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DECEMBER 2006

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Electronic Musician
December 2006

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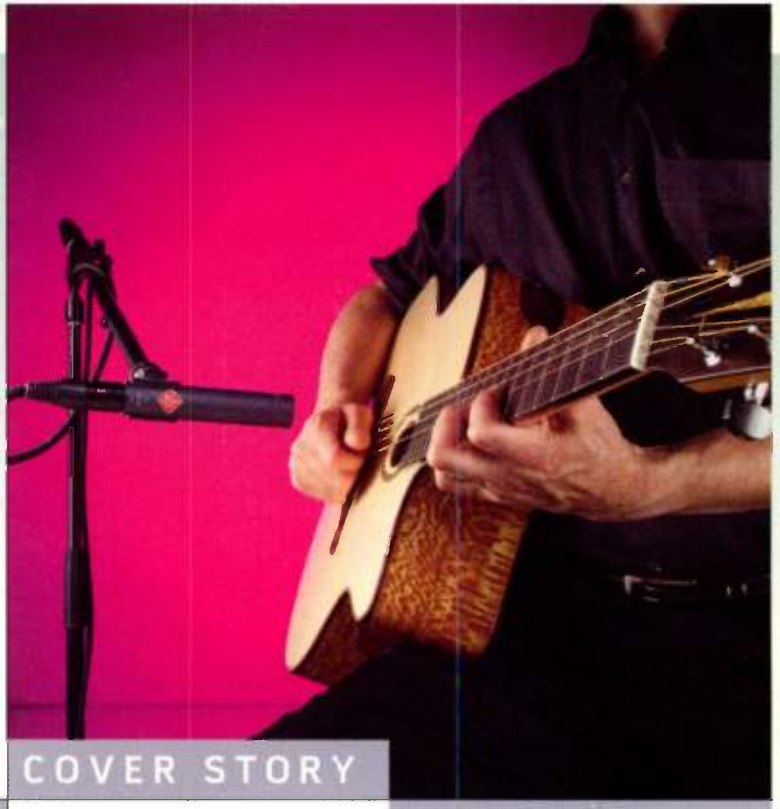

PHOTO BY BOB MONTESCLAROS; MODEL: BOB GAMBLE; GUITAR BY BEARDSELL

COVER STORY

FEATURES

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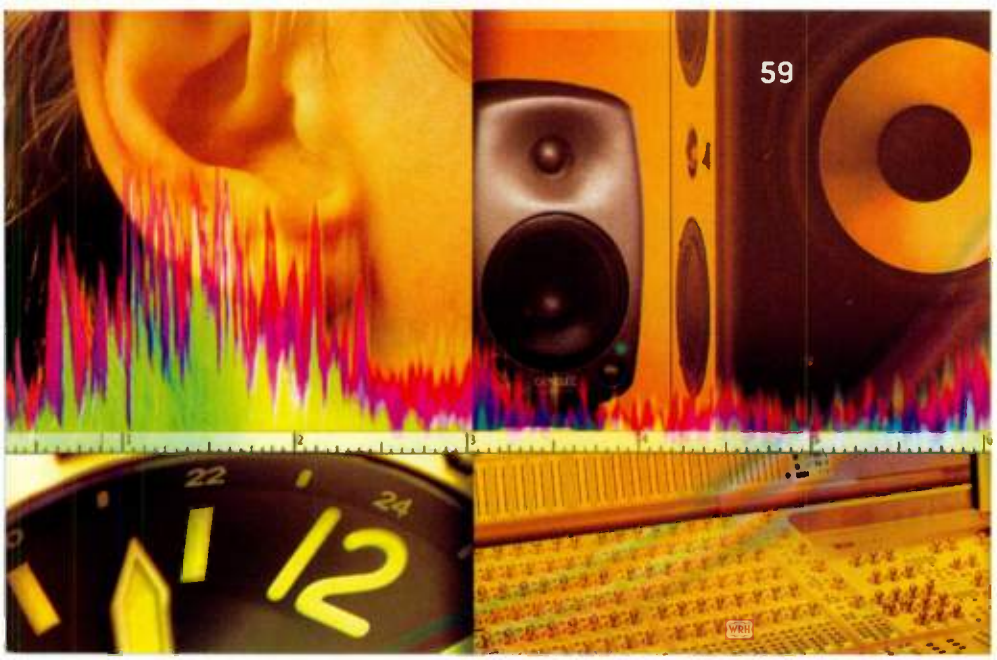

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Graph Tech Ghost Modular Pickup System MIDI guitar pickup

