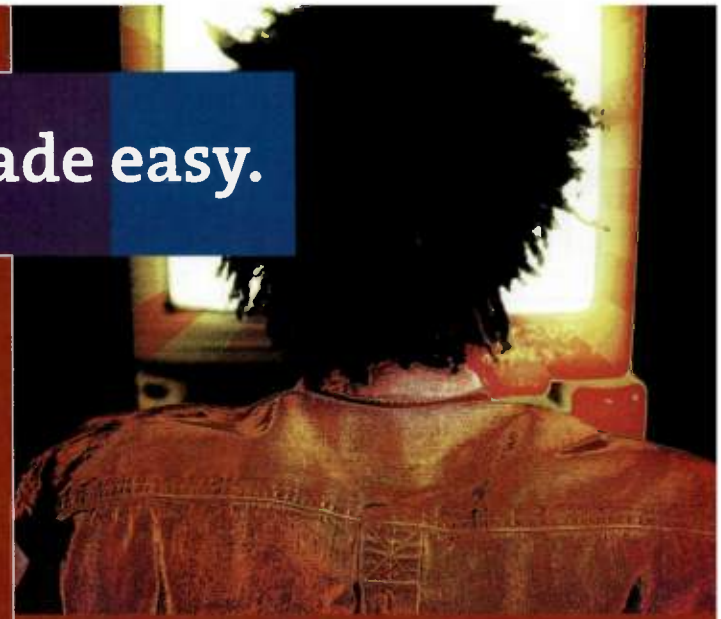

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US-428	4-in / 2-out	2 XLR mic[1], 2 ¼" TRS (bal.), 2 ¼" (unbal.)	YES	2 RCA (unbal.) + headphone	N/A	two S/PDIF	24 bit	32 ch.	GigaStudio & Cubasis VST	8 faders[2] + stereo master, Aux, Pan, Phones, Line, Jog, Transport, DAW	ASIO, WDM, GSIF, Core Audio, MME, Sound Manager[4]
US-224	2-in / 2-out	2 XLR mic[1], ¼" TRS (bal./unbal.)	YUP	2 RCA (unbal.) + headphone	N/A	two S/PDIF	24 bit	16 ch.	GigaStudio & Cubasis VST	4 faders[2] + stereo master, Phones, Line, Jog and Transport	ASIO, WDM, GSIF, Core Audio, MME, Sound Manager
US-122	2-in / 2-out	2 XLR mic[1], ¼" TRS (bal./unbal.)	YEAH	2 RCA (unbal.) + headphone	two inserts	N/A	24 bit	16 ch.	GigaStudio & Cubasis VST	2 rotary input level controls, Line, Phones, Direct Monitor	

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Drummer Stackpole characterized the Schoeps pair as having a smooth, somewhat dark sound, with a beefy low end and nice definition. She also praised this condenser for its openness and accurate portrayal of cymbals and hi-hat, blended with a solid representation of room tone. My take on this model is that it is probably too flat and dark for general use on rock drum overheads, where a condenser mic with a presence peak is often standard. But I did like the punchy low end of the MK 4 capsule and could imagine it being a real winner for applications in which a natural drum sound is more important than high-end sizzle. (Redding Audio's Boland points out that users who want a little more high end can opt for the CMC64V ST stereo set, which comes with the MK 4V capsules that have a rise of approximately 2.5 dB at 10 kHz.)

Warmth is a term that came up again and again as I listened to the Schoeps tracks with the string players. Like the Audio-Technica pair, the Schoeps mics elicited a unanimous positive response from my listeners; this mic pair sounded dramatically different from all the other presence-enhancing condensers, but everyone in the control room praised the Schoeps sound on analog tape. Violinist Kihlstedt noted that her instrument had a full-bodied "old school" sound with nice shimmer and presence, even though her mic sometimes "airbrushed the high end out a little too much." Hughes also commented on the rich, vintage sound of the duo and her arco playing in particular, but added that she preferred the sharper tone of the AKG on plucked lines.

For me, the sweetness of the Schoeps track was a little bit of audio heaven. These silky tones actually soothed my ears at the end of a long day of testing.

As a close mic on acoustic guitar, the MK 4 capsule had an unmistakably round tone with significant retro or vintage appeal. In the close position, it exhibited a slightly boomy low end, but the highs were mellow, and the Schoeps sound was much like that of the AKGs, both on its own and mixed in with the guide track. I preferred the Schoeps to the other models as a distant guitar mic because of the way it kept its smooth, detailed tone from afar.

The Schoeps CMC 6/MK 4 was predictably the flattest-sounding and most unhyped mic in the loudspeaker test. As such it was easy to distinguish, even in blind tests. By not adding or detracting much from the sound source, the Schoeps set achieves a sought-after, though elusive, degree of accuracy.

T.H.E. KA-04/KR-2C

The cardioid version of T.H.E.'s relatively new modular microphone design mates a KA-04 preamp body with the KR-2C cardioid half-inch capsule (the KA-04/KR-2C combination is priced at

Microphone Specifications

Company	Model	Diaphragm Size	Modular	Frequency Response	Maximum SPL
AKG	C 480 B/ CK-61 ULS	0.5"	yes	20 Hz–20 kHz	140 dB
Audio-Technica	AT4051a	0.7"	yes	20 Hz–20 kHz	146 dB
Josephson	Series Four C42	0.6"	no	40 Hz–20 kHz	135 dB
Neumann	KM 184	0.6"	no	20 Hz–20 kHz	138 dB
Pearl	TL 66	0.6"	no	80 Hz–18 kHz	132 dB
Schoeps	U.S. Stereo Set (CMC 6/MK 4)	0.5"	yes	40 Hz–20 kHz	132 dB
T.H.E.	KA-04/KR-2C	0.5"	yes	50 Hz–20 kHz	134 dB

$750). So far, every capsule made by T.H.E. in its Argentinean facility is designed to be used with the KA-04 body, including a broad selection of large-diaphragm, side-address, and measurement capsules. The roster of interchangeable small-diaphragm capsules includes the KR-1D (omnidirectional diffuse field), KR-1F (omnidirectional free field), KR-2W (wide or subcardioid), and KR-3H (hypercardioid).

For this article, T.H.E. submitted a stereo demonstration set with most of the aforementioned capsules and two KA-04 bodies. The components are housed in a charming American-made hinged cedar box. Custom mic sets in these latching cedar boxes are made to order by T.H.E.'s U.S. representative, H. Taylor Johnson (the *T* in T.H.E.).

T.H.E. provides a uniquely simple all-metal swivelmount, which holds the cylindrical mic preamp in a clamping sleeve. To insert or release the mic, two tabs at the top of the mount (neatly engraved with the T.H.E. logo) must be pressed together, widening the sleeve so that the mic body can be slipped in or out. This clever design is also used by Russian mic manufacturer Oktava for its popular MC012 microphone.

No windscreens or individual documentation accompanied these mics. According to the company's product brochure, all the small-diaphragm capsules are available as matched pairs for stereo recording, at no additional cost. Taylor Johnson informed me during the writing of this article that he matches these mics personally, by ear. Sometime in 2004 the company's matching duties will be split between identical T.H.E. testing facilities in the United States and Argentina. No serial numbers were visible on the demo capsules, but the preamps were engraved with individual ID markings (2NJ/2N2.)

Like the Josephson Series Four C42, the T.H.E. pair delivered a very clear, high-resolution rock drum sound without being overbright or harsh. Stackpole called it "tight and focused, direct and punchy but not a lot of room sound." She described this pair as conveying a nice low end, but missing some of the fundamental tone of the cymbals and ambience, making the overall sound more defined but less roomy than the Schoeps and Neumann pairs.

This mic pair was on par with the Josephsons in its ability to convey a drum sound that is not only accurate, but also pleasing to the ear and more or less ready to mix without additional high-end EQ. Despite the relatively distant mic placement, these two pairs always sounded timbrally balanced and three-dimensional.

For her opening pizzicato lines, cellist Hughes instantly liked the sound of this mic. Kihlstedt described the T.H.E. timbre as "a more modern sound" with a narrowly focused EQ that she thought worked better for her violin than for the cello. Further into the audition, both classically trained string players decided that for their tastes, the timbre of these mics was a bit too clear and clinical, accentuating bow sound too much and lacking in the cello's low range. Because my preferences run toward warm string sounds, I agreed: this sound has more promise for pop studio work than for audiophile concert-hall recording.

Pad	Bass Rolloff	Factory Matched Pair	Accessories	Price
–10/+6 dB	70/150 Hz	no	swivelmount, windscreen, and bag included; full range of modular accessories optional	$1,047 each
no	80 Hz	no	swivelmount, windscreen, and case included; full range of accessories optional	$595 each
no	no	yes	shockmount and case included with matched pair	$480 each; matched pair $1,060
no	no	no	swivelmount, windscreen, and case included; full range of accessories optional	$950 each; SKM stereo set $1,950
no	no	no	shockmount and windscreen optional	$500 each
optional	optional	yes	shockmount, windscreen, and case included; full range of accessories optional	$2,165 per pair (not available singly)
no	no	yes	swivelmount and case included; full range of accessories optional	$750 each

Smokin' CONDENSERS

At both positions this mic was my "bold and bright" favorite for guitar. Up close it represented the acoustic guitar more accurately than the other contenders, and it also invigorated the sound of my rather average axe with an exciting, top-dollar sheen. From a distance, it kept its sharp definition without washing out or sounding thin. Like the Josephson C42, the T.H.E. mic cut through the guide track well at varying levels with its woody tone, crisp highs, and transient detail.

In the loudspeaker test over headphones, this mic was again similar to the Josephson and to the Audio-Technica with its clear timbre. But the T.H.E. had a different high-end emphasis that seemed a little too strong in the range of 2 to 4 kHz. When heard over studio monitors, the KR-2C capsule at times rivaled the full lows of the Neumann, but it always brought out an airier high end by comparison, and it occasionally exaggerated cymbal sounds in the mixes.

As I did with with the Neumann KM 184 and for similar reasons, I tested the T.H.E. pair a second time for stereo matching, again with Stackpole's assistance. In this trial, Stackpole and I both found that the two mics in the stereo demo set provided by T.H.E. did not sound identically matched. Specifically, one mic had a midrange bump around 750 Hz. One effect of this bump was to make the acoustic guitar in the mix sound slightly more honky and nasal. Applying a 1.5 dB cut to that mic's channel with mixing board EQ eliminated most, but not all, of the discrepancy to my ear. Stackpole characterized the difference as "very subtle."

UP CLOSE AND PERSONAL

The first comment Hughes made after a brief listen to our string-duo tracks was, "They all sound so different!" That spontaneous response neatly sums up what may be the most complex and persistent issue I faced in preparing this article. Here are seven premium mics designed for critical recording, each carefully made and tested by some of the most respected names in the business. Each manufacturer does its job well and adheres to high technical standards.

And yet none of these mics, considered singly or in pairs, sounded the same in any of the tests I ran. As noted, some were similar in broad ranges of their tonality or in the way they picked up room sound and other nuances in stereo applications. As thorough as these tests may appear to be, they really just hint at how a given mic will perform over time in a specific recording environment. That is why I've done my best to keep my own comments broad and subjective.

When it comes to evaluating and choosing one favorite from a field of similar mics, stylistic factors may be just as important as the sonic qualities I've focused on here. For example, guitar or cello recorded by the relatively flat response of the Schoeps mics may be perfect for a mellow folk or classical recording. But for a salsa or modern-rock session, those same instruments and players may well sound best with a brighter mic such as the Pearl or T.H.E.

Likewise, factors such as recording format and media, preferred mic preamps, use of compression and reverb, and room acoustics can dramatically affect the suitability of a particular mic for your recording setup. And when you add in the all-important element of personal taste to this complex equation, terms like *best* or *favorite* become meaningless.

Nonetheless, being the opinionated reviewer that I am, I can't resist a purely personal assessment of these microphones. As far as basic timbre goes, I can confidently define the poles of this spectrum. On one end is the reliably bright Pearl TL 66, and at the other extreme, the always smooth Schoeps CMC 6/MK 4. I would rank the Neumann KM 184 next to the Schoeps for its consistent and ample low-end warmth. That leaves four other mics, each with its own flavor of presence boosting and bass-to-treble balance, in the middle of the spectrum.

My take on these entries is that the Josephson Series Four C42 and T.H.E. KA-04/KR-2C gravitated toward the bright and excited end of the spectrum. However, they almost always delivered pleasing tonality and impressive depth.

The AKG C 480 B/CK 61-ULS was less predictable and more difficult to categorize. More than once it demonstrated a basically flat response and closely resembled the mics at the warmer end of the spectrum. At other times, typically with distant placement, it had more bite and downplayed lower mids and bass.

The solidly centrist Audio-Technica AT4051a sounded consistently full and balanced in every test, never disturbing my sensibilities with too much treble or bass emphasis. This mic's dependable accuracy and relatively modest price makes it easy to recommend for almost any job around the studio.

Myles Boisen *is the senior mic wrangler, janitor, and group therapist at Guerrilla Recording and the Headless Buddha Mastering Lab in Oakland, California.*

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Frank Talk from Filipetti

PRODUCTION VALUES

Insights on recording from a busy and versatile producer and engineer.

By Maureen Droney

One of engineer and producer Frank Filipetti's singular talents is that he's comfortable behind the board for literally any kind of music. His credits range from James Taylor's *Hourglass*—an album recorded on a Yamaha O2R in a cottage on Martha's Vineyard that won Grammys for Best Engineered Album and Best Pop Album in 1998—to hit albums for contemporary bands such as Korn and Fuel. He has also worked with perennial stars such as Elton John, Barbra Streisand, and Rod Stewart. He's worked on Grammy- and Tony Award–winning Broadway cast albums, done live sessions with Luciano Pavarotti in Italy, and recorded Hole for the CD *Celebrity Skin*. Then there's that remote he did recording the songs of Bushmen tribes in Botswana's Kalahari Desert—well, you get the picture.

Filipetti is quite comfortable working on the cutting edge. As one of the first engineers to champion digital consoles, in the past few years he's become the "go-to" guy for numerous manufacturers seeking trusted ears to evaluate their gear. When they go to him, they have to be ready for honest assessment; he isn't shy about expressing his opinions, whether positive or negative. He doesn't fall for hype, and has been known to personally phone manufacturers to discuss what he thinks can be improved about certain products of theirs that he's purchased.

DAVE KING

Frank Talk from Filipetti

Filipetti, who lives in New York State, had a prior career as a moderately successful singer, songwriter, and drummer with his own publishing deal and recording contracts. But after nine years in the songwriting trenches, he reevaluated and made a left turn into engineering and production and never looked back.

These days he's busier than ever; pinning him down for this story took over two months. I finally caught up with him by phone one Sunday a few minutes before midnight. He was at a hotel not far from Manhattan's Right Track Recording (of which he is part owner,) trying to catch a few hours of sleep between the day-long setup he'd just completed and Monday's 7:30 a.m. session call. I tried to make our conversation quick, but, as is usual for Filipetti, once we were talking, he gave it his all.

What were you working on today?

Setting up for the cast album for the Broadway musical *The Boy from Oz*.

A big setup?

An orchestra: woodwinds, horns, strings, full percussion, and a rhythm section of keyboards, guitars, bass, and drums—along with 4 principals and a chorus of 18. Broadway shows are some of the toughest gigs because you have only one day to record everything. This one has 27 numbers, with a variety of musical formats, including small jazz band, big band, gospel, and Latin.

You've recorded plenty of orchestras. You must have a routine.

You can't really use that routine. With an orchestra, you generally don't have live vocals or a chorus and a rip-roaring rhythm section in the same room. Also, for a normal orchestral setup you'd have 20 to 25 strings to balance the horns and woodwinds. For Broadway shows the pits are incredibly small, and the costs are very high. So the string section has four to six violins and a couple of cellos. It's tough for such a small string section to compete with live drums, guitars, bass, woodwinds, and brass.

How do you work with that problem?

We're using "bugs" on the fiddles: little omni mics you fit on the bridge of the violins that give you an acceptable direct-sound-to-leakage ratio. Unless you're prepared to screen everybody else off, standard string miking brings up drums, horns, rhythm, and a lot of room. With the bug mics and some judicious equalization, you can make it work.

For this show we're using DPA bugs, but there are a lot of others. Schoeps makes a really nice one, and there are lots of homegrown varieties. Guys take little lavaliere omni directional mics and make clips for them. Some fit inside the hole, some sit on the bridge, some sit on the side of the instrument.

They give you a more direct string sound you can treat with artificial ambience?

Exactly.

Any other tricks?

You try to use microphone patterns as best you can, bearing in mind the size and the sound of the room and where the reflections are. It's all about what you screen, how to use mic polar patterns to maximize on-axis signals and minimize what's off-axis. Or at least to make sure the leakage off-axis doesn't interfere with what you're trying to capture.

Those are useful techniques for recording in general.

It crops up all the time. You want to minimize the bleed and make sure that whatever bleed you do get isn't ugly due to the frequency anomalies in the polar pattern. As you move toward the rear of a cardioid mic, moving away from the hot side of its polar pattern, it can exhibit an aberrant frequency response. In some circumstances, you might actually want to go to an omnidirectional mic, because you'd rather have good-sounding leakage than bad-sounding leakage. Generally the omni's polar pattern exhibits a much more uniform response all around the mic. The bottom line is to know the polar pattern *as well as* the frequency response of your mics.

So you're always thinking spatially as well as sonically.

Yes. The same principles can be used when recording a drum kit. In that case, the off-axis leakage isn't coming

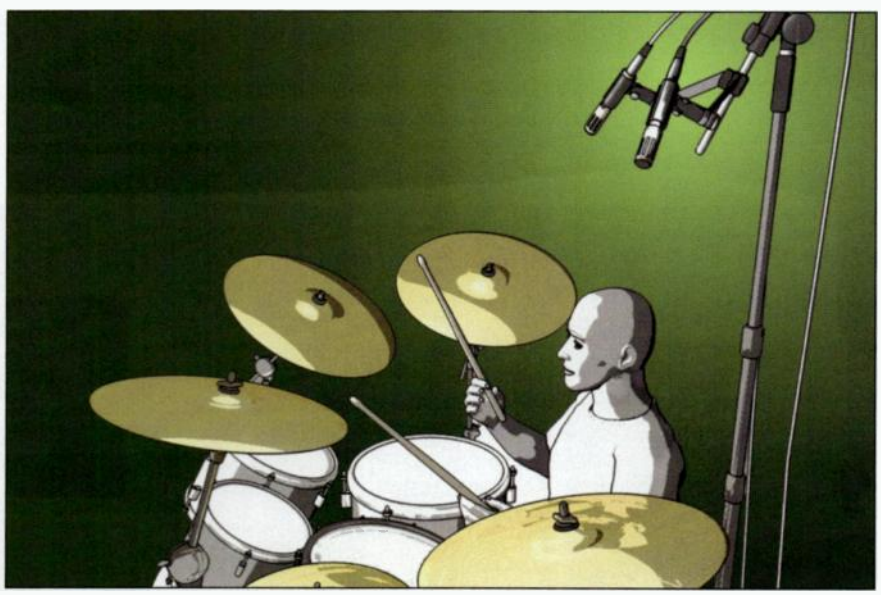

FIG. 1: When Filipetti mics a drum kit, he places his overhead mics behind the drummer, instead of the conventional method of putting them in front. He derives much of his snare and tom sound from the overheads and keeps them equidistant from the snare drum to keep it centered in the stereo image.

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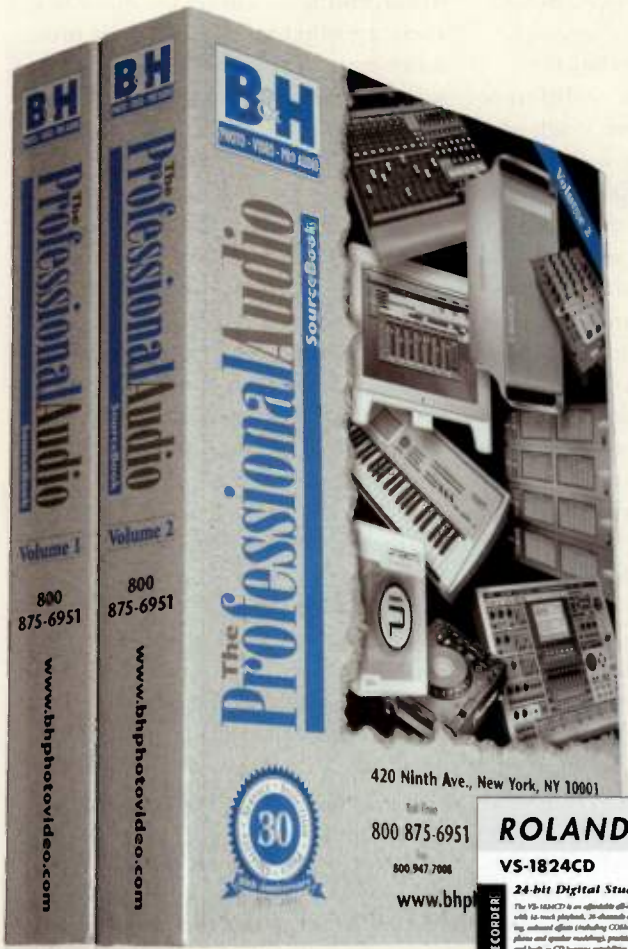

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from 10 or 20 feet away but rather from 6 or 7 inches away. Still, the techniques and application are very similar. Take the overhead mics—some engineers like to use them to capture only the cymbals. The snare, tom, and hi-hat mics provide the sound for their respective instruments in the final mix. As an ex-drummer, I've never been fond of that clean, dry sound. For me, much of the power of the snare and toms comes from the overhead mics. I place them (and their polar pattern) in such a way as to maximize the snare and tom leakage. Consequently, I mic from behind the kit rather than from in front (see **Fig. 1**). I make sure that the distance from the snare to both overhead capsules is identical. I don't want the final overhead balance, which is quite hot, to pull the snare from the center of the mix. Again, sometimes I use an omni if the snare and tom leakage isn't sounding quite right. Whereas some engineers will face the capsules over the cymbals, which gets 80 percent metal to 20 percent of the rest of the kit, I move them farther back over the drummer's head. I look for about 50 percent metal and 50 percent kit.