SoundTech LightSnake USB instrument cable

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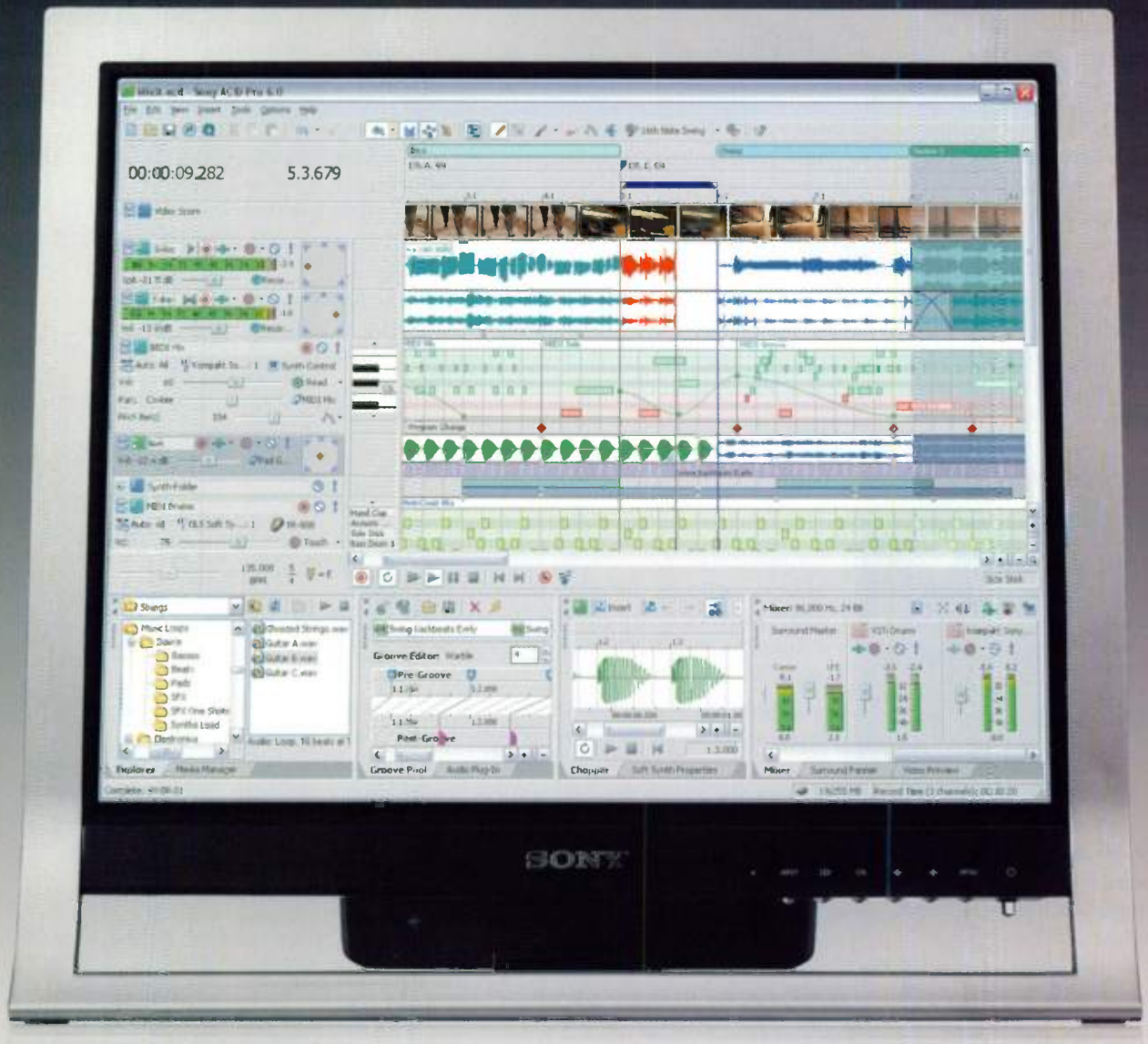

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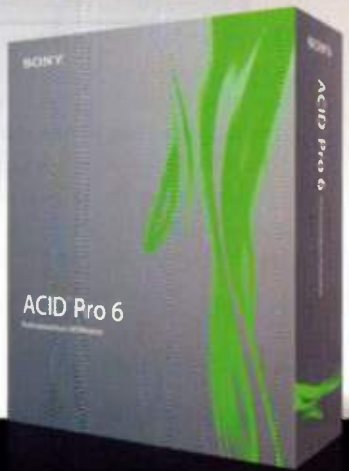

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+ + = 6

AES Online

The 2006 AES show is history, and as usual, we had a blast seeing all the new tools and catching up with old friends. Between product demos, press conferences, and preparing our coverage, we were busier than ants at a picnic. And a fine picnic it was! We saw lots of new applications of existing technology, a few new directions, and plenty of just plain good stuff. To get the whole scoop, read our online show report at www.emusician.com.

This year we produced a cool blog from our AES booth with sister publications *Mix* and *Remix*. Even better, the three editorial staffs coproduced a series of Podcasts and videos that present interviews and product demos captured live at the show. Most of these Podcasts and videos focus on manufacturers explaining their new wares, but some of my favorites are interviews with luminaries such as George Massenburg, Elliot Scheiner, and Geoff Emerick. If you want to know what these legends thought were cool at the show, point your browser to our site.

JULIE BIRUM

A quick note: the AES Podcasts and videos are posted on *Mix*'s Web site (www.mixonline.com), but you also can access them from our site via links; they are, in fact, a joint venture. The Podcasts were edited by *Mix*'s Kevin Becka. The videos are called *Tech Minutes*, and they were shot and edited by Jonathan Sontag and directed by various editors. I highly recommend you check them out.

Producing blogs, Podcasts, and videos is far from being a one-time experiment. *Mix* has created show blogs in the past, including a joint blog with EM and *Remix* at the 2006 Winter NAMM show. This past AES, however, was the first show where we emphasized Podcasts and videos, and we'll have much more electronic coverage at future trade shows, starting with the upcoming Winter NAMM show in January 2007. During the coming year, we'll present a wide variety of mixed-media coverage that will go well beyond trade-show coverage.

As part of our increased online and new media efforts, we have undergone some internal reorganization. For starters, former group publisher Dave Reik has taken on a new challenge that includes directing our circulation efforts and our fast-growing online Bookshelf store, but most important, involves developing some Very Big Plans I can't tell you about yet. Let's just say that sometime around the end of this year or in early 2007 we're going to make you an offer you won't want to refuse. On the editorial side, to help lead *Electronic Musician*, *Mix*, *Remix*, *Music Education Technology*, and *Live Design* into the brave new world of online multimedia information delivery, former *Mix* editor in chief Tom Kenny has been promoted to editorial director. Senior VP Pete May will act as interim group publisher.

The bottom line is that we're retooling, working more closely with our sister publications, and gearing up to provide the best recording and sound-design information available. We feel that it's a great way to end 2006 and launch into 2007!

Steve Oppenheimer
Editor in Chief

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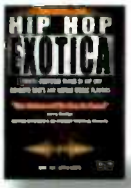

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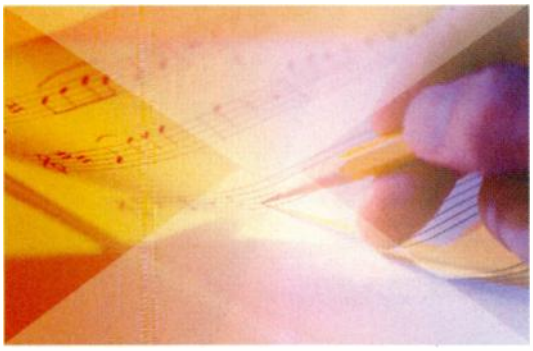

DMITRY PANICH

Letters

Virtually Ridiculous

Which virtual pianos sound the best? (See "Software Eighty-Eights" in the October 2006 issue of EM.) Which analog emulation software sounds like the real thing? Which software synthesizers? I would say that the real thing sounds like the real thing! Perhaps I'm one of the few lucky people who have three analog synths and more, which are often hard to find. However, for the price of some of this software, one could afford to find a recording studio with the real thing—pianos and all.