What about the other drum mics?

When miking a tom—in addition to just placing the mic where it will pick up the best sound on the tom itself—you have to focus on keeping the leakage of the cymbals out of the tom mics. Although the sound may seem okay on the recording date, by mix time, after you've added a fair amount of top-end EQ to your toms, you may find that your cymbals sound excruciatingly bright due to the cymbal leakage in the tom mics. Some people deal with this by gating, but that's never been my choice. Just take an extra couple of minutes on the recording date and play with some hypercardioid, figure-8, or even omni patterns before settling on your standard cardioid mainstay. For example, say you have to mic a rack tom with a ride cymbal slightly to the right of it. If you use a cardioid mic on the tom, you'll end up with a lot of rack tom *and* a lot of off-axis cymbal. Instead, you may want to use a hypercardioid mic or even a figure-8. That way, the cymbal coming in on the right actually hits the capsule at its null point, which is along the sides instead of the rear (see **Fig. 2**). You're not just miking to get the sound of the individual drum, you must always be aware that there are other tonalities in close proximity that, by the time you get to the mix, may cause you problems.

There's the sound of the mic, the polar pattern of the mic, and then there's phase.

Phase is tough on a drum kit because you have so many mics in close proximity. You can try to apply the "rule of threes," which is less a rule than a guideline; you want the direct microphone's capsule at least three times closer to the source than any other mic. The idea being, at that distance, phase cancellation (or more importantly, comb filtering) is less of a problem. With a drum kit, there's only so much you can do.

An example, please.

Say you have a mic on the snare. It's probably about six inches from where the actual attack of the drum is at the center of the head. As I said earlier, I don't like just the sound of that one mic; I like a bigger sound that comes from above the drummer, which I get from my overhead mics. But I don't want those overheads so close to the original source that they cause comb filtering on the snare. So I'll have the mics a couple of feet above the drummer's head. That adds weight to the snare drum as both capsules are temporally far enough apart to enhance rather than cancel frequencies out of it.

Also, your snare and hi-hat mics are always going to be in close proximity. There again, you do the best you can. Put your snare mic on the snare and try to maneuver it so that the hi-hat is facing the null point as much as possible. I usually try to put the hi-hat between its mic and the snare drum, using the cymbal as a kind of screen to the snare.

You recorded and mixed the upcoming Korn record *Take a Look in the Mirror* [Epic/Immortal, 2003] at lead singer Jonathan Davis's personal studio [see Fig. 3]. How did you mic David Silviera's kit?

I used an Audio-Technica AE2500—the dual capsule mic—for the bass drum;

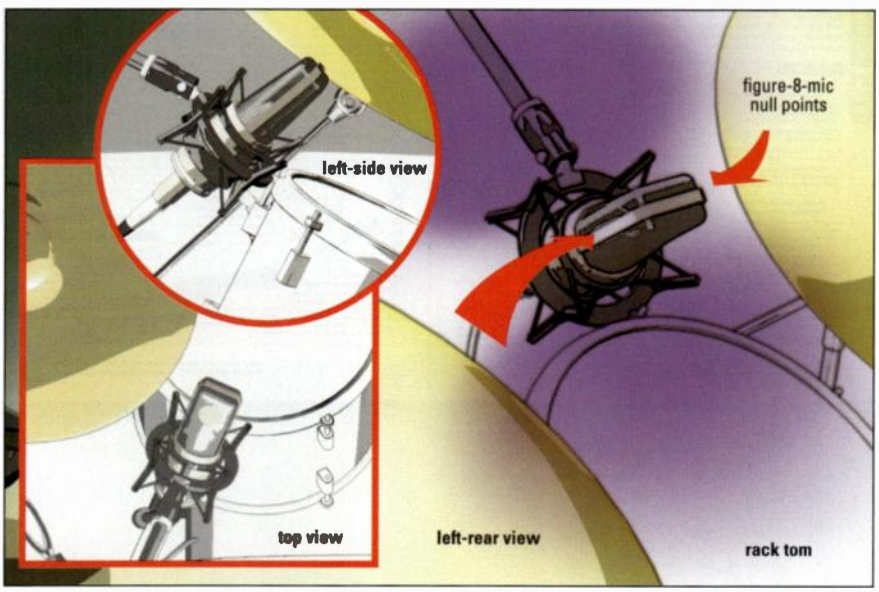

FIG. 2: Using microphone polar patterns to your advantage is an important part of recording, says Filipetti. In this case, a tom is miked with a figure-8 mic, the null point of which is aimed at the nearby ride cymbal to minimize leakage.

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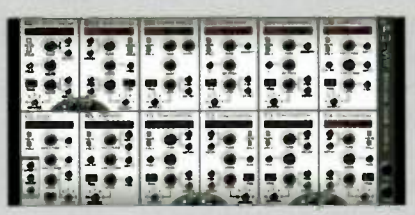

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Frank Talk from Filipetti

an old mic, an AKG D36; and a Neumann U 47 FET. I also had a tube Neumann U 47 in the room aimed at the bass drum. The drum had no front cover, and the AE2500 was slightly inside the rim of the drum right on the beater. The AKG D36 and 47 FET were slightly off-center and back a couple of inches, with the U 47 about two feet away for the extreme low end (see **Fig. 4**).

The snare mic was an Audio-Technica AE3000, with an Audio-Technica AE5100 underneath. The hi-hat was a Neumann KM 84, the overheads were Sanken CU44s, the ride cymbal was a Schoeps, and the toms were Sennheiser 421s, except for his "gong drum," which was mostly an Audio-Technica AT4047/SV. And then a couple of Neumann CMV 3s for the room.

Is using that many mics on a bass drum normal for you?

It's typical for a rock session. On a jazz date, I use two mics. On a Broadway date, I'd probably use just the AE2500.

For the bass drum on rock sessions, you want lots of impact and extreme amounts of low end—attack *and* depth. I use miking and phase to get the initial attack and the low impact that occurs microseconds later. I'll have one mic, possibly two, that are slightly out of phase with each other, and I'll add them together in a way that combs out 300 to 700 Hz. You can also do that with EQ, but I like it better using phase.

FIG. 3: Filipetti recorded Korn's new CD, *Take a Look in the Mirror*, at lead singer Jonathan Davis's personal studio.

The same with rock guitars. You can put one or two mics right on the cone, then add a couple more slightly out of phase with them by moving them back a few inches. It's taking that rule of threes and standing it on its head—putting two mics at slightly different distances from the source to get a comb-filtering effect. That sometimes works well on guitars; you move mics around until you get the sound you're looking for.

Do you listen on headphones while you're moving the mics?

No. While we set up, even before the drummer gets there, I have someone hitting the bass drum. I record a bit, then listen, and also look at the signal path in the waveforms on the Euphonix R1, which is what I record to. Then I play with the phase. It's a process of experimentation.

That's a rock date, when you don't necessarily want the bass drum to sound the way it does in the room. You're accentuating some of the fundamentals and removing some of the midbass harmonics. Unlike a jazz date, when it's important to get the natural sound of the drum—the ring of the head, the natural dynamic. On a jazz date, I don't generally play with phase. In the past, I'd use a dynamic and a condenser and try to get the capsules matched as closely as I could. With the AE2500, a very clever mic that has two capsules—one a dynamic, and the other a condenser, which are totally phase aligned—A-T has done that for me.

You prefer to use mic placement and phase manipulation to create your sound rather than filters and EQ?

Yes. I start with that. I want to get the best sound I can before I reach for the processing. I have no problem going –30 or +20 on an equalizer if I need to. But my instinct, training, and experience have led me to place a greater value on mic placement than on the equalizer and compressor.

Do you use the same techniques on a rock snare?

It varies. I use two to four mics. There's also the cross-stick issue; some guys are good at it and some aren't, and you have to mic with that in mind. I usually place a mic under the snare with the polarity reversed. Some snare drums have a lot of sound coming out of the air hole just off the center of the drum. Sometimes I mic that for impact.

You mentioned putting high end on toms for snap. What frequencies?

Somewhere in the 3 to 4 kHz range. You're looking for the sound of the stick on the head. That brings out the metal in the cymbals, which brings us back to what we were talking about earlier regarding minimizing cymbal leakage.

Do you compress drums?

Usually I don't compress drums on the record side, because you can't undo it. Rock kits often end up being compressed by default, anyway, due to the overall bus compression you're generally using in the mix. I like that better because it compresses more uniformly with the rest of the track. Sometimes I'll put the toms into a compressor because it brings up sustain and adds a certain type of bloom that's good. But compressing the bass drum or snare drum during recording can lose the crack of the initial attack. And once it's gone, it's gone.

Do you often record guitar amps in stereo?

No. For one thing, mono usually has more impact. With Korn, of course, there are two guitar players, so I'll place them left and right in the mix. If I'm looking for a particular effect, or for a solo, I'll mic two different cabinets, put them on separate tracks, and maybe spread them left and right. But that isn't stereo—it's two mono sources. For actual stereo miking, I'd probably use a coincident technique so that the sound would be captured in stereo but would still have a powerful, phase-coherent center.

What mics do you like on the guitar amps?

In the past, my main mics were a Shure SM57 and a [Neumann] U 67. I'd use both and match the capsule distance.

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Frank Talk from Filipetti

But since A-T came out with the AT4047[/SV], that's become my favorite guitar mic. I used it with Korn on three different cabinets. Each cab had a 4047 and one additional mic: a Royer ribbon, a 57, and a U 67. Generally, I put the mics close to the speaker cones.

That gives you a similar sound from the 4047s to help blend, and different sounds from the other mics for character.
Exactly.

Korn's bass player, Fieldy [Reginald Arvizu], has a very distinctive bass sound. How do you work with that?
Fieldy likes lots of click on the top and doesn't like mid bass. His bass sits above and below the guitars, which lets them stake out that middle ground.

Do you use direct and amp sounds on him?
Yes, but 80 to 90 percent amp. I use three or four mics: the Sanken CU44X and the AT4047 on one speaker, and an AKG D36 on the other. He also has a tweeter in his cabinet for high end, and I threw a 57 on that.

Is there a DI you like for bass?
The Vipre, by Groove Tubes—GT. It's a great mic pre and DI, especially for bass.

What did you use for Jonathan's vocals?
On the previous record, we used a Neumann M 49. On this one, we used a Sanken CU44X into a Tube-Tech MP 1A mic pre, then into the line-in of a Neve 1073 mic pre, for a little EQ tweak. Out of there, it went into an 1176 and a Massenberg EQ. From there into a dB [now Lavery Engineering] converter ahead of the R1.

How did that particular chain come about?
On the *Untouchables* album, I worked for a couple of days with Jonathan and [producer] Michael Beinhorn trying a variety of preamps and mics. Generally, I don't get that complicated. But working with Michael you get to experiment. We started with that setup but found that the Sanken worked better with Jonathan this time out. It was actually [producer] Peter Asher who introduced me to that mic.

More often than not, I'll start with a U 47 or a 269, put it through an 1176 and begin recording. Especially when I'm working with a new singer, I tend to go minimal. I want to know what their voice sounds like, and I don't like to do a lot of things until I'm confident I'm making the right decisions. I'd rather do something later than undo something later. We have so many tools now that we can use after the fact; a good, clean signal is the most important thing.

But Jonathan needs to hear a particular sound on his voice, and he likes to hear a finished product. We were trying to maximize the width of his voice—the low end and the high end—and to minimize a bit of his midrange. This chain worked well for that.

Speaking of doing things later, what about ambience on rock vocals?

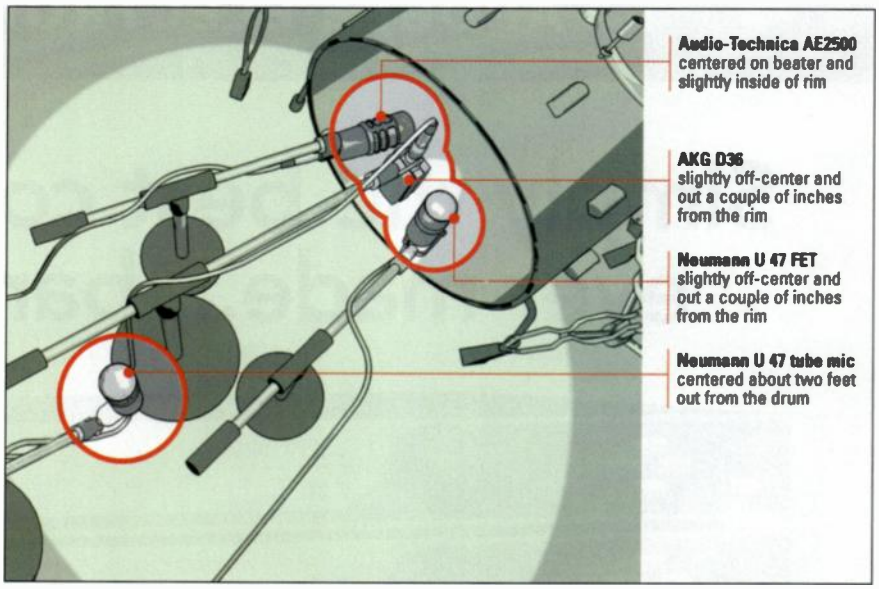

FIG. 4: Filipetti often uses multiple kick drum mics when recording rock sessions. This shows the 4-mic setup he used on the kick of Korn's drummer David Silviera.

FRANK FILIPETTI: SELECTED DISCOGRAPHY

Luciano Pavarotti, *Pavarotti and Friends: For War Child* (Polygram, 1996)
Barbra Streisand, *Higher Ground* (Columbia, 1997)
Billy Joel, *The Stranger* (Columbia, 1998); surround remix
Luciano Pavarotti, *Pavarotti and Friends: For the Children of Liberia* (Polygram, 1998)
James Taylor, *Hourglass* (Columbia Records, 1998); CD and 5.1-surround mixes
Luciano Pavarotti, *Pavarotti and Friends: For Guatemala and Kosovo* (Polygram, 1999)
Meatloaf, *Bat out of Hell* (Sony, 2000); SACD surround remix
Korn, *Untouchables* (Epic Records, 2002)
Rod Stewart, *It Had to Be You: The Great American Songbook* (J Records, 2002)
Fuel, *Natural Selection* (Epic Records, 2003)
Elton John, *One Night Only* (Universal, 2003); concert, CD, DVD, video
Rod Stewart, *As Time Goes By: The Great American Songbook, Volume III* (J Records, 2003)
Wicked, Original Broadway Cast Album (Decca Broadway, 2003)
Korn, *Take a Look in the Mirror* (Epic Records, 2004)
Paul Simon, *Graceland* (not yet released); surround remix

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Frank Talk from Filipetti

With a rock vocal—especially with a band like Korn, where you've got such a huge sound behind the singer—it's very important that the vocals are right in your face. Jonathan likes to have a lot of things on his voice, but he doesn't want it to sound like there's a lot. You'd actually be amazed at how much there is going on. We've got vocoders, delays, and choruses on most of the songs, but all are subtly applied to enhance the presence.

What's the most important piece of equipment that today's engineers need but don't have?

We desperately need a fail-safe, bullet-proof storage mechanism. That's something that's really going to bite us in the ass in the not-too-distant future. I started talking with some of the hardware manufacturers and OEMs, and found that a lot of people in the know are saying that if you let a hard disk sit idle for three to five years, there's a good chance it won't play back and that it will freeze up. You have to run it periodically. And I can guarantee you that in storage houses and at record companies, nobody's in there spinning these disks. People are putting all their stuff on Pro Tools hard disks, and then tucking them away. It's going to be horrifying if all this material goes down the drain.

Any new analog gear you like?

I love the new Tube-Tech multichannel compressor, the SMC 2B. It's my favorite new piece of analog gear. It has three bands—low, mid, and high—and you can set the crossover point so you can compress each band individually. I also like the new tunable JBL 6300 series speakers. I used them on the mix of the new Korn album and was very impressed. I also have to say that I'm very enamored with the TC-Helicon VoiceOne. It's digital, but it can do things you can't do with anything else.

The Grammy you won for *Hourglass* [Best Engineered Album—Non-Classical] was the first to be awarded for a project recorded in a home. What did you learn from that?

I certainly don't want to do that all the time! I wholeheartedly support the great commercial studios, and I want them to survive. They're really important to the art of music. But I learned that if you know what you're doing and you're in a good-sounding space, you can make a recording of very high quality for a relatively small amount of money. I used to get very uptight about situations I couldn't control, and it made me much more aware of the fact that you can just go with it. You can use the knowledge that you have to make it work.

Maureen Droney *is the Los Angeles editor for* Mix *magazine.* Mix Masters, *her book of interviews with top recording engineers, is available from Berklee Press.*

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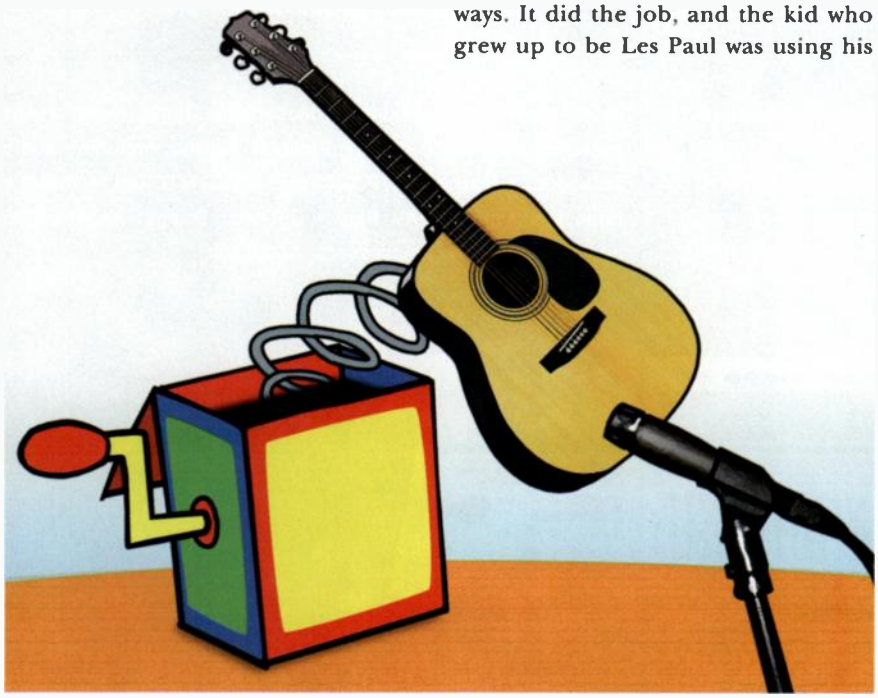
RECORDING MUSICIAN

Recording Outside the Box

Taking the less traveled road to great acoustic-guitar sound.

By Pat Kirtley

In the early 1930s, before the word "transducer" was ever heard, a 13-year-old named Lester Polfuss jammed the needle of a phonograph cartridge into the wooden top of his guitar in a desperate attempt to electrify it on the cheap. It wasn't the first means anyone had discovered to amplify or record the sound of a guitar, nor would it be the last, but it was a quirky alternative to the established ways. It did the job, and the kid who grew up to be Les Paul was using his creative mind to think outside the box.



MIKE CRUZ

For electric guitar, we've long accepted that tone is produced not just by the instrument, but by amps, speakers, and any number of electronic modifiers. But the conventional thinking about acoustic-guitar tone remains nailed in place by a purist viewpoint that says that all tonal flavor should come from the characteristics of the instrument itself and the hands of the player. Freed from this rigid thinking, acoustics can probe the same flexible tonal boundaries as electrics while still preserving their innate qualities. A fearless acoustic attitude is displayed by adventurous players such as Lindsey Buckingham, Steve Morse, and Adrian Legg, who record acoustic tone in uncommon ways.

The sizable body of literature about methods for recording acoustic guitar prescribes mic types, patterns, angles, and distances, which together constitute an industry standard for getting a balanced sound. However, by throwing off the constraints of accepted practice and using aggressively unorthodox techniques, you can create unique textures; get a lush, fat tone; and work around nasty audio problems. The following ideas have worked for others and may be helpful to you too.

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Prepare your ears for another onslaught of grinding guitar riffs, spooky soundscapes, darkish drum loops & nasty noises in general. *Mayhem 2* is the perfect remedy for any mix that's in danger of becoming too pretty. 100's of drum & noise loops, 300 drum sounds, more than 270 savage guitar & bass guitar riffs, eerie soundscapes & background noises, ugly sound FX & more!