Marcel Peelen
via email

Marcel—I have neither the money nor the space for a grand piano. I do have a good 88-key MIDI keyboard, though. So I am a fan of soft pianos like those described in "Software Eighty-Eights." I do have some classic analog synths, but maintenance can be expensive, and I sometimes want sounds that are typical of classic synths I don't own. Soft instruments solve that problem and may provide capabilities not found in the originals. I don't expect a software emulation of a Minimoog to sound exactly like a hardware Mini, but some soft instruments come amazingly close. Besides, no two hardware Minimoogs sound exactly alike either.

And sure, there are times when it's best to use a commercial studio that has a fine instrument. Nothing beats the action and responsiveness of a great piano. However, upgrades aside (you don't have to buy every upgrade), you pay for your soft instruments once, and they are available in your studio every day. —Steve O

Mac Mediation

Wow! Great article! I happened across "Should You Buy an Intel Mac?" in the September 2006 issue of EM just as I bought my MacBook for the purpose of using both operating systems. Since my Mac has Intel processors, I decided to try the Boot Camp thing. It works almost perfectly, but I still have to work out the bugs.

I use a FireWire audio device, and whether you boot up with it plugged in seems to affect Pro Tools' ability to use the hardware properly. When I plug it in after booting up, it runs. When I plug it in before, Pro Tools throws out errors left and right. At times, I also disable the stock sound card in the device manager section of Windows, even though the stock sound card has built-in digital audio capabilities that can be used with most of my audio programs in Windows.

Anyway, as described in the article, I partitioned the drive. I created my Mac drivers install disc. I loaded Windows XP with SP2. All of my programs and plug-ins worked. So now I have a MacBook with all of my Mac- and PC-based programs to go mobile with.

Thanks again!

Joel Simpson
via email

Work Ethic

I just read Larry the O's comments on obstacles to work flow (see "Final Mix: Fragged" in the July 2006 issue of EM), and I agree with him. On the one hand, obstacles that allow ear and perspective breaks are good, while on the other hand, I feel that control surface programming must be open source. I imagine that some controller manufacturers have some sort of programmable interface. Anything that can be mouse driven should be accessible by controller. Users who have a need will make the time to program an interface if manufacturers or skilled programmers/hackers provide a toolkit.

One difference between analog then and digital now is that users are not married to one product the way they would be to a console and tape machine. Instead, people will use whatever gets the job done—and that means going from program to program to use tools or plug-ins.

Eddie Ciletti
via email

Error Log

October 2006, the cover. The line referring to Duncan Sheik as a "Grammy winner" was incorrect; he was a nominee. In the article itself ("Yours, Mine, and Ours," p. 67), Sheik's nomination was incorrectly referred to as "Best Male Vocal" rather than "Best Male Pop Vocal." EM

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Next Month in EM

COVER

Editors' Choice Awards

EM's prestigious Editors' Choice Awards are given to the best new products tested by EM editors and authors in the past year. Get the lowdown on the good stuff.

FEATURES

Recording a Live Act

Get practical advice on how to record live bands and singer-songwriters. Multitracking and recording stereo from the room are discussed. Also covered are studio-mic setups ("live in the studio" recording) and live-mic setups.

Master Class:
Programming for MIDI Guitar

Learn to program synthesizers for optimal MIDI guitar performance.

COLUMNS

Making Tracks:
Rigged Up

Check out our tips for using guitar effects on nonguitar tracks.

Sound Design Workshop:
Golden Gates

We discuss creative ways to use noise gates with sidechains.

Square One:* *The Secret Life of a Sampler, Part 2

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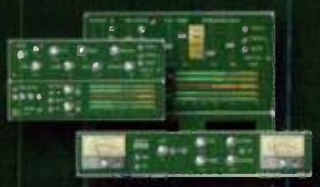

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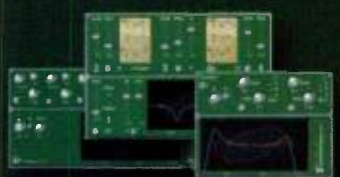

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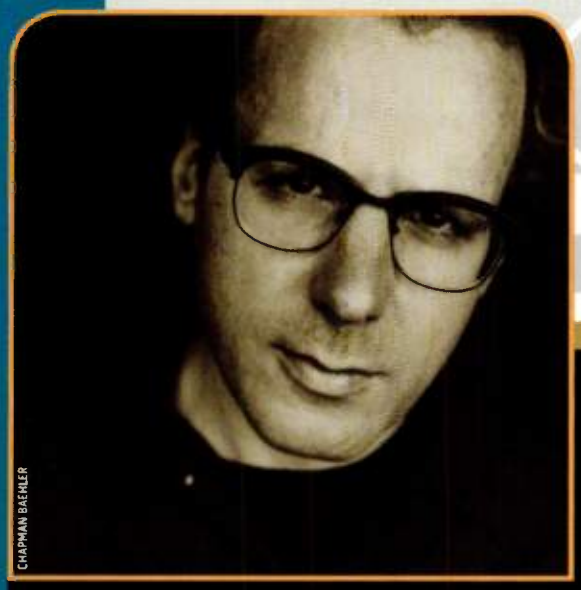

EMspotlight

Froom Lets the Song Do the Talking

With more than 60 albums under his belt, Mitchell Froom has made a name for himself as a producer and musician. His production credits read like a who's who of rockers—Elvis Costello, Los Lobos, Suzanne Vega, Paul McCartney, the Latin Playboys, and Richard Thompson. In this interview from the EM archives, Froom talks about production concepts and his organic approach to helping artists sound their best. By Michael Molenda. emusician.com/em_spotlight

On the Home Page

EM Web Clips

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

EM Guides Online

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

EM Show Report

The Audio Engineering Society (AES) convention is one of the largest annual pro-audio expos in the United States. Visit emusician.com for an exclusive look at the new recording gear and music software unveiled at this year's show.

EM get on the bus

The Bus is the exciting new blog by EM editors Steve Oppenheimer, Mike Levine, Gino Robair, and Geary Yelton. Found exclusively on emusician.com, the Bus will include tips and tricks on a variety of recording topics, cover the latest technologies and industry trends, and offer a behind-the-scenes look at trade shows, product demos, and the creation of *Electronic Musician* magazine.

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By Geary Yelton

sE Electronics USB 2200a

Slated to ship before the new year, the USB 2200a ($599) is a groundbreaking USB condenser mic from sE Electronics (www.seelectronics.com). It has the same capsule, voicing, low-cut filter, and –10 dB pad as the well-regarded SE2200a. But unlike other microphones, the USB 2200a incorporates numerous unique features, such as a built-in mix control that allows zero-latency headphone monitoring for live recording and playback. Additionally, the mic allows you to record an analog signal by using its XLR connection at the same time you record digital audio using its USB connection.

sE Electronics reports that it has developed proprietary integrated circuits and software for a variety of USB recording products, and the USB 2200a is the first of the new line. The initial problem to solve was how to incorporate sufficient voltage conditioning to convert USB's 5V power into 48V to power the capsule without increasing noise. sE claims success over that challenge, resulting in quiet performance for professional recording applications. The USB 2200a, which has a 1-inch diaphragm and a transformer-balanced output, ships with a shockmount cradle.

Line 6 Floor POD

For guitarists in need of an all-in-one amp modeler and multi-effects processor, Line 6 (www.line6.com) is currently shipping the Floor POD ($279). Designed to be affordable without sacrificing professional tone, the Floor POD pairs a dozen of the most popular studio-grade amp models from the POD 2.0 with a built-in foot controller and seven Smart Control effects types, any two of which are available simultaneously. In addition to offering hands-free parameter adjustment, the Floor POD's foot controller operates as a wah or volume pedal. Standard effects such as compression and distortion supplement the Smart Control effects, which include reverb, delay, sweep echo, chorus, phaser, flanger, and tremolo.

The versatile Floor POD can be used as a compact pedalboard for your guitar amp, as an effects processor routed direct to your mixer, or with headphones and a portable MP3 or CD player for private rehearsal. The Floor POD supplies 64 presets, 3 footswitches, 2 unbalanced ¼-inch outputs, and an onboard tuner. Its heavy-duty construction should ensure long life and durability.

PreSonus FaderPort

You say that pointing-and-clicking just doesn't do it for you anymore? Do you long for a real long-throw fader and solid transport buttons but feel that a full-size control surface takes up too much space? Well, PreSonus (www.presonus.com) has a new desktop USB controller that's designed with your fingers in mind. The FaderPort ($229) supplies a touch-sensitive, motorized, 100 mm fader for recording smooth fades and writing automation in real time. A dedicated Pan knob and an assortment of buttons give you hands-on control of Mac OS X and Windows XP applications such as Steinberg Cubase, Digidesign Pro Tools, and Apple Logic. Additionally, a footswitch jack offers hands-free punch-in and punch-out recording.

The FaderPort lets you control all transport functions; write single-channel or group-channel automation; quickly select project, mix, and transport windows; and enable mute, solo, and record functions. Illuminated buttons let you define and manage markers, loop between two points, undo and redo, and navigate between individual channels or eight channels simultaneously. The FaderPort comes with an external 9V power supply that is required for the fader motor; all other functions can be powered by USB.

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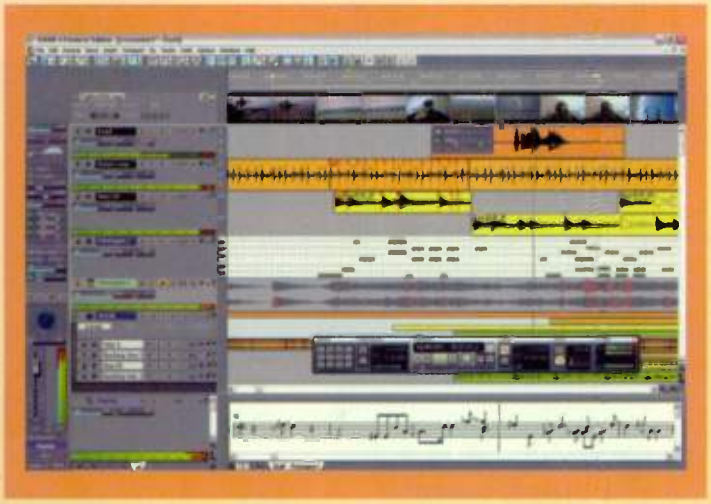
CAKEWALK SONAR 6

Cakewalk (www.cakewalk.com) has made some significant changes to its digital audio sequencers, Sonar 6 Producer Edition (Win, $619) and Sonar 6 Studio Edition (Win, $369). Both versions support 64-bit double-precision audio processing and introduce Active Controller Technology (ACT), which dynamically remaps host and plug-in parameters to hardware controllers. You can customize Sonar 6's user interface by making frequently used features more accessible, altering color schemes, and organizing plug-in menus to suit your preferences. AudioSnap functionality offers nondestructive audio quantization, allowing you to apply seamless tempo changes, stretch clips to new lengths, and lock multiple tracks to the same groove. You can replace recorded drums by converting audio beats to MIDI data. New Synth Rack functionality simplifies working with virtual instruments, and inserting a software instrument can group audio and MIDI tracks into a Synth Track folder. Navigation, automation, editing, mixing, and audio performance all benefit from numerous enhancements.

In addition, Sonar 6 Producer Edition comes with effects and instruments such as VC-64 Vintage Channel, which is a virtual channel strip that has dual EQ and compressor stages and professionally designed presets. Also included is Session Drummer 2, which supplies multisampled drum kits and hundreds of patterns played by live drummers. Producer Edition also has multitrack AudioSnap. Upgrades to Sonar 6 Studio Edition start at $99, and upgrades to Sonar 6 Producer Edition start at $179.



STEINBERG CUBASE 4

Cubase SX and SL have gone the way of the dinosaurs. Steinberg (www.steinberg.net) has replaced them with Cubase 4 (Mac/Win, $999) and Cubase Studio 4 (Mac/Win, $499), which offer new capabilities and enhanced work flows for music production. Perhaps the biggest new feature is SoundFrame, a universal media manager that helps you organize, preview, and retrieve resources such as track and instrument presets and audio, video, and project files. Support for the VST 3 protocol, which improves preset management and optimizes computer resources, is standard. A redesigned graphical user interface improves navigation and lets you customize channel strips and the Track Inspector. Expanded music notation adds new features for scoring and editing printed music. And you can quickly create Instrument Tracks, which combine a VST instrument, audio, and MIDI data with a single command.

Cubase 4 comes with more than 30 new VST plug-ins, including StudioEQ, Multiband Compressor, AmpSimulator, and ModMachine. You also get four exclusive virtual instruments: Prologue, Spector, Mystic, and HALion One. With the appropriate hardware, Cubase 4's new Control Room functionality supports talkback capabilities, speaker selection, and as many as four independent monitor mixes. For users with less demanding requirements, Cubase Studio 4 provides a streamlined feature set with full scoring capabilities, a limited plug-in bundle, and the instruments Prologue and HALion One.