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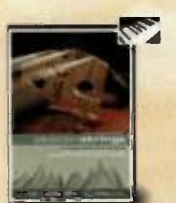

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NONTRADITIONAL MICS

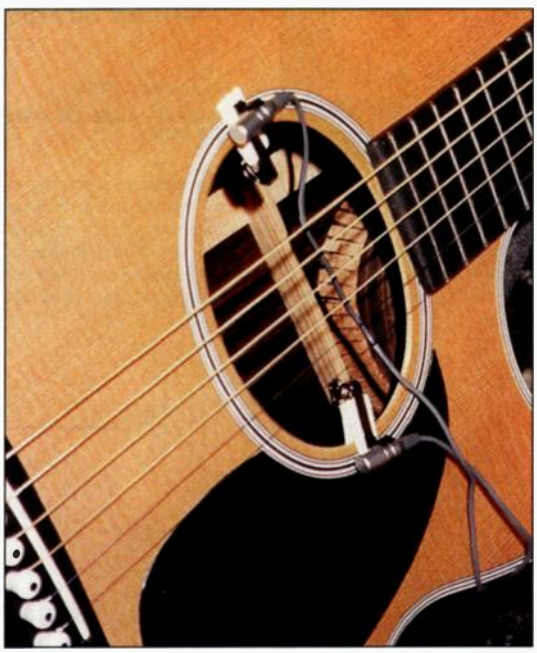
Most reference books recommend starting the quest for acoustic-guitar nirvana with a pair of small-capsule condenser mics. It's a tried-and-true formula, and with a good room and proper placement it yields predictable results.

Small-diaphragm condensers have a signature sound characterized by silky extended highs and natural extended lows—the perfect mic, you might think. But sometimes the extended range emphasizes certain sounds that would be best left out: the scratchiness of pick attack; string squeaks on the high end; and annoying, nonmusical rumbles on the low end. You can roll off the EQ, but a glovelike fit sometimes comes in the form of the garden-variety dynamic mic. Preconceived notions aside, most engineers are aware that many good recordings of acoustic guitar, especially in live-stage setups, are made with the ubiquitous Shure SM57 or similar mics.

The everyday cardioid dynamic doesn't have extended high-frequency response past 15 kHz or so, and it won't catch those 20 Hz lows the way that many condensers will. That can be a good thing, because the low end of a guitar doesn't go that far down anyway. The fundamental of the low E string is about 80 Hz. Dynamic mics have another benefit that fits the character of lightly played acoustic guitar sounds: a noise floor that's typically lower than those of all but the most expensive condensers.

If small-capsule condensers work well, what about really small condensers—the ones the size of a pinky fingertip, which are typically used as lavalieres for voice pickup. Some of those mics are very good, and their size lets you place them in unconventional positions. For years, guitar manufacturers and technicians have been installing these insect-size condensers inside of guitars, but the results are often disappointing. Why? One reason is that inside the box there are a huge number of standing-wave nodes—resonances that work together to vibrate the guitar's wooden surfaces but that, when picked up at almost any fixed location inside the chamber, sound quirky and unbalanced. Putting mics inside the guitar, especially for recording, is too problematic.

I like to use a stereo pair of minimics on the outside of the guitar, in a place that looks weird but sounds great. I put them on either side of the fretboard right against the body, temporarily held in place with double-sided tape. The tape helps to secure the mic and also isolates the mic body from direct vibrations of the wood. Another method that yields similar results is to clip the two minimics to the edges of the sound hole with felt padding under the clips for shock isolation (see **Fig. 1**). Listen to lots of mics on acoustic guitar—don't fear any of them.



PAT KIRTLEY

FIG. 1: A pair of lavaliere condenser microphones attached to the guitar's body with double-sided tape can provide surprisingly good results.

BASIC BLENDS

You can get effective guitar tones by combining standard mic configurations with the output of an onboard guitar pickup. For steel-string acoustics, use the classic condenser-mic pair augmented by a magnetic pickup placed in the sound hole. For years I kept a Bill Lawrence snap-in magnetic in a studio drawer just for that purpose. Most engineers and musicians consider such a pickup by itself unsuitable for recording—it has that characteristic electric-sounding one-dimensional tone that you love to hate in an acoustic. But consider its good qualities, such as rock-solid bass response and extra sustain.

Typically, you record the stereo mic pair on two tracks and the pickup on its own third track. During mixdown, you can refine the blend in a number of ways. One is to EQ away the upper mids and highs from the pickup and remove the bass frequencies from the mic pair at the same time, giving you the microphone equivalent of a two-way crossover. A typical panning scheme is mics full left and right, and pickup centered (see **Fig. 2**). The result? A "bigger than life" acoustic sound that's perfect for situations in which the guitar is prominent in the mix (see **Web Clip 1**). In this scenario, I generally use about 75 percent mic sound and 25 percent pickup sound. A useful bonus is that the onboard pickup is relatively isolated from outside acoustic sounds. If low-frequency rumble in the ambient environment is a problem, such a setup is part of the cure.

Standing this notion on its head, wonderfully spacious sounds materialize when you mic the body of an electric guitar. Even solidbody electrics radiate considerable high-frequency acoustic energy directly from their bodies—airy tones that are entirely lost through the guitar's normal pickups. Hollowbody archtops have an even bigger acoustic tone. The trick is to mic the guitar as if it were an acoustic, and then blend the mic sound with the usual mic-in-front-of-amp tone. No matter the type of guitar or the playing style being recorded, this idea is worth trying in your quest for capturing unique guitar sounds. It's amazingly easy, and there are no critical parameters—simply stick the mic out there and find a blend. Usually, just a little of this acoustic sound will add clarity and dimension.

STRANGE PAIRS

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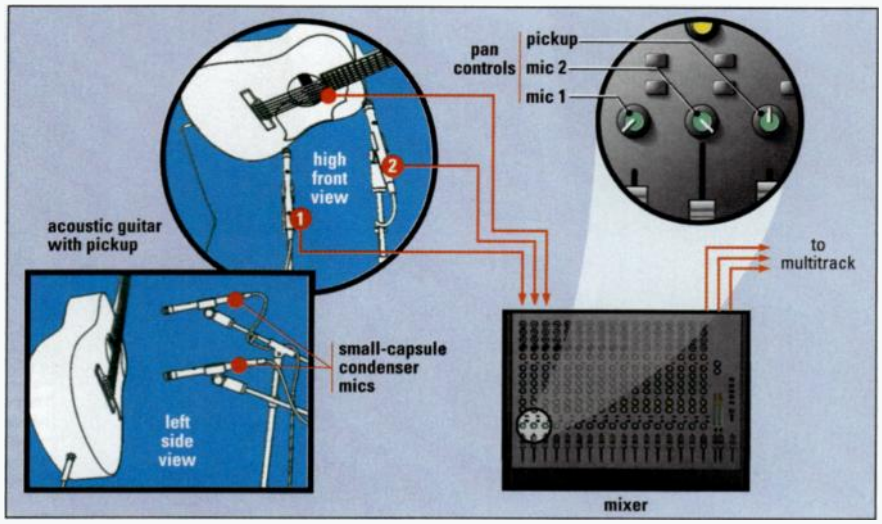

FIG. 2: Using a pair of condensers in conjunction with a magnetic pickup can give you a big sound. At mixdown, try panning the condensers right and left and putting the magnetic pickup's output up the middle.

with paperwork to confirm their common source of electrical DNA. It's a starting point for many classic techniques for recording acoustic guitars. The idea is to create a spacious, balanced stereo field. That's great for audiophile recordings of solo guitar, but a technique that offers far more possibilities is to use two completely different mics. For instance, try putting a small-cap condenser up by the neck and a dynamic down by the lower body. When the tracks are panned fully left and right, the stereo perspective is greatly exaggerated, and sometimes that is exactly what you want (see **Web Clip 2**).

I once got a luscious, fat sound with an AKG C 451 B on one side and a Shure ES615 omni "room-equalization" mic on the other. Similarly interesting results can be had with a large-diaphragm tube mic on one side and a dynamic cardioid "stage mic" on the other. A caveat: pay attention to potential phasing problems between extremely dissimilar mic pairs.

OMNI IS BEAUTIFUL

Directional microphones were developed for good reasons, the main one being focus, or the directional admittance of desired sounds and rejection of unwanted ones. Most of the microphones used in studio work are directional types—cardioid, supercardioid, or bidirectional (figure-8). The ability to reject unwanted sound comes at a cost, and one of the liabilities is the proximity effect: whenever a directional mic is placed close to the source, bass frequencies are overemphasized. That can be a significant problem when recording acoustic guitar, especially if you want to position a mic anywhere near the sound hole.

Omnidirectional mics are totally free of this bass-boosting aberration, and when you can use them, the open, airy quality can be striking. Most people don't think of omni mics when recording acoustic guitars because they pick up more reflected room sounds than the directional types do. However, since they are free of proximity bass-boost, you can place them closer to the instrument and maintain a good balance between guitar and room (see **Web Clip 3**).

MONO, ANYONE?

It's a habit to deploy a stereo mic pair to record acoustic guitar. But the stereo concept itself, as exemplified by a left-right speaker pair, is overrated in my opinion. At no time does the recordist have any control over the listener's speaker setup, which may be anything from an audiophile listening system to a cheap boom box to headphones. A stereo image that sounds crisp and spacious in the control room is sometimes blurred at end-user playback.

Summing a stereo pair to mono, especially with acoustic guitar as the subject material, can bring potential foibles to light. But there's a technique that often reveals sonic truth for all to experience: just use one mic. Doing so can be tricky, because it forces the recordist to find the precise distance and angle that will yield the best balance. But once the spot is found, a single-mic perspective can give a track a satisfying "already mixed" quality.

If you're afraid to commit, try this compromise: record a stereo pair of

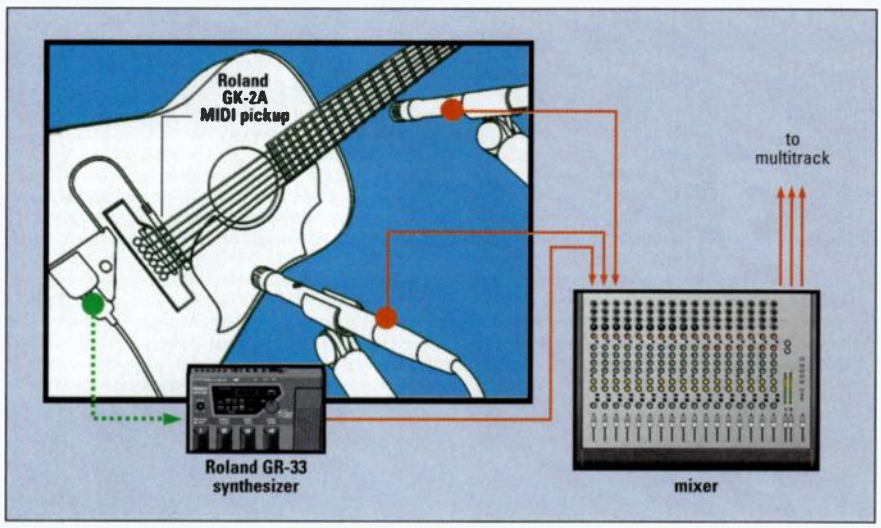

FIG. 3: The method shown here involves miking an acoustic guitar that's equipped with a synth pickup. The pickup drives a synth module whose output is routed separately to the multitrack, where it serves as an ethereal "ghost" track to supplement the main guitar sound in the mix.

"Two Publishers Fought Over My Music Because I Joined TAXI"

Bill Gordon – TAXI Member

You would think that after thirty-five years of writing, and recording my own music, I would have had some major commercial success. Sadly, that wasn't the case.

Then I joined TAXI.

As a result of my membership, I signed a deal with a boutique publisher that got me my first placements in TV and film.

Soon after that, a large-scale, international publisher contacted me about publishing some of my tunes.

The next thing you know, the two publishers were "fighting" over my music!

While nobody threw any actual punches, it was incredible having two great companies slugging it out over my songs. A musician's dream-come-true.

The end result? I'll tell you in a minute.

First, I want to tell you the ironic part – I almost didn't join TAXI. I thought it sounded too good to be true. Man, was I wrong!

TAXI's comprehensive feedback shows me where I'm strong, and where I'm not – a constant reality check. And getting tons of opportunities to pitch my music has helped me stay focused and very productive.

It's obvious to me that the people at TAXI *really* want me to succeed. And they do so much to make sure that I do. It feels like I've got a team of experts who are on "my side."

I can unequivocally say that the people at TAXI are the most informed, honest, honorable, helpful and loyal folks I have ever known in the music business. They never make a promise they can't deliver on, and they do it all at a reasonable price.

Then there's TAXI's private convention which is FREE to members and their guests. The Road Rally isn't just the icing – it's a whole other "cake." It gives me a deeper understanding of the music business, a bunch of great new contacts, and a jolt of inspiration.

TAXI also connected me with a Grammy™ winning producer who has worked with such legends as Jackson Browne, Michael McDonald, Luther Vandross, Maynard Ferguson, Freddy Hubbard, and Sarah Vaughn.

Two years later, he produced my album. The very same album that started the publisher slug-fest I told you about earlier. And how did it turn out?

I'm happy to tell you that *both* publishers signed some of my songs!

Thirty-five years as a working musician, a degree from Berklee, and all it took was a membership to TAXI to make me an "overnight success."

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tracks as usual, but add a third (mono) mic on its own track and compare the two versions. You might be converted to this underused method. If you need more convincing, listen to some late '50s to early '60s, one-mic, classical guitar recordings made before stereo was popular. Do you have only one high-quality condenser tube mic that you usually reserve for recording vocals? Try it on guitar.

GIVE AMPS A CHANCE

Recording directly from an acoustic guitar's built-in undersaddle pickup system rarely yields pleasant results. The sound tends to be strident and one-dimensional. Taking a cue from the late, legendary guitar master Chet Atkins, consider plugging into a guitar amp and miking it.

What might seem like the ultimate heresy to the purist camp was Atkins's preferred method when he recorded his acoustic-electric nylon guitars. And he didn't use one of those new-fangled clinically accurate "acoustic" amps either—his choice was more likely a tube amp made for electrics, typically his vintage Standel with a single 15-inch speaker. His mic of choice was either a Neumann U 87 or an old RCA 44DX ribbon.

MIDI BLENDS

Synthesis and MIDI are basic components of keyboard setups, but they are not as commonly associated with guitar, especially acoustic guitar. Indeed, many people would consider the electronic tones of a synthesizer to be the antithesis of the pure, organic sound of an acoustic. I've found, however, that you can create lots of cool new tonal possibilities by using the synthesizer's tonal palette to embellish that of the guitar when recording.

A simple way to bring synth capabilities to your acoustic guitar is to equip it with a synth pickup. For example, you can mount a Roland GK-2A (temporarily) on your guitar, enabling your instrument to drive a compatible synth and giving you the means to record simultaneous synthesizer and acoustic guitar parts.

A particularly effective approach is to set up the synth with a string patch or slow-attack sample that "follows" the guitar with orchestral lushness (see **Fig. 3** and **Web Clip 4**). Depending on how the mix is done, it can be a subtle background that blends into the reverb, or a

ACOUSTIC ALTERNATIVES

The various nonstandard methods for recording acoustic guitar discussed in this article are summarized below.

METHOD	MIC/TRANSDUCER	ADVANTAGES
Dynamic Mics	Cardioid dynamic mic pair	Narrower frequency range of dynamic mics means they pick up fewer scratches and squeaks in the high end and less rumble in the low end.
Guitar and Synth	Acoustic guitar with MIDI pickup, sound module, mic	Blend of straight acoustic sound with MIDI "ghost track" yields interesting tonalities.
Lavalieres Attached	Stereo pair of mini (lavaliere) condensers attached to neck or sound-hole edges	Mics move with the guitar so player movement isn't restricted. Bass- and treble-side perspectives are both picked up.
Mic the Transducer	Piezo transducer on guitar, dynamic or condenser mic for amp	Miked amp sound is often better than the pickup's direct sound.
Mismatched Mics	Two different mic types	Exaggerated stereo perspective.
Mono Miking	One high-quality condenser	Can sound good if correctly placed. Blends in track well.
Nashville Tuning (four lowest strings replaced and tuned an octave up)	Condenser (or dynamic) mic(s)	Lots of rich harmonics, no low-end muddiness.
Natural Chorusing (two detuned guitars overdubbed playing the exact same part)	Condenser (or dynamic) mic	Organic-sounding chorusing, varied by amount of detuning.
Omni Miking	Omni mic pair or single omni mic	Lets you position mic closer to sound hole without causing proximity effect. Good blend of direct and room sound. Natural tonal balance.
Pickup Blended In	Stereo pair of condensers, magnetic pickup	Big sound with miked signals panned left and right and pickup sound up the middle.
Pickup Through Amp	Pickup-equipped acoustic guitar, amp, mic for amp	Fattens and smooths tone, especially for acoustic-electric guitars.

thick sound that stands independent from the guitar tone, as if a keyboardist were playing along. Another cool trick is to feed the synth pad to the reverb send during the mix, but mute the direct sound of the synth track. The resulting reverb effect (which the ear interprets as originating from the guitar) can range from airy to otherworldly.

Many recording setups, computer based or otherwise, allow you to record MIDI data in sync with the audio tracks (if you're using the GK-2A, you'll need to plug into a GK-compatible synth or interface in order to get a MIDI out). By recording the MIDI data generated by the guitar synth, you create a trigger track that can drive synths in a number of ways during mixdown, widening the spectrum of possible tonalities and choices immensely.

SLICK TRICKS OF THE STARS

You needn't limit yourself to mic or electronic techniques when going for boldly different guitar sounds. A staple trick for years in country music has been to set up a guitar with the "Nashville Tuning," which involves raising the pitch of the lower four strings an octave by using individual strings from a set for 12-string guitar. The octave shift creates a sound with lots of stringy harmonics but no lows or low mids to muddy the mix.

Another trick invented in Music City is much less known. This one came from Chet Atkins (and was also used by Jerry Reed) in the days before digital effects units existed. Atkins recorded a track and then slightly detuned all the strings of the guitar and played identically on another track. The result is something like chorusing, but with its own unique character, and nothing about it sounds digital. By varying the amount of detuning (chromatic digital tuners make it easy to get an exact detuning), you can go from subtle and spacious to something pretty crazy. It works well for guitar solo parts too. The hardest part is playing exactly the same way twice in a row (never any trouble for Atkins and Reed).

BE RECKLESS

While there are few guitar recording challenges as exciting as the ones faced in the early days, the ideas in this column are guaranteed to add refreshing colors to lifeless guitar sounds. I know: I've had occasion to try them all, usually out of blind desperation. ("Must get killer guitar sound!") My reckless extremes were never as deeply driven as the young Les Paul's, but my desire for a unique sound is just as serious.

If nothing else, using these techniques will change your attitude about recording acoustic guitar and help avoid the same-old-same-old syndrome on your next recording. By all means, read the textbooks about recording acoustic guitar and learn the classic techniques, but next time you pull out the six-string, feel free to go a little nuts.

EM WEB CLIPS

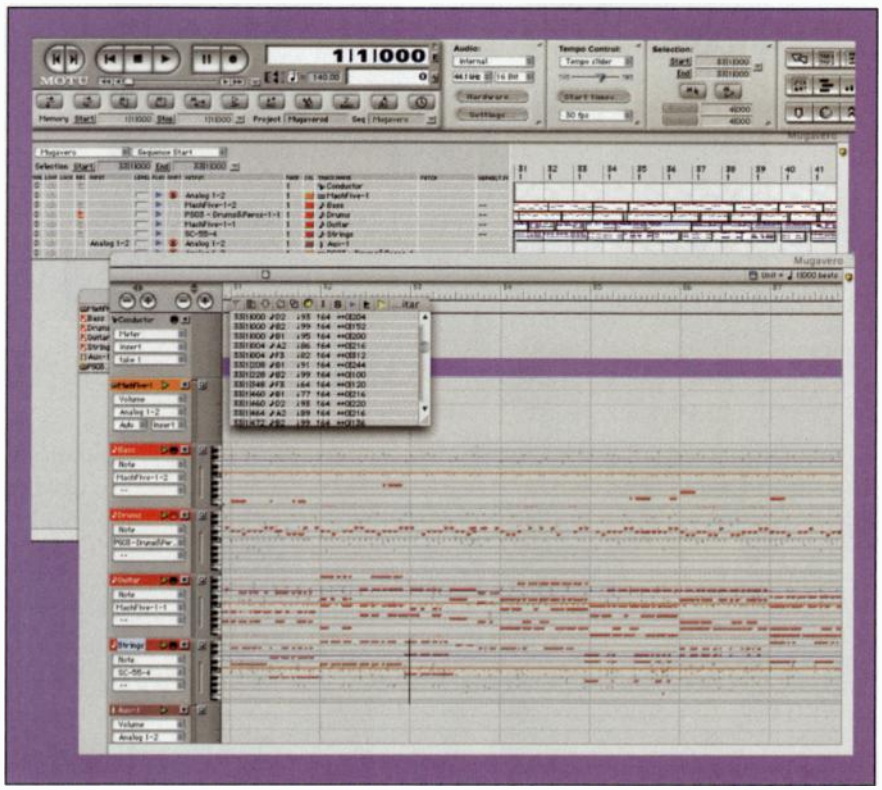
Pat Kirtley *is a guitarist, composer, and recording artist with an extensive background in pro audio. His latest CD release is* Brazilian Guitar *(MainString Music, 2002).*

Tricks for Tracks

The versatile music-editing features of MIDI sequencers.

By Jim Aikin

As any master carpenter will tell you, before you start any project you need to figure out what's the best tool for the job. Today's technology gives musicians a lot of options. If you've been making electronic music for only a few years, you may have jumped straight into digital audio with sampled loops and plug-in effects. Those are great tools, but there are times when plain old-fashioned MIDI sequencing will give you much more expressive control over your music.



If you've never used the MIDI features of your multitrack recorder, you've come to the right place. In this column, I'll explore how a computer (or a standalone workstation) records and plays MIDI data. I'll also discuss the main ways in which you can edit the data to clean up and personalize your recordings.