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Tascam GVI

Giga Virtual Instrument (Win, $399), the long-awaited sampler plug-in from Tascam (www.tascam.com), is now available for Windows XP (a Mac version is in the works). GVI runs on VST and RTAS hosts and offers access to the entire library of sounds for GigaStudio 3 and earlier versions. GVI is 16-part multitimbral and supports 24-bit sampling rates of up to 96 kHz. It plays samples programmed with embedded GigaPulse convolution reverb and iMIDI rules for controlling phrasing and articulation. According to Tascam, the number of simultaneous instances and polyphony are limited only by your computer's power.

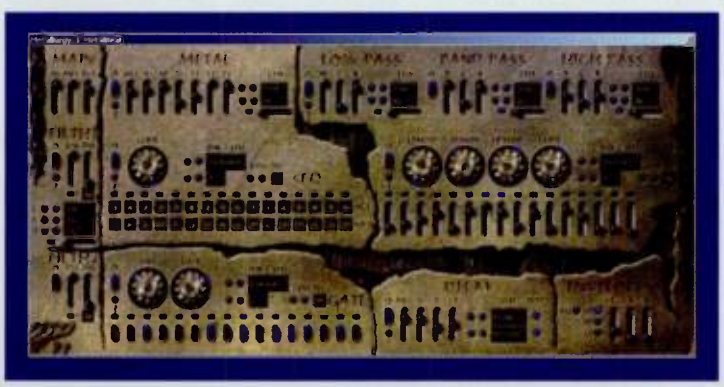
Like its costlier high-end brethren, GVI features comprehensive QuickEdit capabilities, multimode filters, multisegment envelopes, and complex modulation programmability. The plug-in lets you automate parameters such as tuning, volume modulation, and Velocity response. GVI ships with a factory sound set that furnishes a broad palette of samples from SoniVox, Best Service, Larry Seyer, and Bigga Giggas.

Download of the Month

UGO AUDIO METALLURGY 1.0 (WIN)

Metallurgy ($20) is an effects processor with lots of attitude. Its designer, Chris Sciuba (aka Ugo; pronounced *oo-go*, not *yu-go*), is well known for his free VSTi synths String Theory, Texture, Rez, and Motion, which are all built in the SynthEdit programming environment. You can download the free synths and a demo version of Metallurgy, along with a limited number of patches and occasional volume dropouts, from www.ugoaudio.com. If you like what you hear, there's a PayPal link for purchasing the full version of Metallurgy.

Metallurgy was designed with synth leads, basses, and drums in mind, but as its 400 factory presets demonstrate, it can effectively mangle all kinds of material (see **Web Clip 1**). Its signal path starts with a distortion and downsampling effect called Filth 1, which is followed by lowpass, bandpass, and highpass filters in parallel. The filters are mixed and followed by Filth 2, which is identical to Filth 1. Each filter has its own LFO, and you can route a 16-step sequencer to modulate each filter's cutoff frequency.



Metallurgy's most unusual module is a very short delay line called Metal, which produces flanging and resonator effects. You use an accompanying 16-step pattern sequencer to control the resonator's pitch. An LFO is also available to modulate the delay time. Beyond being a flexible flanger, Metal can produce pitched step sequences.

The signal path ends with a 16-step gate sequencer followed by two feedback delay lines. The gate sequencer has variable gate length and wet/dry mix controls. All three of Metallurgy's step sequencers are synced to host tempo; have variable length from 1 to 16 steps; and will loop, reverse loop, or bounce. If you crave more motion than five LFOs and three step sequencers provide, you can use Metallurgy's envelope follower to modulate any of the filter cutoffs and Metal's time parameter.

Beyond Metallurgy, the four free synths are all worth a listen. String Theory is a physical-modeling synth offering a variety of sounds ranging from plucked strings to synthy pads. Texture creates metallic, digital ambient textures. Rez is a monophonic, virtual-analog lead synth. My favorite, Motion, is a 2-oscillator, 2-filter synth that emphasizes step-sequencer, LFO, and envelope modulation. You're bound to find something useful at the Ugo Audio Web site.

EM WEB CLIPS

—Len Sasso

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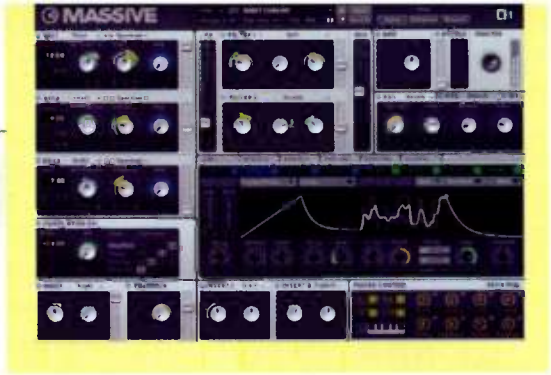
Native Instruments Massive



Native Instruments (www.native-instruments.com) has debuted Massive (Mac/Win, $339), the newest member of its extensive family of soft synths. At the heart of Massive's high-resolution audio engine are three unique Wave-Scanning oscillators, powered by 82 wavetables that were enhanced using FFT resynthesis. The oscillators, combined with ring and phase modulation, filter FM, special phase-shaping modes, and noise generation, furnish timbres that extend from rumbling basses and screaming leads to complex pads and evolving textures. And with internal sampling rates as high as 192 kHz, Massive's sound is free of aliasing and other undesirable artifacts.

Massive's 2 filters encompass 11 filter types and a feedback stage that you can insert at various points in the signal path. Flexible routing lets you add polyphonic insert effects and make modulation assignments by simply dragging-and-dropping. Two master effects sections offer an extensive selection of algorithms. Four assignable ADSR generators support looping and morphing, and you can redefine four custom modulators at will. Massive incorporates an integrated KoreSound browser that lets you filter and search the 420 included presets by specific attributes. The synth runs standalone or as a plug-in in a variety of formats.

Terratec Producer Komplexer

Although its production stopped in 2002, the Waldorf MicroQ remains one of the most highly regarded of German hardware synths. Now, its technology lives on and continues to grow in Komplexer (Win, $249), developed by Terratec Producer and distributed by Synthax (www.synthax.com). Komplexer blends analog emulation with wavetable synthesis and delivers four stereo layers with 256-note polyphony. The VST-compatible plug-in imports sounds designed for the MicroQ, and users can also load their own wavetables.

With three audio oscillators, four complex envelope generators, three syncable LFOs, and two multimode filters (each with eight resonant filter types), Komplexer wields serious sound-sculpting power. It has a sophisticated arpeggiator, an audio input with a vocoder, and five independent effects. Komplexer's 16 × 16 modulation matrix can access dozens of parameters as sources or destinations. You can assign each of eight macro controllers to as many as four parameters simultaneously and assign external MIDI controllers to change parameters in real time. All Komplexer functions respond to sequencer automation.

Digidesign Mbox 2 Pro

Digidesign (www.digidesign.com) is now shipping Mbox 2 Pro ($799), a portable system that comprises a 24-bit, 96 kHz audio and MIDI interface with FireWire connectivity, the latest version of Pro Tools LE (Mac/Win), and a comprehensive suite of effects and instrument plug-ins. The system comes with Pro Tools Ignition Pack, the virtual instrument Xpand, and more than 50 DigiRack and Bomb Factory plug-ins.

Mbox 2 Pro's compact interface boasts features such as two high-quality mic preamps, a stereo phono preamp, two headphone outputs with independent volume controls, BNC word-clock I/O, and support for MIDI Time Stamping. Connections include 16-channel MIDI I/O, coaxial S/PDIF I/O, four balanced ¼-inch line outputs, two unbalanced ¼-inch monitor outputs, an unbalanced ¼-inch stereo line output, two balanced combo XLR/¼-inch mic inputs with 48V phantom power, two balanced ¼-inch line inputs, two ¼-inch high-impedance inputs, and two RCA inputs for a turntable.

An additional $100 buys Mbox 2 Pro Factory, which supplies an iLok USB key and five additional plug-ins. Purchased separately, the five premium effects from Digidesign and Bomb Factory would retail for more than $1,000. EM

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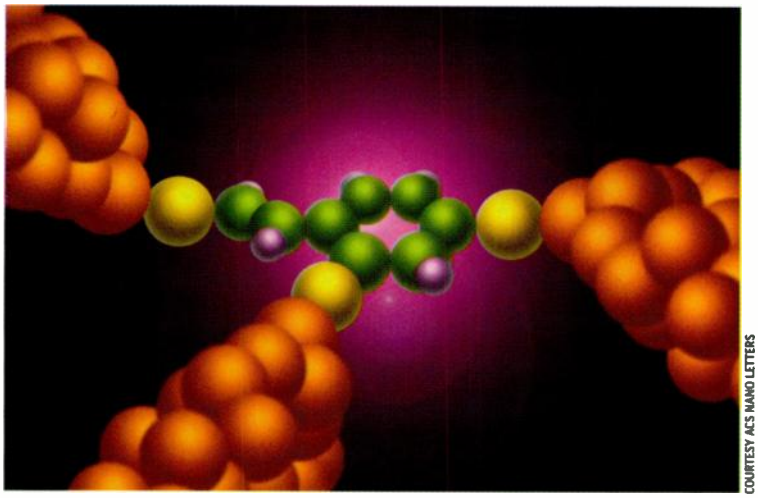
QuIET Please

By Scott Wilkinson

Molecular transistors are on the horizon.

As most tech-savvy folks know, the ongoing shrinkage of electronic components cannot continue forever. At some point, conventional transistors will succumb to quantum effects, rendering them useless in integrated circuits. So here's a novel idea: why not use those same quantum effects to facilitate molecular-sized transistors? That's just what a team of physicists at the University of Arizona (www.arizona.edu) is working on. Research on the Quantum Interference Effect Transistor (QuIET) aims to overcome the limitations placed on conventional transistors by using the wave properties of electrons to turn current flow on and off.

"All transistors in current technology, and almost all proposed transistors, regulate current flow by raising and lowering an energy barrier," says UA physicist Charles A. Stafford. "Using electricity to raise and lower energy barriers has worked for a century of switches, but that approach is about to hit the wall."



COURTESY ACS NANO LETTERS

FIG. 1: In this rendition of the Quantum Interference Effect Transistor (QuIET) by artist Helen Giesel, the central benzene ring consists of six carbon atoms (green), accompanied by atoms of hydrogen (purple) and sulfur (yellow). The three structures of gold-colored atoms represent the three metallic leads of the transistor.

Transistor elements in current ICs measure about 65 nanometers (1 nm is a billionth of a meter) across, and shrinking them smaller than about 25 nm is impractical. Three years ago, UA physicists Stafford, Sumit Mazumdar, and David Cardamone began thinking about how to use quantum mechanics to regulate the flow of current through a single-molecule transistor measuring as small as 1 nm.

The trick is to use the wavelike nature of individual electrons. As with sound and water waves, electron waves interfere with each other. When they are in phase, the interference is constructive, resulting in greater amplitude. When they are out of phase, they interfere destructively, causing the amplitude to decrease.

"Our approach is more finesse than brute force," says Cardamone. "We don't put up a wall to stop current. We regulate how electron waves combine to turn the transistor on or off." The initial iteration of this idea uses a benzene molecule, which consists of six carbon atoms in a ring (see **Fig. 1**). Other researchers have succeeded in attaching two leads to such a molecule, but the UA team expects three-lead devices to be just around the corner. That will allow one lead to act as a valve for current flowing between the other two.

"In classical physics, currents through each arm of the ring would just add," Stafford explains. "But quantum mechanically, the two electron waves interfere with each other destructively, so no current gets through. That's the 'off' state of the transistor." The transistor is turned on by changing the phase of the electron waves so that they interfere constructively.

According to Stafford, "It took a while to go from the idea of how this could work to developing realistic calculations of this kind of system. Within a few weeks, we were able to do the simplest kind of quantum chemical calculations that neglect interactions between different electrons. But it took some time to put in all the electron interactions that demonstrate that this really is a very robust device." The team has determined that the QuIET has characteristics very similar to conventional field-effect and bipolar-junction transistors.