MIDI recordings are much easier to work with than digital audio recordings. That is because MIDI tracks contain only performance data, not actual sound. In order to listen to a MIDI track, you have to send it to a MIDI sound module, such as a synthesizer or sampler, which responds to the data by playing notes.

MIDI data is very efficient: a single Note On message, which occupies only a few bytes of computer memory, can trigger a sound that's many seconds in length. Even an old, slow computer can record and play dozens of MIDI tracks at once with perfect timing. The downside is that a Note On message contains no information about what the actual sound will be. The same message could

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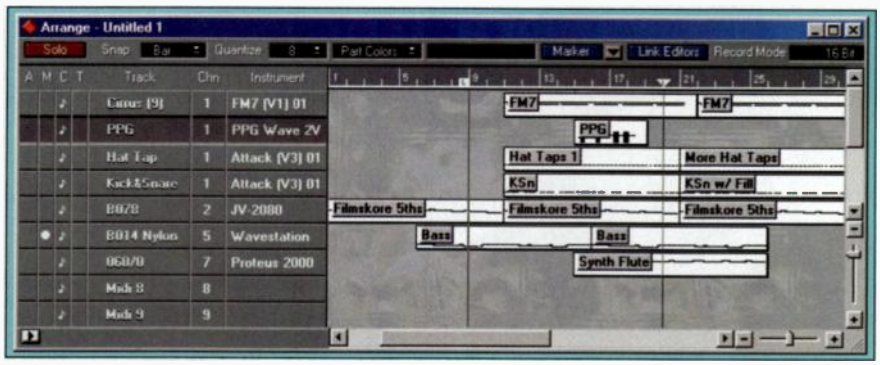

FIG. 1: The main track display in Steinberg's Cubase VST shows track names and other track parameters on the left and the recorded MIDI data extending across the area on the right. Several of the parts have been block-copied by Alt+click-dragging to lengthen the piece of music.

trigger a flute note or a sampled explosion. Or, if the synth on the receiving end isn't powered up, the Note On could result in no sound at all. It's up to you to make sure MIDI playback produces the desired sounds.

The following discussion applies to any MIDI sequencer, whether it's a computer program or a sequencer built in to a workstation keyboard. If you're not clear about the various types of MIDI messages, refer to "Square One: MIDI Me" in the July 2003 issue of **EM** (available online at www.emusician.com).

WHY USE MIDI?

If a sampled loop has exactly the sound you want, there's no need to mess with MIDI. MIDI is the tool of choice when you need to fine-tune the details of a performance. With a MIDI sequencer you can:

- Add filter sweeps and other expressive gestures to a line or just a single note using Control Change messages.
- Change the feel of a drum pattern, subtly or drastically, by changing the timing of MIDI events.
- Create your own beats by triggering individual percussion sounds.
- Try a different lead, bass, or electric-piano sound while keeping the performance (notes and rhythms) exactly the same.
- Change the tempo or transpose a whole song to a new key with absolutely no loss in audio quality.

Though you can use MIDI tracks and sampled (prerecorded) loops or other digital audio in the same piece of music, it's difficult to change the rhythm or tone color of a sampled loop by editing MIDI data. There are some ways to do it, but discussing them would take us well beyond the scope of this article.

THE BIG PICTURE

Most sequencers record MIDI data into tracks, which run horizontally across the computer screen in the track or arrangement window (see **Fig. 1**). Usually each track is assigned to a single MIDI channel. During playback, all of the track's data is transmitted on that channel, and any synth assigned to that channel will respond to the data by playing the notes recorded in the track. If you don't want to listen to a particular track, you can click on its mute button.

It's important not to confuse tracks (a sequencer feature) with channels (a MIDI feature). In most sequencers, it's easy to assign several tracks to transmit on the same channel. For instance, when building up a MIDI drum part, I often put the kick and snare on one track, the hi-hat part on a second track, and crash-cymbal hits on a third track. All of the tracks transmit on the same channel and are played by the same drum module. By doing this, I can copy and paste a cool hi-hat pattern without having to mess with the kick and snare.

Conversely, you can often find a track setting called Any, which allows a track to transmit data on more than one channel. You can then put data that has several different channel assignments into a single track. Usually there's no reason to do this, but most sequencers will allow it. With a few exceptions, each MIDI message has its own channel assignment. This channel

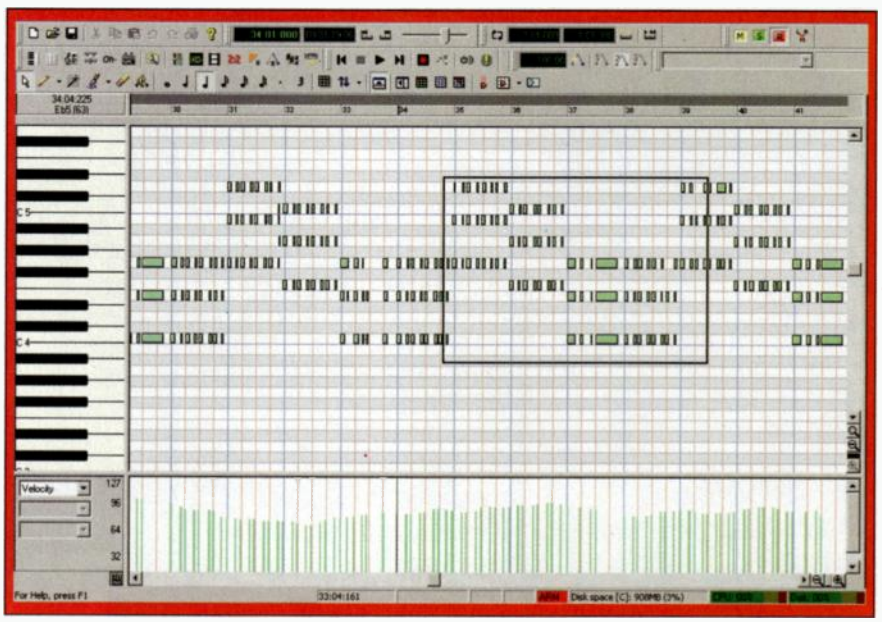

FIG. 2: In the piano-roll edit window in Cakewalk Sonar XL, the keyboard display on the left indicates which MIDI notes have been recorded, and the strip along the bottom shows the note Velocities. Here, the mouse is being dragged across a group of notes (the ones inside the rectangle) so they can be edited.

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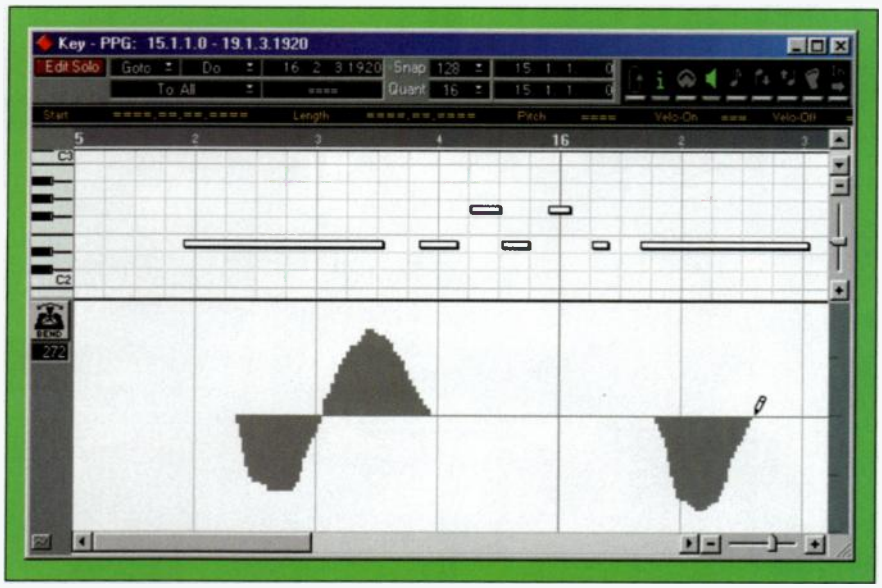

FIG. 3: The graphic controller-editing display in Cubase is positioned below the piano-roll display, making it easy to see how your controller moves relate to the notes. Here, Pitch-bend data is being edited with the pencil tool.

will be overridden by the track's channel assignment unless you set the track to Any.

The MIDI output channel is just one of the playback settings you can make for each track. The most important settings for tracks are listed below.

Transposition. By moving each track (except the drum tracks) up or down in half steps, you can play the music in a different key. By transposing a single track up or down by 12 half steps, you can hear the part in a different octave.

Volume. When playback starts, each track can send a MIDI Control Change 7 (Master Volume) message. A synth assigned to that channel will adjust its output volume based on this message. This is a quick way to set up a rough mix for a MIDI-based song arrangement.

Program Change. When you select a Program Change message (and, if need be, a Bank Select message) for the track, the sequencer will send out these messages on the track's channel just before starting playback. This ensures that each synthesizer will have the proper sound selected.

Velocity scaling. MIDI notes all have Key Velocity values, which can be anywhere from 0 through 127 (though the values you will see in your sequencer are from 1 through 127). In most synths, Velocity is used to make the sound of each note louder or softer. Adjusting all of the Velocity values for a track up or down is another quick way to bring the sound of a synth forward in the mix or reduce it so it blends in better.

Velocity scaling is a better choice than Master Volume when you've assigned several tracks to the same MIDI channel, because Master Volume is a global message that will be applied to all of the sounds played by the synth on a given channel. Velocity data is attached separately to each note. With Velocity scaling, for instance, you can boost the level of the hi-hat without affecting the kick and snare on the same channel, as long as the hi-hat is in a separate track from the kick and snare.

SEQUENCE EDITING

Changing the track playback parameters is a quick, easy way to change the sound of a MIDI sequence, but you can go much deeper. Most sequencers offer several types of editing utilities and editing environments, with which you can bend, shape, mangle, and torment the MIDI data. The editing facilities in workstation sequencers tend to be simpler than those in computer-based sequencers, but some workstation sequencers are quite powerful. Consult your owner's manual for details. The most important types of editing are:

Track window drag and drop. The track window will allow you to separate MIDI data into short segments (variously called *chunks*, *parts*, or *clips*). You can drag these segments to an earlier or later position in the composition, delete them, and copy them. After improvising a 32-bar bass line, for instance, you can separate out the 2-bar phrase you like best using a scissors tool, delete the rest of the take, and copy the 2-bar phrase over and over for the length of the song.

Piano-roll display. In the piano-roll window (see **Fig. 2**), MIDI notes are displayed graphically. The display looks rather like the rolls of paper used in

THE BIG RED BUTTON

Recording music in a MIDI sequencer works in much the same way as recording audio, except that you never have to worry about creating distortion by overloading the input. You put the sequencer in record mode, select ("arm") the track where you want to record, and then listen to the previously recorded tracks while playing your MIDI keyboard or other MIDI controller. Your performance is captured in the armed track.

As in audio recording, the MIDI keyboard's output has to be connected to the sequencer's input in order for anything to be recorded. Your sequencer probably supports such amenities as overdubbing, automatic punch-in and punch-out, and loop recording (in which you can keep trying the take over and over until you get it right). If your keyboard skills are minimal, you can take advantage of step entry, with which you record one note or chord at a time.

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player pianos, except that it runs horizontally rather than vertically. Time runs from left to right, so longer notes appear as longer lines in the display. Pitch runs vertically, with high MIDI notes at the top and low ones at the bottom.

In the piano-roll display, you'll be able to grab single notes or groups of notes with the mouse and drag them around as needed. Another mouse tool can be used for inserting new notes, and a third tool will shorten or lengthen existing notes.

Notation display. In the notation window, MIDI notes are displayed in the form of conventional rhythmic values on a staff. By printing out the music shown in this window, you can get a lead sheet or a full score for your music. Some musicians prefer to edit their MIDI tracks using the notation display because it's familiar. Notation displays have some drawbacks, however. For one thing, they can show notes only in conventional rhythmic values. A note that has an in-between length (duration) can't be notated accurately without using tuplets or dotted and tied rhythmic values that are hard to read and harder to edit.

Graphic controller editing. After recording a sweep or bend with your keyboard's modulation or pitch wheel/lever, you can edit the controller movement graphically using a pencil tool, as shown in Fig. 3. You can smooth out a move that's a little rough, or lower the peak if it's too high. As in the other editing environments, you can add new data using the pencil tool.

Event list. The tool for micromanaging your MIDI data is the event list. Here you'll see a list of all the data in a track, and you'll be able to edit each event (note, Control Change, Program Change, and so on) as desired. For instance, you can lower the Velocity of a single note that you hit a little too hard, shorten a long note, or change the pitch of a wrong note without disturbing its timing.

QUANTIZATION

Computers are capable of very precise timing. The timing of MIDI data in a sequencer is based on the sequencer's pulses per quarter note (ppqn) setting. If the sequencer is running at 480 ppqn, for instance, there will be exactly 480 different time locations available within the space of each quarter note. Every MIDI message in a track will be located at one of these time locations (they're also called clock ticks) and will be transmitted each time the sequencer's master clock reaches that time position during playback.

The sequencer's event list will display the time of each event in a format called bar:beat:clock. For instance, you might see the following:

007:03:120

This says that the MIDI event occurs in bar 7, beat 3, clock tick 120. If the sequencer is running at 480 ppqn, a value of 120 means that this event falls precisely on the second 16th note of the beat. Reading large clock-tick values sometimes requires a little head scratching, but it's not terribly difficult.

If the event described above is, say, a syncopated kick-drum note and if its time is shown as 007:03:129, that means it's a little late compared with an ideal 16th note. You can tidy up the rhythm by lowering the 129 to 120 in the event list, but if you've recorded hundreds or thousands of MIDI notes, editing their timing one note at a time is far too laborious. Fortunately, there's an easier way.

After selecting a group of MIDI notes, which could be an entire track, a single segment in the track window, or a group of notes in the piano-roll window, you can quantize them. When notes are quantized, their start times are moved so that they line up with an evenly spaced rhythmic grid. If your keyboard technique is a little sloppy, quantizing can make you sound like a virtuoso (see the sidebar "The Big Red Button" for a word on recording). Quantizing usually affects only notes, leaving Control Change and other MIDI messages untouched.

The downside of quantizing is that it can make the MIDI performance sound a little too perfect—a bit robotic, in fact. Most sequencers offer features that let you quantize your rhythms without overdoing it. You may be able to set a

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strength percentage, for instance, so that notes are not locked to the nearest beat, but moved only 50 or 75 percent of the distance between their starting position and the beat.

In swing/shuffle quantizing, notes on the offbeats are delayed by a certain amount. That gives the groove the looser feel that is often heard in jazz and blues styles. The shuffle amount may also be displayed as a percentage value (typically between 50 and 70 percent). It's important not to confuse the shuffle percentage with the strength percentage. The two features have very little in common, except that they both affect the timing of notes.

A more sophisticated option is groove quantizing, in which the timing of notes

▼

The very same message could trigger a flute note or a sampled explosion.

is corrected to a groove template, rather than to a fixed rhythmic grid. For instance, the groove template might push (advance) the timing of beat 2 in each bar just slightly, giving the backbeat a more aggressive flavor. Using groove templates is a great way to give your MIDI percussion a human feel.

TRACK OR TREAT

MIDI sequencing is a mature technology, so most sequencers, even the inexpensive ones, have dozens of sophisticated features. In this article, I've had room to discuss only a few of the most important types of MIDI edits. All you need is the sequencer owner's manual, a decent multitimbral MIDI synth, and a little patience, and you'll be well on your way to making great music.

Jim Aikin *writes about music technology for a variety of publications. He has been composing music with MIDI sequencers since 1985.*

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REVIEWS

MOTU

DIGITAL PERFORMER 4.11 (MAC OS X)

DP makes a smooth transition to OS X.

By Mike Levine

Digital Performer (DP), Mark of the Unicorn's flagship application, has long been one of the premier digital audio sequencers for the Mac. MOTU therefore generated considerable interest last April when it released version 4.0, which vaulted DP into the world of OS X. (MOTU still supports OS 9 with DP version 3.11.) As of this writing, DP is up to version 4.11, which supports Apple's Panther and Jaguar operating systems and CoreAudio, CoreMIDI, and Audio Units (AU) standards. It also has a greatly enhanced user interface and a slew of new features.

The program is now more elegant, more user-friendly, and more powerful than ever before. What's more, its AU support opens it up to a much wider world of virtual instruments and plug-ins.

For this review, I tested 4.11 on three different Mac models with two versions of OS X. My user experience was quite

FIG. 1: The look of Digital Performer 4 is similar to that of version 3, but a new menu structure makes the program even more user-friendly and intuitive.

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KURZWEIL

KME61 Rumour PC1x KSP8 PC2x Mangler K2600XS

positive, and making the transition from DP3 to DP4 was easy.

This review will focus on the most significant new features introduced from version 4.0 through version 4.11. For a look at some of Digital Performer's other features, please see the January 2002 **EM** review of DP 3.01, available at www.emusician.com.

FIRST THINGS FIRST

Getting started with DP4 is easy. Installation is smooth, and MOTU has included a helper app called FreeMIDI Converter that searches your hard drive for your FreeMIDI preferences and transfers your setup into an OS X Audio MIDI Setup configuration. And because DP4 supports CoreMIDI, FreeMIDI and OMS are no longer necessary (yippee!). If you need to tweak your configuration or set up a new one from scratch, Apple's Audio MIDI Setup utility makes it truly painless.

Once your MIDI setup is squared away, you'll need to specify your audio interface in DP's Configure Hardware Drivers dialog. DP's support of CoreAudio means that you can use it with any CoreAudio-compatible hardware. That opens the program up to a wider range of third-party hardware support. You can even use more than one audio interface if you'd like, because DP now supports multiple simultaneous audio interfaces.

The program can record either 16- or 24-bit audio, and the sampling rate depends on what your computer or recording hardware will allow. I tested DP using a MOTU 828mkII FireWire interface, mostly at 16-bit, 44.1 kHz resolution (the 828mkII can go up to 96 kHz), and the sound quality was uniformly excellent.

ON THE MENU

Version 4 of Digital Performer has a snazzy new logo and splash screen, but otherwise it remains graphically similar to DP3. (MOTU adopted an OS X–style look in version 3.) A closer look at the menu bar reveals some important changes, however: the menus have been significantly reworked, and the result is an even more intuitive user interface (see **Fig. 1**).

Gone are the Basics and Change menus, and in their place are the Studio, Setup, and Project menus. In previous DP upgrades, it seemed that new features were added to various menus because they needed a place to reside, not necessarily because it made sense to put them there. With DP4, all that's changed, and the menu structure is quite coherent.

The Setup menu mainly contains options for setting global preferences: Configure Audio System, Audio System Clock, Automation Setup, Time Formats, and so forth. The Studio menu lists many of the features you'll need during a session, such as the Performance window, the Audio Monitor, the MIDI Monitor, Click and Countoff Options, and the Clear All Clipping Indicators command, to name a few.

The Project menu is perhaps the most significant one in the new scheme. Here you'll find some of DP4's handiest improvements. In older versions of the program, you created tracks from a mini-menu located at the top left of the Tracks Overview. No more. At the top of DP4's Project menu you'll find the Add Track command (see **Fig. 2**), which lets you choose MIDI, Mono Audio, Stereo Audio, Aux, Master Fader, Surround, or Instrument tracks (which I will discuss in a moment).

The Project menu also gives you access to the various editing windows, the Soundbites window, the Clippings window, and more. Even if you end up relying on the user-customizable key commands rather than the menus once you get to know the program, it's still nice to know that everything is where you expect it to be.

VIRTUALLY YOURS

The new Instrument Track feature facilitates virtual-instrument-plug-in setup. All correctly installed virtual instruments appear in the Instrument Track menu; select one, and a new Instrument track appears in the Tracks window, already configured for that particular virtual instrument. Create a MIDI track and select that instrument (and a MIDI channel) as an Output, and you're ready to rock (see **Fig. 3**).

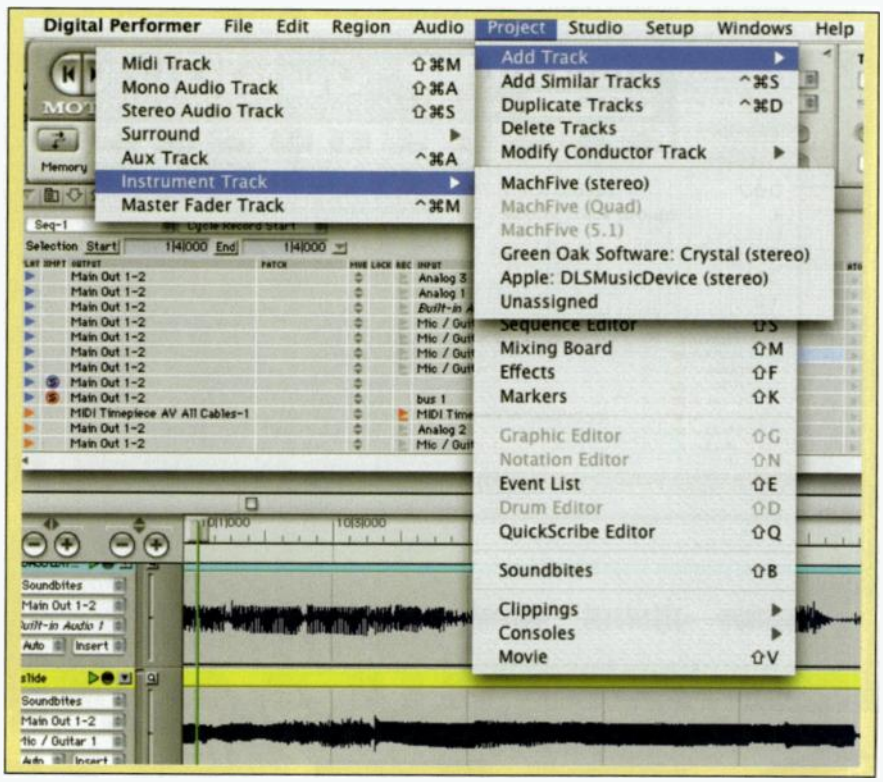

FIG. 2: One of DP4's most useful features is the Project Menu's Instrument Track option, which allows you to quickly set up virtual instruments.

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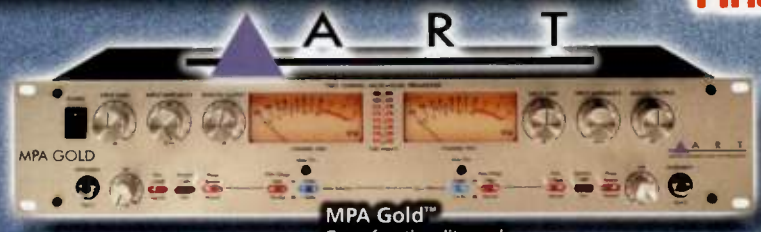

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WRH

I tried out the Instrument Track feature using MOTU's MachFive sampler and Green Oak Software's freeware AU soft-synth Crystal. I was able to set up both instruments quickly and easily.

Digital Performer 4's support of the AU standard means that users aren't limited to instruments that run under MOTU Audio System (MAS). Now any AU instrument or effects plug-in is fair game. And if you use a converter such as FXpansion's VST to AudioUnit Adapter, you can make all your VST plug-ins and instruments DP4-compatible, giving you even more options.

DP's implementation of Audio Units is pretty smooth. After you install a new AU plug-in and start DP, a screen pops up to tell you that the program is checking the plug-in and then whether it successfully loaded. DP recognized almost all the AU plug-ins I attempted to install. After installation, your AU plug-ins reside in the plug-in menu alongside the MAS plug-ins.

DP4 also lets you manage your plug-ins more efficiently through a new feature that warns you when you open a file that contains assignments for plug-ins that are no longer in your system (which could happen if you moved the file to another computer, for instance). If you have missing plug-ins, a dialog box appears and offers you the option of clearing the plug-in assignments.

If you choose not to clear them, the names of the missing plug-ins show up in the inserts of their appropriate tracks in the mixing board, italicized and in parentheses to show that they're not active. When the file is reopened in a system where the missing plug-ins are installed, the assignments become active again. That means you can work on a file, open it on another computer that doesn't have all (or any) of your plug-ins, and then bring the file back to your original computer with your original plug-in assignments intact. If you have more than one computer, or if you work on your files in more than one studio, this feature could be quite handy.

CHILL OUT

Another significant addition to DP4 is the Audio menu's Freeze Selected Tracks command. This feature takes whatever tracks (or single track or portion thereof) you've selected and temporarily writes them to disk, including any plug-ins you might have applied.

FIG. 3: Digital Performer 4.11's support of Apple's Audio Units makes it possible to access a wide range of virtual instruments and plug-ins.

Minimum System Requirements

Digital Performer 4.11

G3 (dual-processor G4 recommended); 128 MB RAM for MIDI sequencing; 256 MB RAM for audio recording (512 MB recommended); Mac OS X 10.2

It then mutes the voices of the source tracks, freeing up most of the CPU resources that those tracks used. Later, when you want to unfreeze the tracks, you simply select them, press the Shift key, and choose Unfreeze Selected Tracks from the Audio menu.

Although you could achieve similar results using the Bounce to Disk command, the Freeze feature is more convenient in many situations. It's also quite useful for freezing Instrument tracks and Aux tracks, giving you an easy way to record the audio from those sources.

The only downside of the Freeze function is that it writes the selected track to disk in real time. (According to MOTU, that is necessary to ensure that virtual instruments—triggered by MIDI—play back accurately.) So if you select a track that lasts three minutes, it will take three minutes to freeze it. The Bounce to Disk function works considerably faster.

WHAT'S UP, DOCS?

It might sound boring at first blush, but the new Document Templates feature is actually an exciting addition to DP. Whereas in prior versions you were limited to saving a single customized New Template, you can now save any number of templates to choose from when you create a new project. If you handle several types of projects in your studio, having custom templates could be a real time-saver.

Another improvement is the list of Recent Files that's available from the File menu. It shows you the last ten files you've had open, which can reduce the need to search your hard disk for that song you were working on.

Although the Mixing Board window hasn't changed significantly, it does have a new Input/Output display. This

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FIG. 4: The Mixing Board window now has pull-down input- and output-assignment menus (located just below the faders) for each track.

consists of two oval-shaped windows at the bottom of each channel strip: one showing input assignment, and the other output assignment (see **Fig. 4**). Clicking on either brings up a menu you can use to change a channel's assignments.

MAGIC MARKERS

If you do much work scoring to picture, or even if you just use DP's Marker feature a lot, you'll appreciate two new marker-related features. First, there's Shift to Marker. This lets you select a note or Soundbite and shift it directly to a particular marker. The marker names are displayed in a pull-down menu that appears in the Shift dialog box when the Shift to Marker feature is selected.

Second is the Snap to Markers option. When checked (in the Graphic Editor or Sequence Window mini-menu) dragging a Soundbite or MIDI note close to any marker will cause the event to snap to that marker. If you're trying to drag, say, an audio file of a sound effect to a particular SMPTE frame of a video, the snapping feature makes it easier.

Even better, you can drag Soundbites from the Soundbites window so that they snap to markers in the Sequence Editor. Imagine this scenario: You have a bunch of sound effects that you want to audition at a certain hit point on a video. You load them into the Soundbites window, and then audition them by dragging them into the Sequence Editor and making them snap to the appropriate marker. Very efficient.

In my tests of the Snap to Markers feature, it worked much more easily with audio files than with MIDI notes. For the latter, I had to zoom in really close and drag the note almost right onto the marker to make it snap correctly.

QUILL IN HAND

Users of the QuickScribe Editor (DP's notation feature) will be happy to know that it has been beefed up significantly. For one thing, the transcription engine has been enhanced, and—according to MOTU—it's now even better at accurately transcribing unquantized MIDI data. I found that it worked quite well in that regard.

In one test I chose a slow tempo and recorded passages (through MIDI) from my controller keyboard that included a variety of rhythmic values, quarter-notes, eighth-notes, eighth-note triplets, 16th notes, and 32nd notes among them. QuickScribe's transcription was almost totally accurate, and there were very few of the puzzling and unreadable rhythmic interpretations that can plague notation software. It was quite impressive.

MOTU has greatly enhanced QuickScribe's ability to configure scores for printing (see **Fig. 5**). The Arrange Palette gives you the ability to add arrangement necessities such as repeats, codas, and D.C. and D.S. symbols. Arranged Score mode also gives you the option of hiding measures or groups of measures that are no longer necessary in the printed score once the repeats and other arrangement shortcuts have been entered. However, hidden measures will still play back accurately in the sequence.

MOTU has made the QuickScribe Editor so friendly that even notation neophytes (and notation phobics) will be tempted to use it periodically. Users who are fluent in written music will find it much more powerful.

MIDI GOODIES

As an OS X application that supports CoreMIDI, DP offers access to the Apple Software Synthesizer, a pretty decent GM sound set. You can choose from five levels of sound quality (Min, Low, Medium, High, and Max), which use differing amounts of CPU power. DP also supports Interapplication MIDI, which lets you publish unlimited inputs and outputs that DP can use to communicate with other MIDI software.

DP4 offers full support of OS X's MIDI-device patch lists and has ported over all of its FreeMIDI patch lists. This lets you see patch names in the patch lists of supported devices. Not all devices are supported, however, so you might have to manually configure the patch lists for some of your MIDI instruments.

REASONS TO BE CHEERFUL

Users of Propellerhead's Reason will be happy to know that DP4 supports the ReWire 2.0 spec. One of the main benefits is that you can now use Reason (or another ReWire-equipped virtual instrument) as a virtual sound

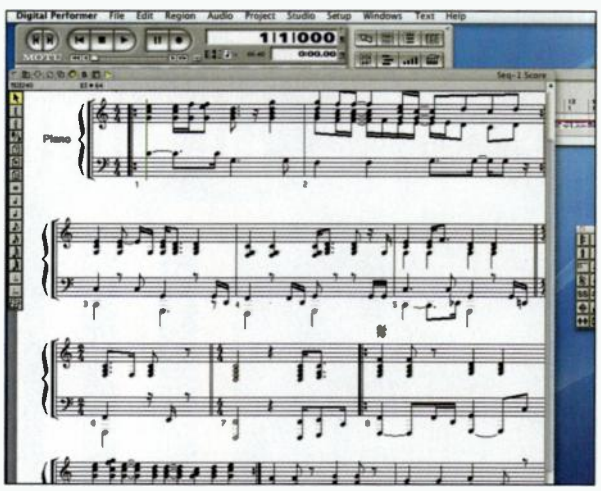

FIG. 5: The QuickScribe Editor's Arranged Score mode lets you to prepare print-ready scores that include arrangement symbols such as repeat signs and codas.

module that will respond to MIDI sequences in Performer.

Owners of Pro Tools hardware will be pleased to know that starting with version 4.1, DP can be configured to run under the Digidesign Audio Engine (DAE) in addition to running native under MAS. That means you can use DP with Pro Tools|HD Accel and Mix hardware, and that DP can support TDM plug-ins.

PERFORMER'S PERFORMANCE

Clearly, Digital Performer 4.11's features have been significantly enhanced. But for a product of this type, the proof of the pudding is in how it performs in the real world, and the new DP shines in that respect.

When I began the testing for this review, I was using a single-processor

▼

Now any AU instrument or effects plug-in is fair game.

G4/450 MHz running Apple's Jaguar (10.2.6) system software. On that machine, OS X ran a lot slower than OS 9.2 had. As a result, DP4 ran slower than DP3 had run under OS 9. But despite its pokiness, DP4 was still usable. I ran up against the limits of my processor when using a lot of plug-ins, but that was no different than when I was using DP3.

Later I used Digital Performer 4.11 on a G4/733 MHz running Jaguar 10.2.8 and on a dual G5/2 GHz running Panther 10.3.1. DP was much peppier on the faster G4. Naturally, it ran best on the G5. That's where I was able to get the most from the program, running multiple virtual instruments and a ton of plug-ins.

I did a few large projects with DP4 during the time I was testing it for this review, and it was quite stable. It did "unexpectedly quit" (OS X's euphemism for crashing) on me occasionally, but mainly as DP launched. (I've

noticed that this happens periodically with many applications in OS X.) As a result, I rarely lost any data. I didn't even have to restart the computer, because when a program crashes in OS X, it doesn't usually take the computer down with it, which makes life a whole lot easier.

FROM 9 TO X

Over the years, digital audio sequencers have evolved into massive programs that are expected to handle MIDI, audio, notation, and an incredibly wide range of tasks. Accordingly, the DP4 manual (which is quite informative) is almost 1,000 pages long. Considering the program's massive feature set, MOTU has done a yeoman's job of porting its flagship application to OS X.

Although their feature sets are not completely identical, the major Mac sequencers have always been pretty similar in their overall capabilities.

PRODUCT SUMMARY

Mark of the Unicorn
Digital Performer 4.11
digital audio sequencer
$795

FEATURES	4.5
EASE OF USE	4.0
DOCUMENTATION	4.0
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: CoreAudio, CoreMIDI, Audio Units, and ReWire 2.0 support. Compatible with wide range of audio hardware. Works with Digidesign Pro Tools hardware. User-friendly menus and interface. FreeMIDI converter app. Freeze Selected Tracks feature conserves CPU resources. QuickScribe offers arrangement icons and good transcription quality.

CONS: Freeze Selected Tracks feature renders tracks to disk in real time (slowly). MIDI notes don't snap as easily with Snap to Marker feature as Soundbites do.

Manufacturer
Mark of the Unicorn (MOTU)
tel. (617) 576-2760
e-mail info@motu.com
Web www.motu.com

The biggest differences are in the user interface. Digital Performer's interface has always been relatively user-friendly, and it's better than ever in version 4. The new menu structure, the Project menu with its Instrument tracks, and countless improvements and tweaks by MOTU's engineers have made the program more efficient and more intuitive.

For me, the move from OS 9 to OS X was much more jarring than the transition from DP3 to DP4. Even with all the improvements in DP4, it still feels comfortably like the same program. So if you're a DP user who's been considering taking the plunge into OS X (and you have a relatively fast Mac), the time is right. Digital Performer 4—with its expanded functionality and familiar interface—will help make your transition successful and productive.

Mike Levine *is a senior editor at* **EM.**

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1st through 6th chair French Horn Section

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2nd & 3rd chair Trumpet Section
1st through 3rd chair Trumpet Section

Trombones
1st chair Bass Trombone
1st & 2nd chair Tenor Trombones
1st through 3rd chair Trombone Section

Tuba
C Tuba
Eb Tuba

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Melodic Legato
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Half Step Trills
Whole Step Trills
Staccato
Double Tongue "ta"
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Muted Staccato
Muted Flutter Tones
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MINI-ME AND MINI-DAC

A dynamic duo for tracking and listening anywhere.

By Nick Peck

Apogee has released a pair of high-quality portable digital interfaces that together form a complete high-end I/O setup that's great for recording and playback on the go. The Mini-Me is a stereo mic/line/instrument preamp with a 24-bit, 96 kHz A/D converter. The Mini-DAC is a 24-bit, 192 kHz D/A converter.

Both devices offer USB connectivity. However, the Mini-DAC was designed to interface conveniently with the Mini-Me, so it takes just one USB connection to record to your computer through your Mini-Me while simultaneously monitoring with the Mini-DAC.

FROM A TO D

As an A/D converter, the Mini-Me offers 44.1, 48, 88.2, and 96 kHz sampling rates and supports 16-, 20-, and 24-bit word lengths. All three word lengths are available at 44.1 and 48 kHz, but the 88.1 and 96 kHz sampling rates operate at 24 bits only. Apogee's UV22HR dithering algorithm is automatically applied during 16- and 20-bit operation.

The Mini-Me is made of sturdy, lightweight aluminum, with the controls laid out somewhat snugly on the front panel (see **Fig. 1**). All connections are on the rear. When the left and right input-level controls are set fully counterclockwise, the XLR inputs on the back operate at line level instead of mic level. A pair of small, recessed screws on the front panel, which require a jeweler's screwdriver to access, are used to calibrate the incoming line level. I found this arrangement inconvenient for field recording: I'd prefer a simple mic/line input switch instead, allowing the knobs to control the level of either. When a ¼-inch instrument input is connected, the knobs automatically function as instrument-level controls. Two four-LED ladder-style input meters reside between the level controls

The Mini-Me's front-panel power switch doubles as a +48V phantom-power selector. Holding the switch to the far right for a couple of seconds activates or defeats phantom power. The 12-position Sample Rate knob selects between the various combinations of sampling rates and bit depths. The Mix control sets the balance between the direct signal and the signal at the USB input. The Mix control doubles as a push button that switches between mono and stereo operation when one mic input is used. The front panel also includes a headphone level control, a switch to select dynamics processing (off, limiting only, or compression and limiting), and a switch to select one of three compression curves.

On the rear panel, the Mini-Me has a pair of Neutrik combo jacks. Mic and line signals are received through the XLR jacks, and the ¼-inch jack is for high-impedance signals only. The rear panel also has an ⅛-inch stereo headphone jack, a USB port, S/PDIF and AES/EBU digital outputs, and a power jack for the external power supply or an optional battery. For field recording, a lead-acid battery (available from Eco-Charge, Inc.) is the only way to go.

PRODUCT SUMMARY

Apogee Electronics
Mini-Me
mic preamp and ADC
$1,495 ($1,295 without USB)

FEATURES	4.0
EASE OF USE	3.5
AUDIO QUALITY	4.5
VALUE	2.5

RATING PRODUCTS FROM 1 TO 5

PROS: Great sound quality. Built-in compressor and limiter. Highly portable. Rugged.

CONS: No word-clock in. Must be the master clock source. Recording through USB limited to 44.1 or 48 kHz. Line-input level can only be adjusted from tiny calibration screws. Compression algorithms are somewhat heavy. No status indicator for mono/stereo headphone switch.

Manufacturer
Apogee Electronics Corp.
tel. (310) 915-1000
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FIG. 1: With its front-panel Sample Rate control, the Mini-Me makes it easy to choose the sampling rate and word-length combination that fits your recording needs.

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As a bonus, the Mini-Me can output digital black from the AES/EBU and S/PDIF ports at all four sampling rates when the unit is not sending digital audio information. This allows the Mini-Me to be a high-quality, low-jitter master clock-source for a digital studio. However, the unit does not have a standard BNC-style connector for dedicated word-clock output.

Mini-Me Specifications

Analog Inputs	(2) XLR/¼" TS combo jacks
Analog Outputs	(1) ¼" stereo headphone
Digital Outputs	(1) XLR AES/EBU; (1) S/PDIF coaxial; (1) USB
Sampling Rates	44.1, 48, 88.2, 96 kHz (USB max. rate 48 kHz)
Mic Amp Gain Range	12–65 dB
Word Lengths	16-, 20-, and 24-bit (24-bit only at 88.2 and 96 kHz)
Dynamic Range	105 dBA
THD+N	–94 dB
Frequency Response	20 Hz–20 kHz (±0.2 dB @ 44.1 kHz Fs)
Power Source	6–16 VDC (adapter included)
Dimensions	5.4" (W) × 1.5" (H) × 9.8" (D)
Weight	2 lb.

SIGNAL TAMERS

The Mini-Me features defeatable limiting and compression circuits to eliminate unwanted digital overs. Apogee's Soft Limit limiter begins rounding peaks at about –4 dBFs, which results in fairly transparent limiting that leaves most of your recording untouched. The Push-It compressor circuit is designed for more aggressive gain control, audibly squashing the sound and allowing an overall hotter signal to be digitized. The unit offers three preselected gain-reduction curves, but no adjustable compression parameters.

I liked the Soft Limit feature quite a bit because it let me record sharp, transient sounds—such as metal clanks—without worrying too much about overloading the circuit. It worked well on a number of sources for which the maximum dynamic level couldn't be predicted, such as concerts and loud sound effects.

I didn't care much for the Push-it compressor, however. Whether recording vocals, piano, percussion, or airplanes, the Mini-Me's compression algorithms added a hard, flat sound that didn't complement or enhance the recordings. My tendency was to stick with the Soft Limit feature, record at a slightly lower level, and then apply plug-in or outboard compression later if needed.

SURFIN' USB

The Mini-Me can act as a front end to any digital recording device through its AES/EBU and S/PDIF ports, but it also includes a USB port for connecting directly to any USB-equipped computer. Apogee has developed ASIO drivers for Windows 98SE, ME, 2000, and XP and for Mac OS 9 and OS X. The CoreAudio drivers in OS X support the Mini-Me as well. As a result, the Mini-Me works with most current DAW applications running on a laptop or desktop machine. The bandwidth of the USB protocol limits the maximum audio resolution to 24-bit, 48 kHz. However, the Mini-Me's AES/EBU and S/PDIF outputs continue to send audio at sampling rates of up to 96 kHz while you're using the USB port.

Under OS X, the CoreAudio driver recognized the Mini-Me as soon as I connected it to my computer. Configuring it as the input and output source under system preferences was effortless, and within a minute of plugging it in for the first time, I was recording audio into the shareware audio editor, Amadeus II. The audio played back flawlessly over USB, allowing me to mix the computer's signal in my headphones with audio coming in through Mini-Me's line inputs using the direct/USB Mix knob.

I used Emagic Logic Platinum 6 on a G3 iBook to test the Mini-Me in a multitrack environment. I recorded a metronome pulse, then recorded clapping against the pulse. USB latency was in the range of 40 to 60 ms, which is about what I have noticed with other USB interfaces plugged into that computer.

FIG. 2: The Mini-DAC's front panel has only three controls, including a convenient input-select control.

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AIRWAVES

To record a series of WWII aircraft for a video game project, I toted the Mini-Me, a Crown SASS-P stereo mic, and a Tascam DA-P1 DAT recorder to the Planes of Fame Air Museum in Chino, California. I ran the mics directly into the Mini-Me, sending a 16-bit, 44.1 kHz signal from its S/PDIF output to the DA-P1.

The Mini-Me's mic preamps and A/D converter were a marked improvement over the DA-P1's internal electronics. The recordings of the planes' massive 60-year-old engines had a nice combination of throatiness and detail. Quiet Foley sounds, such as cockpit levers and switches, came out clean and noise-free.

In the studio, I recorded acoustic guitar, vocals, piano, and jingling keys (my favorite high-frequency test) at several sampling rates. The results were consistently musical, clean, and detailed. The Mini-Me's conversion handled the metallic high-frequency material without creating noticeable distortion or harshness.

Mini-DAC Specifications

Analog Outputs	(2) XLR; (1) ⅛" stereo; (1) ¼" stereo headphone
Digital Inputs	(1) AES/EBU D-sub; (1) coaxial S/PDIF; (1) optical S/PDIF/ADAT/S/MUX; (1) USB (optional)
Sampling Rates	44.1, 48, 88.2, 96, 176, 192 kHz (±10%)
Word Lengths	16- and 24-bit
Frequency Response	10 Hz–20 kHz (±0.2 dB at 44.1 kHz)
THD+N	–107 dB
Dynamic Range	119 dBA
Crosstalk	–125 dB
Power Source	6–16 VDC (adapter included)
Dimensions	5.4" (W) × 1.5" (H) × 9.8" (D)
Weight	2 lb.