The QuIET research could result in much smaller, more powerful computers and other devices with the ability to process many more channels of high-resolution audio and video than today's music products can manage. Another possibility is the emergence of integrated devices that combine several advanced functions into one tiny unit with far more power than anything currently available. How ironic it is that a technology called QuIET could help make a joyous noise. EM

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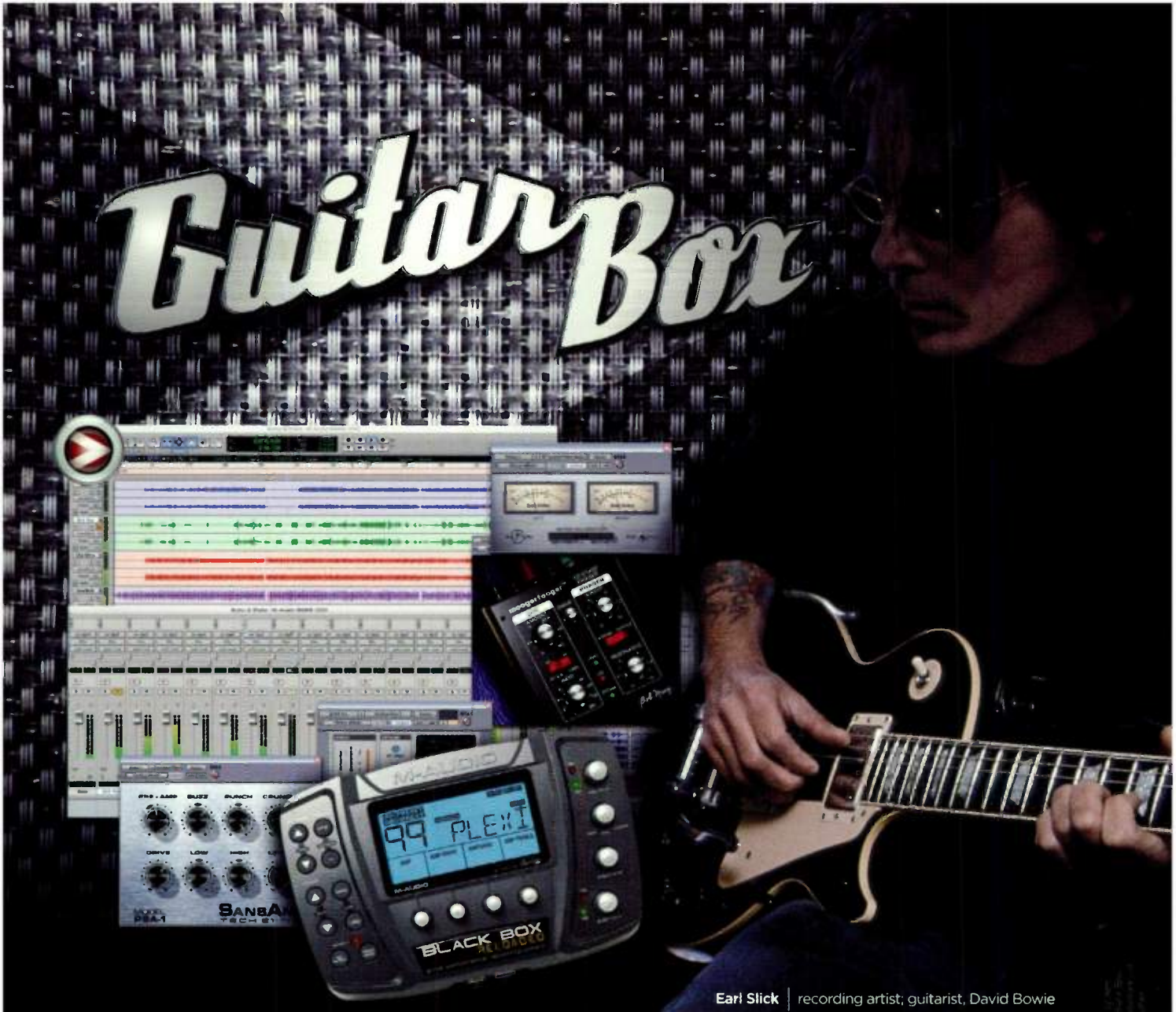

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Direct Results

By Mike Levine

Few would dispute that the ultimate electric guitar tones are recorded from miked-up tube amps. That is, if you have a nice amp you can crank up, a good-sounding recording space (if you're using room mics), and time to experiment. But what if you don't? Or what if you have limitations on volume in your studio, don't want the amp sound to leak into other mics, or don't want to decide on the final guitar sound until mixdown? In those cases, direct recording may be the best alternative.

In recent years, amp- and cabinet-simulating modeling processors and plug-ins have made the task of getting a realistic tone from a DI (direct injection)–recorded guitar much easier. Those products let you process your guitar signal (during either tracking or mixdown) with amp, preamp, cabinet, and effects emulations. Plug-ins like Native Instruments Guitar Rig 2 (see **Fig. 1**), IK Multimedia's AmpliTube 2 (see **Web Clip 1**), and Waves GTR 2.0 (reviewed in this issue) not only give you a large selection of amp, cabinet, and effects models, they also let you specify the routing of your virtual components. Even though modeling devices and plug-ins may not be able to precisely match the sound of a boutique or vintage tube amp, they offer a much wider range of tones.

Record excellent guitar tones without miking a cabinet.

DI IQ

At its most basic, a DI setup contains a guitar patched into a digital audio sequencer or a multitrack through a high-impedance guitar input in an audio interface or a mic preamp, possibly with the signal first going through a direct box. In order to accurately capture the tone of your pickups, it's important to use an input with an impedance that's optimized for guitar. Guitar inputs on many preamps, audio interfaces, and DI boxes have a 1 MΩ input impedance, which is

the same value found on the inputs of classic tube amps from companies like Vox, Fender, and Marshall. If your input device has an impedance substantially below 1 MΩ, you may notice a rolling off of high end. Impedances above 1 MΩ can yield a brighter, more sparkly tone.

If your audio interface doesn't have a dedicated instrument input, you'll probably want to use a direct box. If you're recording through a mixer, a direct box is usually the best way to go too. A direct box converts your high-impedance, unbalanced guitar signal into a low-impedance, balanced signal that's output on an XLR connector and can be patched into one of the mic preamps on the mixer or audio interface. (Most of the time you'll want to go through a mic preamp, but some DI boxes, such as Groove Tubes' Ditto Box, provide sufficient gain to patch into a line input.) It also outputs the signal through a ¼-inch pass-through jack for connection to an amp or guitar processor if desired. On some direct boxes, this output is buffered, which allows for longer cable runs without signal loss.

On any direct box, the main XLR, low-impedance output gives you the ability to send your signal unaffected through long cable runs, such as in a studio with tie lines. The main output is also balanced, which can help prevent noise on the line.

FIG. 1: Native Instruments Guitar Rig 2 exemplifies the trend in guitar amp plug-ins to let you configure not only which components to use, but their routing too.