FROM D TO A

The Mini-DAC is meant to be used at the end of the digital audio signal chain, typically feeding a pair of powered monitors. It handles a variety of inputs, including AES/EBU; coaxial S/PDIF; optical S/PDIF, ADAT, and S/MUX; and, with the optional input card, USB. All of the standard sampling rates from 44.1 to 192 kHz are supported.

The Mini-DAC has only three controls: a power switch, a level knob, and the Input Select knob (see **Fig. 2**). The 12-position input selector allows you the flexibility to monitor any of the four pairs of ADAT inputs (channels

PRODUCT SUMMARY

Apogee Electronics

Mini-DAC

DAC

$1,195 ($995 without USB)

FEATURES	4.0
EASE OF USE	3.5
AUDIO QUALITY	4.5
VALUE	2.5

RATING PRODUCTS FROM 1 TO 5

PROS: Great sound quality. Highly portable. Rugged.

CONS: Must disconnect and reconnect the USB cable when switching from 44.1 kHz to 48 kHz audio.

Manufacturer

Apogee Electronics Corp.
tel. (310) 915-1000
e-mail info@apogeedigital.com
Web www.apogeedigital.com

1-2, 3-4, 5-6, or 7-8) coming in over Lightpipe, as well as either of the two rear-panel AES/EBU inputs at sampling rates of up to 96 kHz. In addition, you can choose to monitor the AES/EBU inputs together in double mode for 176.4 and 192 kHz operation. The front panel also includes a ¼-inch headphone jack, a set of sampling-rate LED indicators, a pair of signal-lock LEDs, and a pair of signal-level LEDs. It would be nice if the Mini-DAC had four signal-level LEDs per channel the way the Mini-Me does.

The rear panel has a power input; a S/PDIF coaxial jack; an optical jack for S/PDIF, ADAT Lightpipe, and S/MUX signals; a spot for the optional USB input; and a DB9 AES/EBU I/O jack. A DB9 cable, with two AES/EBU XLR inputs at the other end, is included. I would prefer to have two AES/EBU XLR inputs on the rear panel, but there's not enough room. The analog outputs are a pair of +4 dBu XLR jacks and an unbalanced –10 dBV ⅛-inch stereo jack.

DETAILS, DETAILS

Whether it's monitored through headphones or through the XLR outputs, the Mini-DAC's sound is clean, detailed, and unhyped, with a smooth frequency representation across the spectrum. I listened to several projects of my own as well as recordings I know very well, such as the Beatles' *Sgt. Pepper's Lonely Hearts Club Band* and the recent Steely Dan releases, and the Mini-DAC revealed subtle nuances and details I had never heard before. I have no doubt that the Mini-DAC would be a very useful tool for presenting an objective sonic perspective during the mixing process.

A peculiar bug popped up when I used the Mini-DAC's USB connection. When powered up, the Mini-DAC slaved to USB at a sampling rate of 48 kHz, and all of the 48 kHz files sounded terrific when played back—tight, clean, punchy, and loaded with detail. However, the 44.1 kHz files sounded terrible—noisy, grainy, and flat—because the Mini-DAC did not automatically resync to 44.1 kHz, causing improper sampling-rate conversion. Unfortunately, there are no controls in the Mac's system preferences or on the Mini-DAC to set the USB's sampling rate. To resync USB to the proper sampling rate, I had to disconnect the USB cable from the Mini-DAC and reconnect it. When the Mini-DAC received signals from the AES/EBU ports, it resynced correctly to any changes in the sampling rate.

PREACHING CONVERTERS

Apogee's Mini-Me and Mini-DAC make a fine pair. Their sonic qualities are first-rate, and their flexibility in accepting and translating a wide variety of digital formats will make them welcome in many professional studios. More importantly, they can run on battery power, and their portability, solid construction, and USB capabilities make them ideal for field and concert recording applications.

If you work only in a stationary recording studio, there are other converter options available that offer more features for less money. But if the notion of a high-quality, lightweight, and mobile recording system is of interest to you, then these babies are definitely worth examining more closely. ❁

ROGER LINN DESIGN

ADRENALINN II

A unique effects processor for guitarists takes a step up.

By Geary Yelton

Although Roger Linn has had a measure of success as a guitarist and songwriter, he is best known for designing the first digital drum machine (the Linn LM-1) as well as Akai's popular MPC series of sequencer workstations. In recent years, he has turned his attention to creating new effects processors for guitarists, a path that led to developing the AdrenaLinn, winner of a 2003 **EM** Editors' Choice award (for more info, see the August 2002 and January 2003 issues, available online at www.emusician.com).

Like the first AdrenaLinn, the new model hosts dozens of effects in a stompbox the size of a paperback novel (see **Fig. 1**). In addition to offering basic drum-machine functionality, it is a sequencer-driven dynamic filter, distortion processor, amp modeler, delay line, noise gate, and more. You can load one effects Preset and one drum pattern at a time and switch them by turning a knob, pressing a footswitch, or sending a MIDI command.

In many ways, the AdrenaLinn II is identical to its predecessor, but it has numerous enhancements, including simplified effects programming, three times the maximum delay time, twice as many amp models, and revised drum sounds and patterns. Owners of the older model can order a kit ($99) to upgrade their AdrenaLinns to the newer version.

TAKE II (AND CALL ME)

The shell of the AdrenaLinn II is a black cast-aluminum box with a vinyl overlay that's available in yellow or blue. Front-panel controls consist of two rugged footswitches, four detented infinite-rotation knobs, four tiny buttons, and a trim pot for adjusting the input level. A red Clip LED and 16 green indicator LEDs supplement the main display, a large 3-character LED. The construction is reasonably robust, and the unit should easily withstand anything short of outright abuse.

FIG. 1: Roger Linn Design's AdrenaLinn II supplies so many effects that it takes the place of several stompboxes. It also works as a basic drum machine and offers filter effects other devices can't.

The back panel provides a ¼-inch mono input for your guitar, two ¼-inch outputs, MIDI In and Out ports, and a connector for the included 9 VDC wall wart (see **Fig. 2**). A ¼-inch stereo headphone jack is a welcome addition that was missing from the earliest models of the AdrenaLinn. All the connections are labeled on the top of the front panel, making them easy to identify from overhead. I am disappointed that the unit has no power switch. I also wish that the device could be battery-powered for increased portability.

One especially thoughtful touch is that the AdrenaLinn II has two output settings: one for a flat-response sound system and another for a guitar amp. The Amp setting compensates for the boost in upper midrange frequencies that a guitar amp typically produces; it affects the sound of only the amplifier models.

On the front panel is a printed matrix from which you can select rows of functions for each of the four knobs. By default, turning the knobs selects Presets, Drumbeats, Tempo, and Volume, and the alphanumeric LED displays the selected value. Holding the Main button for half a second selects four secondary functions, and tapping on the button returns to the main functions. For additional functions such as editing Presets, arrow keys step through eight rows of four parameters printed on the front panel: two rows for effects, two for amp models, one for delay, two for Drumbeats, and one for MIDI. By pressing the up and down buttons at the same time, you can also access hidden parameters such as alternate modulation sources and filter modes.

Because the main display shows only three characters, most words must be abbreviated. A legend printed on the front panel shows the 3-letter names for the 14 effects types and 6 amp models. Because more amp models are available, though, and because parameter names are also abbreviated, you might want to keep the user manual handy.

PAUSE FOR EFFECTS

The AdrenaLinn II provides a remarkable variety of effects that includes virtually everything except harmonization

STRIKE SONIC GOLD.

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GEARY YELTON

FIG. 2: The AdrenaLinn II's uncluttered back panel has a single analog input, two analog outputs, a stereo headphone jack, two MIDI ports, and a wall-wart connection.

and reverb (which are probably better left to more expensive or dedicated devices anyway). Although its specialty is filter effects, the AdrenaLinn II is also capable of rotary-speaker simulation, vibrato, tremolo, autopan, flanging, chorus, and even arpeggiation. Each effect offers a handful of variations; for example, tremolo can be a slow stereo pan, a hard switch between left and right, a slow pan to the left followed by a hard pan to the right (called sawtooth tremolo), or a randomly selected pan position for each tremolo pulse.

A Preset combines one effect, an amp model, and a delay. You can set up the Bypass switch to determine the combination that goes into effect when you press it: effect only, effect with delay, and so on. Time-based effects can always be synced to tempo, which is determined by the current Drumbeat or by an external MIDI Clock. A single cycle of a flanger, for example, can be a quarter note, one bar, or four bars long. Random Filter is a Preset that selects a new cutoff frequency every eighth or 16th note.

Some effects are variations on their classic counterparts. Most filter effects use an envelope generator triggered by an audio signal to control the filter's cutoff frequency, which produces an auto-wah type of effect first popularized in the late '70s by a stompbox called the Mu-Tron (see **Web Clip 1**). Talk Box is a simulation that doesn't require you to put a plastic tube in your mouth. It's really just another autofilter effect, but it sounds as though it could occasionally substitute for the real thing (see **Web Clip 2**). Rotary Speaker has two speeds, but the shift between them is abrupt, unlike the shift on a real Leslie.

The AdrenaLinn II offers several Presets that use Filter, Tremolo, or Arpeggio Sequences. The factory settings provide 20 sequences of each type, and if you have the patience, you can program a different sequence for each of the 100 User Presets. Each sequence is 32 steps long. For each step, you can determine the Level and whether a two-stage envelope generator is on or off. The Level parameter controls cutoff for Filter Sequences, volume for Tremolo Sequences, or pitch for Arpeggiator Sequences. Because Arpeggiator Sequences always play the same note pattern no matter what you play (see **Web Clip 3**), I found the Filter Sequences to be much more versatile (see **Web Clip 4**). You can, however, modulate Arpeggiator Sequences with a MIDI keyboard or other controller.

Several of the effects offer MIDI-controllable parameters. You can control envelope rates with Velocity, for example, or control filter frequency with Aftertouch or one of several other MIDI CCs. You can also assign MIDI CCs to modulate flanger frequency.

The delay section provides a single-tap delay, which doesn't sound especially exciting at first glance. However, you can specify delay times that relate to tempo in a variety of lengths: one every two measures, three every four measures, one every measure and a half, and other variations. The maximum delay time is an impressive 2.8 seconds, long enough to record and play loops in real time (see **Web Clip 5**). Even at the highest settings, though, it can't produce an infinite repeat.

MODEL BEHAVIOR

Even if you use such effects rarely, you'll probably make frequent use of the AdrenaLinn II's guitar-amplifier modeling. You can select from 24 amp models ranging from simulations of genuine classics to simple fuzz and clean console EQ, and they all sound quite good. Real amp simulations include four Fenders, three Marshalls, a Hiwatt DR103, a Budda Twinmaster, a Roland JC-120, a Soldano SLO-100, two Mesa/Boogies, and two Voxes. Five additional settings are Roger Linn Design Thin, Blues, Deep, Bright, and Rectified.

AdrenaLinn II Specifications

Analog Input	(1) unbalanced ¼" TS
Analog Outputs	(2) unbalanced ¼" TS; (1) ¼" stereo headphone
MIDI Ports	In, Out
Effects Presets	(100) factory, (100) user
Drum-Pattern Presets	(100) factory, (100) user
Effect Types	14
Amp Models	24
A/D/A Conversion	24-bit, 40 kHz
Internal Processing	32-bit
Display	3-character LED
Power	9 VDC adapter
Dimensions	7.5" (W) × 1.4" (H) × 4.7" (D)
Weight	2 lb.

Control Yourself

Your music hardware and software gives you all the instruments and recording capabilities you could dream of. But operating these fantastic tools with a mouse is like juggling with one hand tied behind your back. Whether you're a control freak or just tired of mousing around, the Evolution UC-33e lets you unleash the full potential of your music.

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24 assignable rotary knobs

Drawbar mode for faders

Recall and store up to 33 presets

9 assignable faders

10 assignable MIDI buttons

Snapshot sends current controller positions

4 additional programmable MIDI buttons

Easy USB connection, PC and Mac compatible

Can also be used as USB MIDI interface

The UC-33e is the affordable, easy-to-use remote USB controller for any computer music system or MIDI setup. It allows instant and simultaneous control of up to 47 different parameters—and with 33 memory locations, changing setups is fast and easy. Controller Mute lets you preadjust controller positions without sending data, then send all current settings at once using Snapshot. Use the UC-33e with a conventional MIDI setup, for live performances, or as a remote control for your existing studio rack. 30 presets allow immediate control of the most popular virtual instruments and host applications like Reason, Live, Logic, Cubase SX, Native Instrument's B4 and more.

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EVOLUTION

PRODUCT SUMMARY

Roger Linn Design
AdrenaLinn II
effects processor
$499

FEATURES	4.5
EASE OF USE	3.5
SOUND QUALITY	4.5
VALUE	4.5

RATING PRODUCTS FROM 1 TO 5

PROS: Lots of unique effects. Real-time MIDI control. Outstanding value.

CONS: No power switch. No battery power. Can't display Preset or parameter names.

Manufacturer
Roger Linn Design
tel. (510) 898-4878
e-mail sales@rogerlinndesign.com
Web www.rogerlinndesign.com

For each amp, you can set the overdrive gain, 3-band EQ, and output volume. You can also specify whether the effects occur before or after the amp in the signal chain. All of the overdrives sound great, though not necessarily as warm as the distortion a real tube amp produces.

DRUM AND DRUMMER

The Drumbeat section plays four-voice patterns using individually sampled hits, but it lacks some of the capabilities of a full-function drum machine. Pattern length is always two measures, and you can't chain patterns together to sequence a song. Nor can you send MIDI Program Change messages to change Drumbeats without also changing the effects Preset. You can change them with MIDI Song Select messages, however, so you can chain patterns with the aid of an external sequencer.

Even without onboard pattern sequencing, the Drumbeat section makes the most of its resources. Like the original AdrenaLinn, the AdrenaLinn II has memory enough for 100 factory beats (F00 through F99) and 100 user-programmable beats (U00 through U99). Straight from the factory, the User Bank duplicates the Factory Bank to supply beats you can edit. Location 99 contains a metronome click, and 100 is silent. Styles range from straight rock to syncopated funk to swinging hip-hop to some pretty happening techno patterns (see **Web Clip 6**). A few make good use of the onboard processing, but most are dry. Each beat has its own tempo, but you can easily override that default. The collection offers enough variety that you should have no problem finding a groove that inspires you.

As you might expect from the man who invented the LinnDrum, the quality and selection of the drum samples is consistently good. Most are recordings of real drums, but synthesized sources such as the Roland TR-808 are available, too. Each Drumbeat contains 16th notes, eighth notes, or eighth-note triplets; the two measures are identical in most factory beats. Each step contains a bass drum, snare, hi-hat, and percussion voice. Each of the first three voices offers a choice of 9 samples, and the last gives a choice of 15 sounds. Whereas the bass drum is always a kick and the snare drum is always a snare, the hi-hat voice could be a different cymbal or a tambourine. A percussion sound might be a tom, a triangle, an electronic zap, or any sound that doesn't fit the other categories.

THE SOUND AND THE FURY

Guitarists have been reaching for new sounds since electric guitars first appeared, and effects processors have steadily evolved to meet their needs. The AdrenaLinn II is a significant step in the evolution of stompboxes. Considering the number of effects that are packed into this single stompbox, the AdrenaLinn II is a bargain in anyone's book. It does so many sounds that it might take you awhile to explore everything. Luckily, the user manual is quite good. Best of all, the AdrenaLinn II offers effects that you won't find anywhere else.

EM WEB CLIPS

Associate editor **Geary Yelton** *used to play bass professionally, but now he's a veteran knob-twister. Still, his favorite instrument is a Gibson guitar that's even older than he is.*

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FOCUSRITE

ISA 428 PRE PACK

Carrying the legacy of the ISA 110 into the digital age.

By Eli Crews

From the lineage of the Focusrite ISA 110 comes the ISA 428 Pre Pack, which combines four high-quality transformer-based preamps in one device. When fitted with the optional MH-442 A/D card, the ISA 428 can be used as an 8-channel front end for your DAW or digital recorder.

ON THE SURFACE

The front panel of the ISA 428 stays true to the ISA heritage; the blue and gray panel is adorned with yellow knobs and red buttons and LEDs (see **Fig. 1**). What look like four oversize VU meters are actually peak meters in disguise, complete with LEDs that indicate channel overload. Beside each meter are buttons for +48V phantom power, phase reverse, and activating the send and return for each of the four preamp channels.

Below each meter are three knobs. The first knob sets the base gain amount in four 10 dB steps. The second knob is a continuously variable control that offers 20 dB of gain to the mic and line inputs and 30 dB of gain to the channel's instrument input. All four instrument inputs are located at the far left of the front panel, which keeps the cables conveniently out of the way when you're reaching for the controls.

One of my favorite features on any preamp is a variable low-cut filter, and the ISA 428 has one. The highpass filter ranges from 16 and 420 Hz, allowing you to block only subsonic frequencies or to drastically alter a source's sound. Given the 18 dB-per-octave rolloff of this filter, setting it anywhere above its middle range will thin the sound out considerably. My only gripe about the filter is a small one: only two landmarks are given between the extremes, and somewhat odd ones at that—34 and 210 Hz. It would be nice to have a few more frequencies indicated.

Below the knobs are buttons for selecting the input source—Mic, Line, or Inst—and input impedance—Low, ISA 110, Med, or High (more on this in a moment). The rest of the front panel is dedicated to controls and meters for the A/D card: there's a Soft Limiter button; a Clock Select button, which determines a sampling frequency of 44.1 through 192 kHz; a Bit Depth Select button for choosing a 16-, 20-, or 24-bit word length (with dither added to the 16- and 20-bit output); and an Ext sync button for clocking the device from word clock or Superclock, using the rear-panel BNC jack. The Lock LED illuminates when the ISA 428 detects the clock input.

On the far right of the faceplate are eight 6-segment LED meters that indicate output levels—either analog only or analog and digital if the A/D card is installed. In digital mode, the meters are postlimiter and reflect the reduced level if the limiter is engaged and active. There are eight output meters instead of four because the MH-442 is an 8-channel converter. To take full advantage of this, the ISA 428 comes with four additional rear-panel line inputs that provide access to the extra channels, greatly increasing this product's value.

The rear panel features four mic inputs on XLR jacks, four balanced line inputs on TRS jacks, four analog outputs on XLR jacks, and individual balanced ¼-inch send and receive jacks for each channel (see **Fig. 2**). The A/D card has two BNC connectors for sync, two ADAT Lightpipe ports, and two 9-pin D-type connectors. Two ADAT ports are needed because at 88.2 and 96 kHz each

PRODUCT SUMMARY

Focusrite
ISA 428 Pre Pack
mic preamp
$1,995
MH-442 A/D converter $695

FEATURES	4.5
EASE OF USE	4.5
QUALITY OF SOUNDS	4.5
VALUE	3.5

RATING PRODUCTS FROM 1 TO 5

PROS: Clean sound. Sounds great on a variety of sound sources. Variable high-pass filters. Switchable impedance on mic inputs. Inserts are bypassable using front-panel switches. Dithers to 20 and 16 bits. A/D card converts eight channels of audio. A/D card has a variety of digital output options.

CONS: A/D card gets very hot.

Manufacturer
Focusrite USA Inc./Digidesign (distributor)
tel. (866) FOCUSRITE
e-mail sales@focusrite.com
Web www.focusrite.com

FIG. 1: Focusrite's ISA 428 Pre Pack is a 4-channel mic preamp descended from the transformer-based ISA 110 of the mid-1980s.

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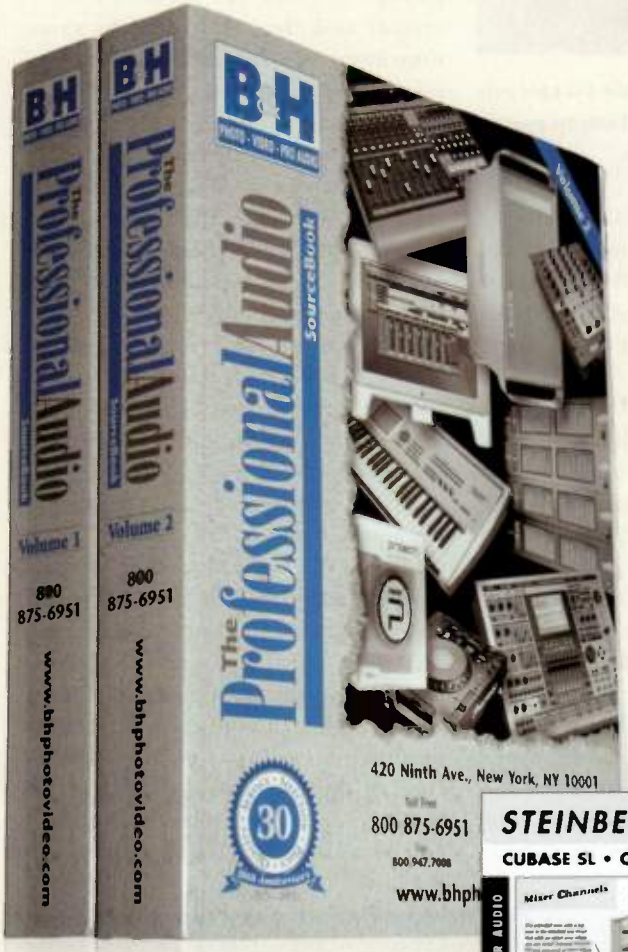

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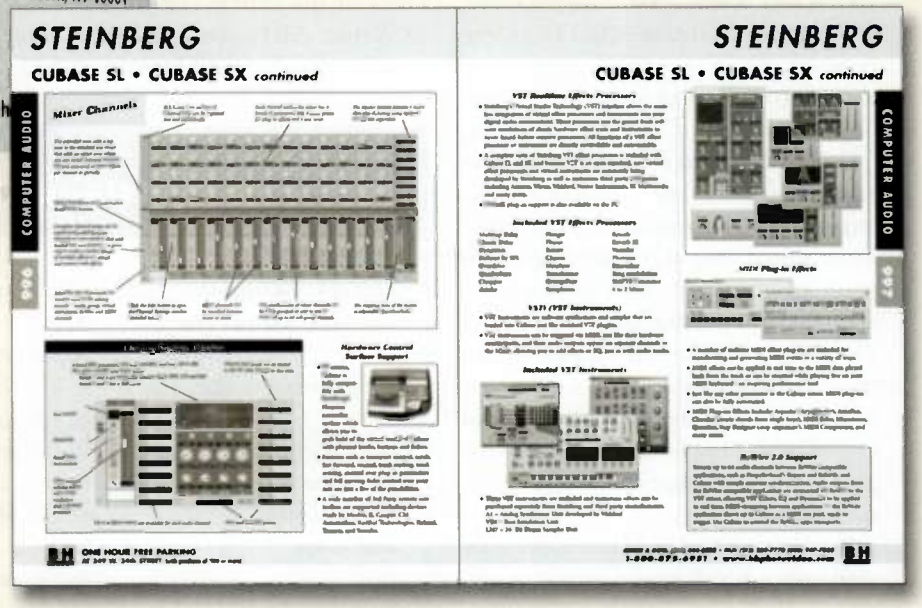

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FIG. 2: The ISA 428 provides balanced inserts on all four channels. The optional MH-442 card puts eight A/D converters at your fingertips, and the ADAT, AES/EBU, and S/PDIF digital outputs operate simultaneously.

port carries four of the available channels. At lower sampling frequencies, all eight channels appear at each port.

DRUMS AND WIRES

I first used the ISA 428 for a concert recording of a jazz trio. I used a pair of Neumann KM 184s for the drum overheads, and I sent the signal from the AES/EBU output to my Metric Halo Mobile I/O 2882 and into MOTU Digital Performer 4. The sound was clean, clear, and very detailed—as good as any drum sound I've recorded in that acoustically challenged room.

Back in the studio, I tried the ISA 428 on more drums, recording direct to 2-inch analog tape using an EV RE20 on the kick, a Shure SM57 on the snare, and a pair of Oktava MK012s on the cymbals. The results were excellent, with a sound that was thoroughly punchy and crisp.

One of the most impressive features of the ISA 428 is the switchable input impedance. The amount of impedance a mic signal sees at the preamp input plays an enormous role in the tonal characteristics of the amplified signal. The ISA 428's manual states that the "best" input impedance for a given mic is around ten times the mic's output impedance. This would lead you to expect a different impedance setting for each type of mic you use.

However, when I used three of my favorite mics, which have a variety of output impedance ratings—the Blue Dragonfly Deluxe is rated at 50Ω, the RCA 77DX ribbon mic at 300Ω, and the Schoeps 221B tube mic at 200Ω—I ended up choosing the Low (600Ω) impedance setting on the preamp. The same goes for the drum tracks noted earlier. To my ears, the Low setting sounded the smoothest in all but one case: when using the SM57 (rated at 150Ω) on the snare drum, I used the High setting because it offered just the right amount of crispness to really make the drum stand out with a snap.

BLUE AND GRAY WITH GREEN

Next up was a session with woodwind specialist Aaron Bennett playing piccolo. Typically, I would use the Focusrite Green 1 Dual Mic-Pre because it ably captures the transparency and breathiness of wind instruments. Using the Blue, RCA, and Schoeps mics, I compared the ISA 428 with my Green 1. The difference between the two preamps was most noticeable with the ribbon mic: the ISA 428 brought out much more of the clarity and detail of the RCA than the Green 1, which by comparison sounded muddy and somewhat flat and two-dimensional. The differences between preamps were less obvious with the Blue and Schoeps mics, though the ISA 428 seemed a little crisper and cleaner, but at the same time more mellow with the shrill highest notes of the piccolo.

I also recorded Bennett playing baritone saxophone through the same setup. Whereas the Blue and Schoeps mics gave me a slightly richer sound through the ISA 428 than through the Green 1, the RCA really let me hear the differences between the two preamps. The ISA 428 offered a greater clarity in tone, especially on the rich harmonics produced by the low reed instrument. Multiphonics are one of Bennett's specialties, and the ISA 428 captured every nuance and complexity present in his tone.

I tested the ISA 428 further by comparing it with the Universal Audio 6176 and a Focusrite Platinum VoiceMaster. This time, the test subject was electric bass, both miked through an amp using an AKG 414 EB and running direct into the instrument input on the front of each of the preamps. I expected the preamps' tones to be drastically different, because the 6176's preamp is tube

ISA 428 Pre Pack Specifications

Analog Audio Inputs	(4) XLR; (4) ¼" TRS; (4) ¼" TS
Analog Audio Outputs	(4) XLR
Additional Analog Ports	(4) ¼" TRS insert sends; (4) ¼" TRS insert returns; (4) XLR A/D inputs (functional only with MH-442 card)
Digital Ports (MH-442)	(2) BNC word-clock I/O; (2) ADAT Lightpipe outputs; (1) AES and S/PDIF 9-pin D-Sub output; (1) AES-only 9-pin D-Sub output
Bit Depths	16, 20, or 24
Sampling Rates	44.1, 48, 88.2, 96, 176.4, or 192 kHz
Gain Range	line input ±18 dB; mic input 60 dB; instrument input 30 dB
Input Impedance	line input 10 kΩ; mic input 600Ω, 2.4 kΩ, 6.8 kΩ, and ISA 110 setting (variable); instrument input 1 MΩ
Signal-to-Noise Ratio	120 dBA
Total Harmonic Distortion	0.003% with 0 dBu, 1 kHz input
Noise Performance	line input –96 dB; mic input –128 dB (EIN with 150Ω input resistance at 60 dB of gain)
Dimensions	2U × 10" (D)
Weight	15.76 lb.

based and the VoiceMaster is part of Focusrite's lower tier of products. I was right—the ISA 428 had a tighter, less floppy low end than the 6176, and the high-mids were much smoother and almost sounded compressed. In addition, I was able to drive the ISA 428's gain much higher without the sound breaking up. The VoiceMaster held its own remarkably well, but the ISA 428 sounded cleaner in the low end and had a bit more shimmer. The ISA 428 gave me one of the best direct-bass sounds I have ever recorded, and I am usually not a huge fan of recording the bass direct.

OVER AND UNDER

I also compared the ISA 428's converters by sending its digital and analog outputs simultaneously into my Digidesign Digi 001's inputs and my Metric Halo Mobile I/O's AES/EBU input. The differences were subtle, because all three of these devices have above-average A/D converters. I was particularly impressed with how good the ISA 428 sounded next to the Mobile I/O, because Metric Halo's converters are widely heralded as some of the best available.

However, I was slightly less impressed with the ISA 428's Soft Limiter. Focusrite states that the Soft Limiter's opto circuit is a new design that should prevent digital overs entirely and minimize the "undesirable" distortion that standard limiter circuits generate. During my tests, digital overs did get through when I slammed the converters with a hot signal, and there was certainly undesirable distortion, though admittedly less than would have occurred without the limiter engaged. Although the limiting was hardly noticeable at normal-to-high operating levels, I did hear it pumping and popping when pushed to extremes. When engaged, it also seemed to drop the digital output level and negatively affect the tone somewhat. As a result, I would engage the Soft Limiter only when dealing with unpredictable sources that were prone to extreme transient peaks.

110 PERCENT

The ISA 428 Pre Pack is a versatile multipurpose preamp that sounds excellent and offers plenty of control of its very flexible tonal characteristics. Coupled with the optional MH-442 converter card, the ISA 428 serves as an easy, no-compromise way of adding eight inputs to a digital-recording rig.

At $500 a channel, the ISA 428 isn't the least expensive mic preamp you'll find. Nonetheless, Focusrite has lived up to its reputation for making quality gear, and it has done so at a competitive price point.

Eli Crews *co-owns and operates New, Improved Recording in Oakland, California. Read all about him or drop him a line at www.newimprovedrecording.com.*

STEINBERG

XPHRAZE 1.1 (MAC/WIN)

Rephraze your sound palette with this unique plug-in synth.

By Len Sasso

At first glance, Xphraze looks like a standard subtractive synth with a step sequencer placed prominently in the middle of its interface. Looks can be deceiving: Xphraze is anything but standard. The included multipart patches (called Combis) will keep you busy, but don't overlook the programming power under the hood. Once you get your hands on this synth, you won't want to let go.

Xphraze is a VST 2.0–format soft-synth plug-in for Mac and PC, and as such, requires a VST 2.0–compatible host. For this review I used Steinberg's Cubase SX 2.0 and Emagic's Logic 6.3.1 (after converting Xphraze to Audio Unit format using FXpansion's VST to AudioUnit Adapter). Xphraze needs a lot of horsepower, so you'll want a hefty computer to make the most of it. My G4/800 MHz PowerBook could typically manage four voices with complex Combis and several more with simpler ones, even after making use of the handy tips in the manual for conserving CPU power.

Xphraze combines two classic synthesis techniques—wave sequencing and vector synthesis. Wave sequencing is accomplished using the Phraze Generator, the front-panel section that resembles a step sequencer (see **Fig. 1**). It acts as the oscillator for the subtractive-synth-style modules surrounding it. Vectoring is controlled on the Advanced control panel (see **Fig. 2**). Vectoring determines the relative level of four independent wave-sequenced patches, which, in essence, amounts to vectored wave sequencing.

The subtractive-synth modules associated with the Phraze Generator include a multimode filter and amplifier (each with a dedicated, loopable envelope generator), two additional auxiliary envelope generators, four LFOs, and a multi-effects processor with 24 effects covering the usual bases. The interesting twist here is that the Phraze Generator's output is treated as a single sound source—for example, the entire Phraze is affected by a single envelope sweep.

FIG. 1: Xphraze's main control panel is where you set up patches for its four independent Phraze synthesizers. The step-sequencer-style panel in the middle (called the Phraze Generator) is the sound source that is processed by the surrounding subtractive-synth modules.

Minimum System Requirements

Xphraze 1.1

MAC: G4/400 MHz; 70 MB RAM; Mac OS 9 or OS X 10.2; VST 2.0–compatible host

PC: Pentium III/600 MHz; 70 MB RAM; Windows 2000 or XP; VST 2.0–compatible host

There's a very flexible modulation matrix (see **Fig. 3**) for routing the envelopes, LFOs, and a variety of MIDI controllers to virtually any synthesis parameter. The same modulation source can be routed to multiple destinations, each with its own polarity and amount. That allows you to do things like pitch-bend patches in different directions, pan outputs apart and together with the same MIDI controller, and modulate envelope attack and decay times with Velocity.

The envelopes and LFOs are especially noteworthy for their flexibility. The envelopes can have as many as 128 breakpoints, and each segment has an adjustable curve. Breakpoint times can be set in note increments or time values with the smallest step being a 128th note or 7.8 ms. Any range of breakpoints can be designated as a loop, and looping can be forward or alternating. As you can imagine, envelopes can get pretty complex, and fortunately, they can be copied and pasted. Other useful envelope features include normalizing and instant halving and doubling of tempo. The LFOs are from Steinberg's Plex plug-in. Each LFO is actually a 16-step sequencer with adjustable smoothing between steps. Shape buttons let you select typical LFO waveforms with full smoothing, and a random button produces a random sequence, which, when smoothing is turned down, acts like a sample and hold.

Xphraze contains four identical synthesizers, each of which can be assigned to any of the first four MIDI channels. Combis usually layer the synths on the same channel, but you can also use Xphraze as a multichannel synth. The File Browser at the left of the control panel allows you to manage presets at all levels, from full Combis to individual

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synthesizer patches to Xphraze patterns. It is also used for loading your own multisamples into the Phraze Generator.

WHAT'S IN A PHRAZE

The name Xphraze leaves little doubt that the Phraze Generator is at the heart of this synth. It is a 32-step sequencer in which each step plays an oscillator waveform or an audio sample. The key here is that you can change the sound source at each step. For example, you could use different samples (even from entirely different multisampled instruments) at each step to produce a drum, or bass, or lead sequence. You could also use a different waveform at each step to evolve a complex wave sequence or crossfade between extended audio clips at each step to produce long evolving pads and ambiences. Because the step size can range from a 64th-note triplet to 8 bars, you can produce a wide range of effects using almost any kind of audio material.

In addition to having its own multisample or oscillator waveform, each Phraze Generator step has six modulation parameters that can be routed to modulate virtually any parameter of the surrounding synth modules. The sixth parameter slot has an overall-amount control to scale the individual step settings—a nice touch. This control can be modulated by MIDI controllers or by Xphraze's built-in LFOs and envelopes. In short, a Phraze is a whole lot more than a sequence of notes.

Each Phraze Generator can hold four Phrazes, though only one can play at a time. You select Phrazes with MIDI Program Change messages or with MIDI Note messages using Xmode. (More on the live-performance modes, Xmode and Xmix, in a moment.) Phrazes and loops need not be 32 steps—you can adjust the beginning and end as well as the loop boundaries. Because you can easily copy and paste between Phraze buffers, changing end and loop points is a great tool for quickly getting variations of your favorite Phrazes. Phrazes can be looped forward, backward, or alternating with or without repeated endpoints. Looping can be turned off so that each note triggers one pass of the Phraze. Steps can be crossfaded by varying degrees, facilitating smooth transitions for pads and ambiences as well as complete separation for percussive sounds. Finally, the two triggering modes allow Phrazes to be restarted with each incoming note or to pick up at the current Phraze position.

BEYOND THE PHRAZE

Xphraze's Advanced control panel (which you access by clicking the Advanced button to the left of the onscreen keyboard) is used to set up vector synthesis (automated mixing of the four patch slots), create your own multisamples, manage global effects routing and mixing, and activate the performance modes, Xmode and Xmix. Xmode lets you use MIDI notes G, A, B, and C in your choice of octave to switch Phraze buffers for any or all of the patch slots. Xmix (short for *Xtended Remix*) lets you use the notes between C1 and B3 on a selected MIDI channel to mute and unmute patch slots and to select Phraze buffers. Notes C4 and above on the Xmix channel play normally.

The Master FX section features an independent multi-effects processor for each of Xphraze's four outputs. Those effects offer more controls than the patch effects, but they can't be modulated. Any patch can be routed to any of the four outputs, which means that you could assign each patch its own Master effect. More typically, all the patches would be routed to the same output, and one Master effect would be used to process all four patches.

The multisample editor lets you create your own multisamples as sources for the Phraze Generator. You can have up to 16 user multisamples for each patch

PRODUCT SUMMARY

Steinberg
Xphraze 1.1 (Mac/Win)
software synthesizer
$249.99

FEATURES	4.5
EASE OF USE	4.0
QUALITY OF SOUNDS	4.5
VALUE	4.5

RATING PRODUCTS FROM 1 TO 5

PROS: Unique and flexible synthesis architecture. Broad spectrum of factory sounds. Excellent documentation.

CONS: Requires powerful CPU. Limited multisample mapping.

Manufacturer
Steinberg
tel. (818) 973-2788
e-mail info@steinbergusa.net
Web www.steinbergusa.net

FIG. 2: The Advanced page is used for Vector Synthesis (left), master-effects management (center), and setting up user multisamples (right).

FIG. 3: Xphraze's modulation matrix lets you route a variety of sources, including built-in envelopes, LFOs, and MIDI controllers, to modulate virtually any synthesis parameter. You can route a single source to multiple destinations with different polarities and amounts.

slot, and they function exactly the way the factory multisamples and oscillator waveforms do. To create a multisample, you load audio files either singly or in batches into Xphraze's Multisample Mapper. If you load a batch of samples, Xphraze will automatically map them to the correct keys if root-key and key-range information is included in the sample file; otherwise they will be mapped to consecutive keys starting with C3. The automatic mapping worked for all of the samples I tried, but the mapping to consecutive keys when no information was present didn't work in every case. Once samples are loaded into the Multisample Mapper, you can adjust their positions and borders. A more fully featured mapping scheme (with Velocity zones, for example) would be nice, but the ability to create multisamples is in itself a terrific feature.

Xphraze takes an automated approach to vector synthesis. Each corner of the vector-synthesis rhombus (visible on the left side of Fig. 2) corresponds to one of the four patch slots, and the position of the vector inside the rhombus controls the mix of the four patches. You can automate the vector's position using a dedicated envelope generator as well as a mix of the four LFOs from the first patch slot. That allows you to create a huge variety of automated mix patterns. The MP3 example Vectorama (see **Web Clip 1**) uses envelope and LFO vectoring to control the mix of four ambient patches. The vector-synthesis section can be turned off, leaving individual patch volumes to control the mix. Although you can't control the vector position manually or with MIDI, you can accomplish much the same thing by using the Modulation Matrix to assign a MIDI controller with differing amounts and polarities to the individual patch volumes.

Xphraze is a most unusual synth plug-in. The manual is excellent; the factory Combis include a wide range of building blocks for sound creation. A synth that does this much takes time to learn, but if you have a fast enough computer, Xphraze is well worth the effort and is a great value. EM WEB CLIPS

Len Sasso *can be contacted through his Web site at www.swiftkick.com.*

Quick Picks

BRIAN MOORE GUITARS

iGuitar 8.13

By Mike Levine

If you're looking for a guitar that offers quality tone and an entrée into the world of guitar synthesis and sequencing, the Brian Moore iGuitar 8.13 ($1,595) may be just the ticket. Featuring a pair of Seymour Duncan humbucking pickups that can be changed over to single-coil with a coil-tap switch, an RMC piezo bridge pickup for acoustic-guitar-like tone, and a 13-pin output for driving Roland and compatible guitar synths, the iGuitar 8.13 offers a wide range of sonic possibilities. It looks great, to boot.

For its guitar features alone, this Korean-made instrument—which is part of Brian Moore's i2000 series—has a lot to offer. It sports a gorgeous figured-maple top; a mahogany body; a 22-fret, bolt-on maple neck; and Sperzel locking tuners. It also has a cleverly placed output jack that's tucked into the upper back of the guitar's body, keeping the cable hidden and secure.

The Brian Moore iGuitar 8.13 offers not only a pair of humbucking pickups and a piezo bridge pickup, but also a 13-pin output for driving Roland or compatible guitar synths.

Clearly, though, it's the 13-pin output that sets this instrument apart from most guitars (Godin and Fender also make guitars with 13-pin outputs), allowing the iGuitar to connect to devices such as Roland's GR-33 guitar synth and GI-20 GK-MIDI interface. Those devices have MIDI outputs, facilitating the connection of the iGuitar to MIDI sound modules and MIDI sequencers and notation programs.

Getting Connected

There are a number of ways you can hook up the iGuitar 8.13. If you use the 13-pin connector only—hooked into a compatible device—you'll get synth and magnetic-pickup output. (Units with 13-pin outputs, such as the GR-33 and GI-20, have ¼-inch guitar out jacks for connecting to amps and processors.) If you want piezo output as well, you'll also need to use the included stereo Y-cable. It plugs into the guitar's ¼-inch output and terminates in two separate ¼-inch jacks, one for the magnetic output and one for the piezo output. If all you want is the output of the magnetic pickups, simply plug a ¼-inch TS guitar cable into the iGuitar.

The iGuitar 8.13 offers plenty of control for its various sound options. You get separate knobs for Magnetic Volume, Piezo Volume, and Tone (which pulls out to activate the coil tap). In addition to the aforementioned pickup selector, there are two toggle switches. The first is a three-position switch that lets you choose the magnetic sound, the synth output of a connected 13-pin device, or both together. (That works only when the guitar's signal is coming through the 13-pin cable.) The other, called Step Up, Step Down, sends out a Program Change to any attached 13-pin device.

Moore Sounds

When testing out the iGuitar, I first tried the straight magnetic-pickup guitar sound. The Seymour Duncan humbuckers were appropriately fat sounding, and with the guitar's three-position pickup selector and the coil-tap switch, there were plenty of tonal possibilities (see **Web Clip 1**). The RMC piezo pickup offered a solid, acoustic-like tone (see **Web Clip 2**), which would be great for live use but isn't going to replace an acoustic guitar for recording.

I also tested the guitar's 13-pin output through a Roland GI-20, plugged into my computer's USB. I controlled synth sounds through MIDI, and it tracked very nicely (see **Web Clip 3**). Tracking was even better when going directly into a Roland GR-33, which doesn't require MIDI to trigger its sounds.

By plugging the iGuitar in through the GI-20 and using the GI-20's USB output, I was able to easily record MIDI data directly into music software, including Sibelius's G7 guitar-notation program and MOTU's Digital Performer.

Aye, Guitar

The iGuitar 8.13 is a useful tool for the recording guitarist because it's a quality instrument and it gives you the ability to drive guitar synths as well. The piezo pickups are a nice addition but will probably be of more use in live-performance situations than in the studio.

If you don't already have a 13-pin sound source such as the Roland GR-33 or a GK-MIDI interface such as the Roland GI-20 (for connecting to MIDI synths and music software), you'll need to buy one if you want to get the most from this guitar. So factor that into your budget when considering a purchase. If it all adds up to more than you wanted to spend, you could also look into Brian Moore's i1000 line of guitars, which offers instruments with similar capabilities at lower prices. EM WEB CLIPS

Overall EM Rating (1 through 5): 3.5

Brian Moore Guitars; tel. (800) 795-PLAY or (845) 486-0744; e-mail info@brianmooreguitars.com; Web www.brianmooreguitars.com

TOADWORKS

Death Rattle

By Eli Crews

One way to get a unique guitar sound is to buy a complicated and pricey device that models every possible combination

Powerful and versatile, the ToadWorks Death Rattle stomp box delivers boutique quality at a mass-produced price.

of stompbox, amplifier, and cabinet, and spend your hours dialing in a few pleasing sounds. Another approach is to buy a few boutique pedals and use them to create a highly personalized palette of interesting and usable timbres. The ToadWorks Death Rattle ($144.95) works for the latter approach and offers plenty of flexibility from only a few controls.

Simplicity Is the Key

Roughly twice the size of the average stompbox, the Death Rattle offers three effects: Tweed (based on the Fender amp), Plexi (based on the Marshall amp), and Boost. Using the center footswitch, you can choose either the Tweed or the Plexi distortion circuit. The footswitch on the right activates the Boost circuit, which functions as a nonlinear level boost (with a bump in the low-mids) when activated on its own. When combined with one of the amp effects, it acts as a simple gain booster. The footswitch on the left is a true bypass. That gives you six sounds: unprocessed, Tweed, Plexi, Boost, Tweed/Boost, and Plexi/Boost.

Six large knobs give you control over the Tone and Level of each distortion circuit, the Boost level, and the overall output gain. The footswitches are smooth and sturdy, and the unit can be powered from a single 9V battery or a 9V (tip-negative) wall-wart adapter.

Sound Judgment

For this review, I ran a Fender Mustang and Gibson Les Paul through the Death Rattle before going to a Bedrock tube head with a 4×12 cabinet. With both guitars and in all pickup positions, I was able to dial up a number of tasty sounds.

As advertised, the Tweed circuit has the warm, crispy overdrive associated with Fender amps, and the Plexi has the beefy, gritty crunch that's associated with Marshalls. I'm more of a Fender-amp fan, but the Plexi circuit gave me great sustain and plenty of the metallic chunk you just can't get from a Fender amp alone. I was also impressed that the signal didn't sound processed, as it does through many amp emulators and pedals. The Plexi channel in particular sounded much more natural than I expected.

A couple of the combinations weren't so hot. For example, running the Mustang through the Plexi circuit sounded too washy for my taste, although turning up the Plexi's Tone control helped somewhat. In fact, the Death Rattle's tone controls are more usable and musical than the tone controls on similar pedals: they seem to do more than merely cut off the high end.

Booster Seat

The boost circuit is the only part of this pedal I take issue with. When an input signal is present and you engage the Boost switch after adjusting the Boost knob, you get a loud pop. If you leave the Boost knob alone, you won't get the pop when you hit the Boost switch. ToadWorks knows about the problem, noting on its Web site that it's an artifact of the circuit design. Nonetheless, it makes on-the-fly adjustment of the control somewhat dicey.

A new version of the Death Rattle, which became available in January (after this review was completed), corrects the popping problem in the boost switch. It also adds individual Gain controls for each of the distortion circuits and increases the amount of volume boost in the distortion circuits. A wish-list item for me would be a button that engages both distortion circuits in tandem. But I don't want to seem too greedy.

Mine

The Death Rattle's sound, build quality, versatility, and price make this boutique pedal an incredible value. Needless to say, this stompbox is not leaving my studio.

Overall EM Rating (1 through 5): 4.5
ToadWorks; tel. (619) 895-8111; e-mail info@virtualtoad.com; Web www.virtualtoad.com

ELAB

Smokers Delight (Audio/WAV/REX)

By Marty Cutler

There was a time in the 1970s when jazz, funk, and R&B became reacquainted, resulting in a rhythmic and earthy style of music that was bolstered by some of the harmonic sophistication and ambiguities of jazz. Undoubtedly, the instrumentation and analog recordings helped shape a unique and atmospheric sound that relied on real Wurlitzer electric pianos, rounder and fatter-sounding kicks and snares, an unabashed use of analog signal processors, a healthy appetite for dissonance, and the use of supple grooves. ELab's *Smokers Delight* ($99.95) hearkens back to that time before digital synths and quantizing.

The three-CD set includes an audio CD for auditioning files and two CD-ROMs filled with REX and Acidized WAV files of drum loops, short instrumental phrases, and single-hit drum samples, such as kicks, snares, and hi-hats. Each folder on the first CD-ROM mingles the REX and WAV files associated with a song motif; the second CD-ROM carries a grab bag of compatible REX- and WAV-file loops and single-instrument hits.

In essence, all of the featured songs are loops, but the inherent flexibility of REX files lets you milk plenty of variation from smaller phrases. At first, the combination of formats may seem confusing—until you understand that the REX and WAV files are meant to be layered as individual track components. The instrumental phrases are not provided as REX files, but the WAV-file instrumental phrases are mostly short snippets that you can often move around in much the same way you can the REX-file constituents.

Domo Obbligato

The *Smokers Delight* files are imbued with a jazzy, hip-hop attitude; wah-wah guitar,

***Smokers Delight* from eLab offers a huge library of REX and Acidized WAV files that captures the funky sounds and styles of the '70s.**

Rhodes or Wurlitzer electric piano, and muted trumpet obbligatos add jazz and funk flavor. Many of the loops sound as though they were lifted from vinyl—right down to the accompanying pops and crackles. Noisy though they may be, the strength of these audio files is their atmospheric quality. Some people may like the vinyl ambience; I feel that vinyl noise carries the digital musician's fetish for analog artifacts a bit over the edge. I would prefer to have the noise in a separate audio file. Then you could use it, lose it, or mix it to taste.

I especially like "Homeys," with its Miles-like muted trumpet; subtle turntable scratches; bluesy, upper-register piano fills; and lightly swinging hi-hats. "Jellos" is similarly treated with muted trumpet but adds more dissonant keyboard comping for a much edgier, more "outside" tonality that clearly illustrates the difference between early '70s funk-jazz and the mostly pale attempts to reproduce it in today's smooth jazz.

Within Reason

I tested the tracks in MOTU Digital Performer 4.1 as well as in Propellerhead Reason 2.5. The files opened in Reason's Dr.Rex player without a hitch; opening the files in Digital Performer required dragging them to the hard disk and using a file-typing application to change the file's Creator to ReCy and its Type to REX2. (According to MOTU, this problem has been fixed in version 4.12, a free update.) A minor gripe is that the file names don't differentiate between mono and stereo; that bit of information could be helpful during production.

The *Smokers Delight* documentation and organization is helpful, but it falls a bit short. The audio CD plays each complete motif followed by its component files, and it covers much of the first CD-ROM. Unfortunately, it omits the 300 bonus samples on the first disc and all of the samples on the second CD-ROM. Considering the sheer number of files in this set, a second audio CD would have made a nice addition. The documentation lists every folder on the CD-ROMs, and all of the files are numbered consecutively, showing the instrument names and base tempos.

The things that I love about this collection tip the scales in favor of a hearty recommendation. Rarely have I heard a collection that evokes so much of the vibe and sound of the era of funk-jazz hybrids by Stevie Wonder, Creed Taylor, and others. The sheer preponderance of great loops combined with a price tag of less than $100 makes this library a terrific value. If you want to fill your sampler or loop-sequencing software with vital, authentic-sounding funk of the '70s, be sure to check out *Smokers Delight*.

Overall EM Rating (1 through 5): 3.5

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PRIMESOUNDS

A Funky Future (Audio/WAV/REX)

By Marty Cutler

Although PrimeSounds' *A Funky Future* ($99.95) may not fit everyone's definition of pure funk, it does deliver a fine collection of grooves, bass lines, and guitar-driven pads and effects.

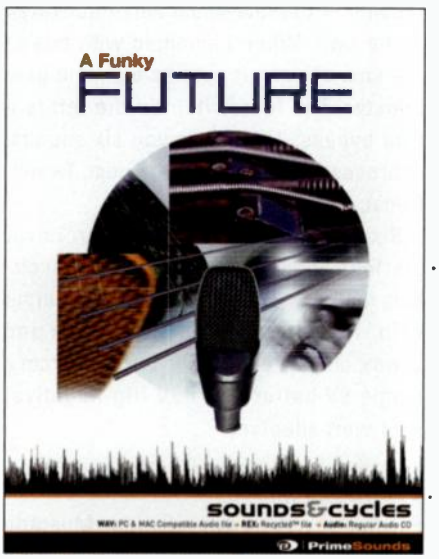
An accompanying audio CD lets you audition all of the files in the CD-ROM library. The documentation doesn't list the individual files and tempos; instead, it provides a breakdown by tempo of the drums, percussion, bass, and guitar loops—but without individual file names. That isn't the most thorough accounting of the offered loops, but it provides a rough guide to the contents of the audio CD and the CD-ROM.

Acid Flashback

The CD-ROM provides a collection of Acidized WAV and REX2 files, so even if your sequencer doesn't support REX files, you can still use the WAV-file loops. The files range from down-tempo feels of 80 to a more energetic 120 bpm, but the overall mood of the collection is dark, mysterious, and somber.

The drum, guitar, and percussion tracks rely on extensive processing, often leaving few clues as to the origins of the instrument sounds. Tracks are severely equalized, vocoded, filtered, distorted, and reshaped in other ways. Many of the percussion tracks are drum-machine grooves creatively processed through highpass filters to change the character of the loops, to remove low-frequency content, and to avoid interference with the drum grooves. Other tracks have minimally audible traces of acoustic percussion origins.

In some instances, the drum and percussion grooves use duplicate patterns, but different DSP treatments alter the feel and timbre. Because of the individualized processing applied to each loop and the varied amounts of swing, no two drum grooves sound or feel exactly alike. That



PrimeSounds' *A Funky Future* offers a varied collection of creatively processed grooves, bass lines, and guitar-driven pads and effects in WAV and REX formats.

can be a problem when linking different grooves together. Nonetheless, you can certainly extract a good deal of rhythmic flexibility from the REX versions by varying the amount of swing and altering the pitch, placement, and volume of individual groove slices. With a bit of creative-edge editing in my sequencer, I was able to layer the fill from a different loop over an otherwise disparate groove (see **Web Clip 1**).

The bass loops are, by comparison, relatively dry and free from processing, allowing for a more intimate, human sound. Three different basses provide a bit of sonic variety: Double (acoustic bass), Warwick (fretted electric bass), and Rubber (an electric fretless bass of unknown lineage). Audible fretboard-noise artifacts, release noises, squeaks, string buzz, and scrapes stand in sharp relief to the programmed drums and percussion.

When I first auditioned the bass loops in isolation, I was nonplussed by the looseness of the playing, but when I set them in tracks alongside the drum and percussion grooves, they fit perfectly. In general, I prefer the bass loops that are played at slower tempos; those performances take advantage of more subtle articulations and display a more refined sense of timing. At times the intonation is questionable, with bends and glissandos landing a bit flat or sharp, but even that adds warmth and humanity to the overall vibe.

Believe It or Not

It's hard to believe the guitar tracks weren't created with a MIDI guitar. Many are sparkling and evolving pads and drones with sprays of shifting harmonic content emerging and disappearing—sometimes over as many as eight bars. Other guitar loops clearly use tempo-synchronized filters, phase shifters, and vocoders for rhythmic compatibility with the percussion. Unfortunately, the guitar loops are not available in REX format. Granted, most of the guitar sounds are pads and drones, so their long tails make them difficult to divide into beat-sliced components. But many are palpably rhythmic and would benefit from the more flexible pitch-and-time-altering capabilities of the REX format.

A few file-naming anomalies and the skimpy documentation hardly blemish the overall quality of the *Funky Future* package. Because the bass and guitar loops provide the harmonic underpinning, it's truly unfortunate that the guitar parts are available only in WAV format; that locks you into a far more limited rhythmic and harmonic palette. Still, I'm impressed with the rhythmic and tonal variety found in this collection. The bass, drum, and percussion parts work as a great ensemble cast. If you compose dark-sounding ambient music, you should definitely investigate the moody atmospheres of *A Funky Future*. ●

EM WEB CLIPS

Overall EM Rating (1 through 5): 3.5

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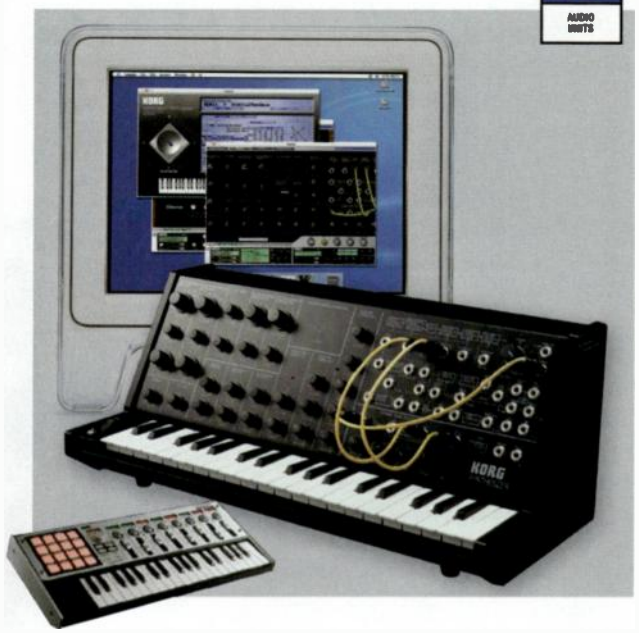

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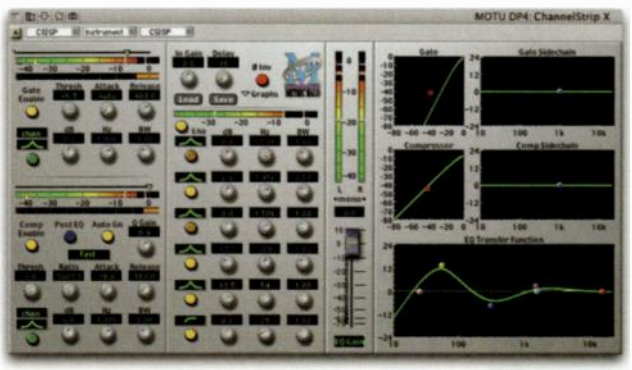

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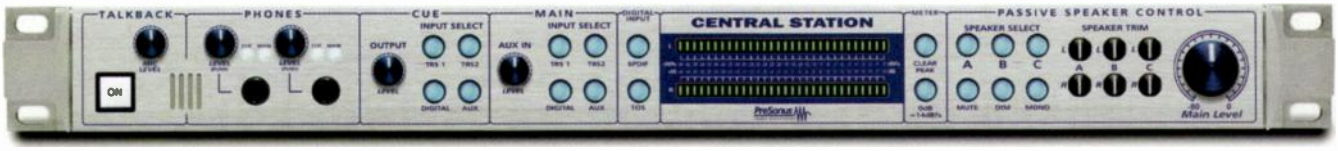

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By Larry the O

Where Do You Want to Go in This Life?

I advocate making excruciatingly detailed plans for projects, studios, and the like, using databases, diagrams, and lists aplenty. Yet I also advocate going with your gut when evaluating work, and following your creativity. After all, what's life if you don't take some risks?

Some people plan much of their lives in advance. That is a necessity for those who wish to pursue professions such as medicine or law, which require the fulfillment of a very specific sequence of achievements, concluding with the granting of a license to practice the chosen profession on a willing yet unsuspecting public. Such people move straight from high school into the prescribed, step-by-step process of obtaining their license. They commit much time and money to their pursuit, and their path is yet further determined by their need to repay student loans.

There are good things to be said for this approach. For one, a person does not have to expend thought and energy sorting through what one wants to do; that much is established. This minimizes distraction and maximizes focus, which is requisite for leaping the substantial hurdles of these paths. Another positive is that these professions tend to increase an individual's opportunities for achieving financial success.

The opposite extreme can be seen by looking at those people who move from one path to another depending on which way the wind blows them. Those individuals are not always drifters, either; some of them are bright and motivated but find that their interests are turned by tides of history and circumstance.

The advantages to this way of living are that one often acquires a broad perspective and a variety of skills. If you have been a sailor, a stockbroker, a cheesemaker, and an orchestra manager, you have experienced more points of view than most people do. Eventually, a person may settle into one path and pursue it with a frame of reference that is far more broad than those held by others in the field. In fact, such broad experiences could well lead a person to make some great breakthrough in his or her field. Conversely, he or she might keep moving from business to business, and never put down roots in any particular field.

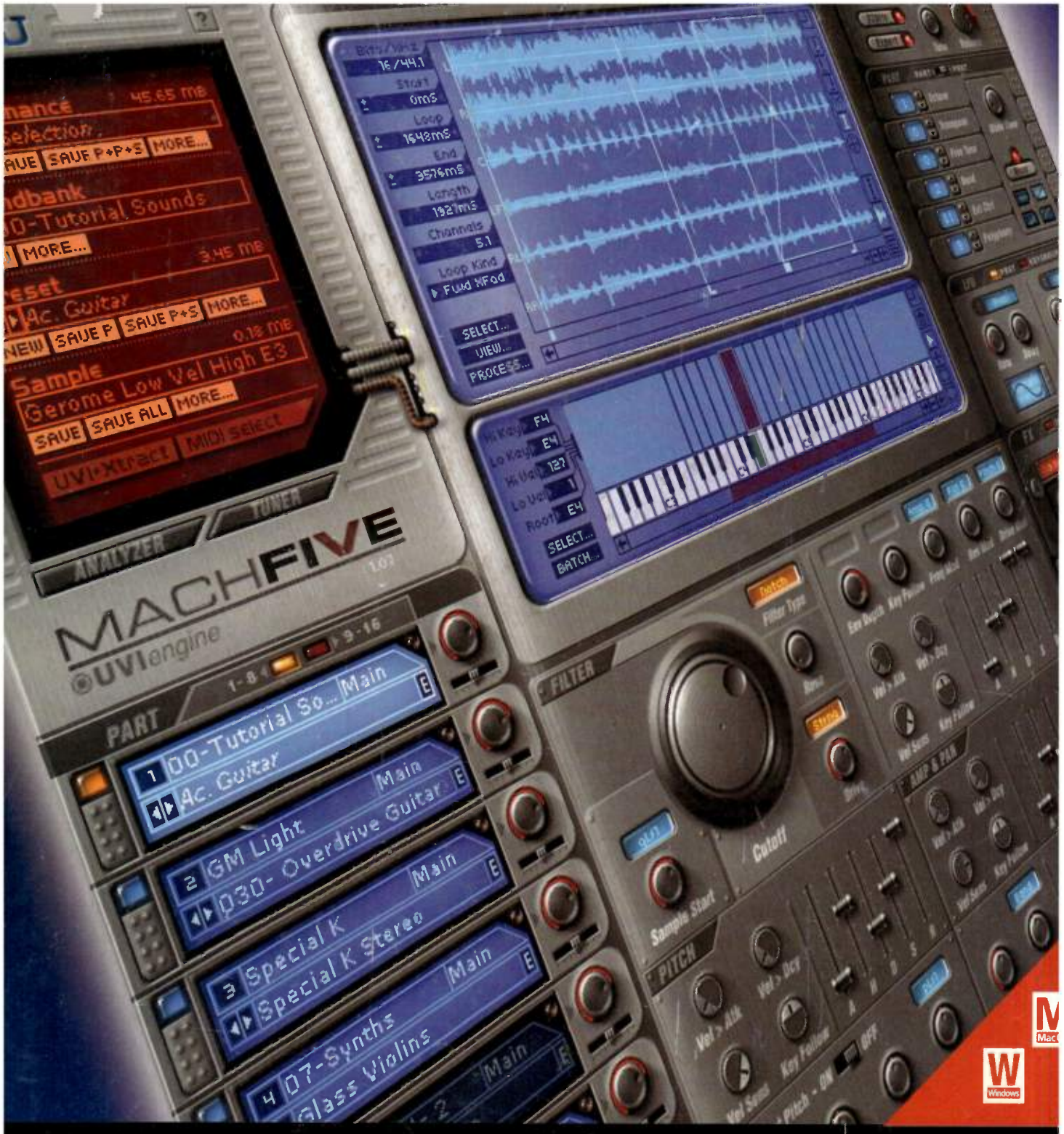
In my life, I have fallen somewhere in the middle of those two extremes. When I was a teenager, I discovered that I had an intense interest in music. Shortly thereafter I discovered what would become my entrée into audio: synthesizers. The strength of my attraction to music acted as a focusing influence akin to, but not as rigid and formalized as, a decision to pursue law or medicine.

At the same time, I've tended to follow interesting opportunities that have presented themselves until they ceased to be interesting, at which point I would strike out in another direction. Through luck and skill, I have never lacked for interesting opportunities, but I have sometimes felt that I was doing little more than drifting.

The part that always struck me as curious was how often my career appeared to be following a logical and well-timed path. So many times I have answered a knock at the door and found myself in the middle of a situation that was just coming into fruition. Interestingly, in hindsight, I seem to have had clear vision all along the way. If I did have such vision, I was not aware of it at the time. Regardless, after some years of this, I have come to believe in myself, which has served me well more often than not.

The moral of this discourse is that each person is ultimately responsible for finding his or her own path in his or her own way, and it is best to be unafraid to do so. If you are someone that flourishes only in structured situations, observe, embrace, and pursue that. Strive for balance, yes, but don't fight your nature. If you are simply unable to lock yourself into one direction, figure a way to make that work, perhaps by freelancing in a few different areas.

Examining one's direction should be an ongoing effort, and questioning oneself helps avoid becoming close-minded and stuck in a rut. Self-evaluation and self-doubt, however, need not go hand in hand.



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