DI boxes come in two basic flavors. *Active* direct boxes contain active electronics, including a buffer amp that lowers the impedance of the incoming signal. These devices require either battery, 48V phantom, or AC power. *Passive* direct boxes alter the impedance by way of an internal transformer and require no power.

While many DI boxes simply convert your signal, others offer additional features. For instance, the Behringer GI100 adds speaker simulation. Radial Engineering's JDV active direct box (as well as several of the company's other devices) has a circuit called Drag Control, which is designed to re-create the relationship between a guitar pickup and a tube amp in situations where you're using a solid-state input device. It gives you a knob that alters the impedance load to your pickups, letting you dial in the most natural sound. Radial also offers a standalone Drag Control product called the Dragster.

Quality Is Key

Not all DI boxes, preamps, and other input devices sound the same. Just as different mic pres yield different sounds when fed by the same mic on the same source, different input devices give you different sounds from the same guitar. The quality of the input device's components is a key factor.

Use the highest-quality input device you have access to, and find one that works well with your guitar. Even if you're going to be reamping the part later, it's still important to start by recording

REAMPING THE OLD-FASHIONED WAY

Although you can reamp with a modeling plug-in in your sequencer, or a modeling effect in your personal digital studio, the traditional process of reamping entails taking a guitar track that's already been recorded and patching its output to a guitar amp (see Fig. A). You then play back the guitar track with a mic (or mics) on the amp and rerecord it onto a fresh track.

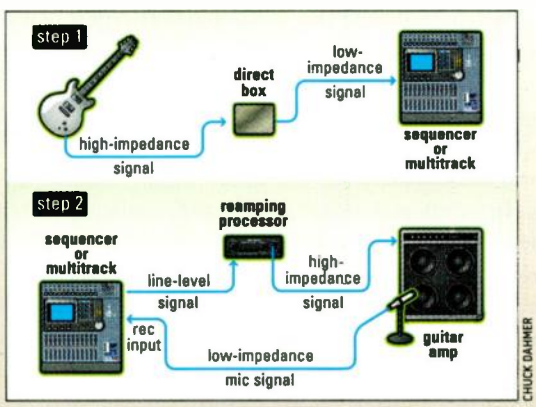

FIG. A: Traditional reamping involves recording an unprocessed track and then patching it out of the recorder into a guitar amp, where the signal is miked and brought back into the multitrack.

It's not advisable to simply take a line out of your sound card or multitrack and run it into a guitar amp for reamping. The line-level signal will be mismatched with the high-impedance input on your guitar amp. You'll need to run it through a box such as the Radial X-Amp, or the Little Labs Red Eye, which converts your line-level signal back to a high-impedance guitar signal suitable for plugging into an amp. You could also plug the converted signal into an outboard modeling processor.

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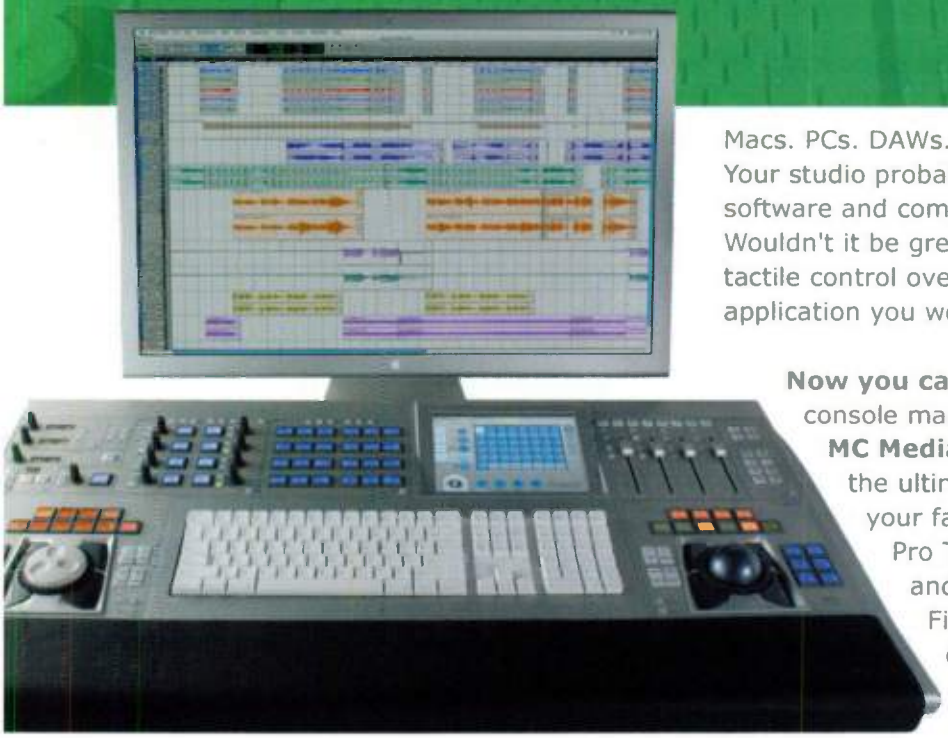

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the best possible sound. A number of engineers I've spoken with like to use a tube preamp for DI guitar parts, because they feel it helps inject a more realistic tone, even to a part that will later be reamped. (In the process of reamping, you take a clean DI guitar track and patch it out to an amp, first going through a dedicated reamping processor to convert the output to an impedance appropriate for a guitar input, and then rerecord it with a mic through the amp. See the sidebar "Reamping the Old-Fashioned Way.")

FIG. 2: The M-Audio Black Box is one of several guitar-recording devices on the market that offer amp, cabinet, and effects modeling, and a USB audio interface for recording direct.

Decisions, Decisions

When deciding how to record a DI guitar part that will be processed with a modeler, you have one fundamental choice: will you record the modeled tone directly onto a track of your sequencer or multitrack, or will you record it unprocessed and insert the processor's tone during mixdown? There are advantages to each method, and your gear may dictate your decision.

Most outboard modeling processors are better suited for use as direct boxes that output a processed sound to a recording device, rather than as insert effects that are used on unprocessed, previously recorded guitar tracks.

Generally, modeling processors have only high-impedance inputs, which aren't designed for the line-level, low-impedance outputs that you would need to connect them to if you were using them as inserts. So unless your processor has line-level inputs (such as the Line 6 Pod XT Pro and the Tech 21 SansAmp PSA 1.1), its architecture dictates that you use it as a DI and record its processed sound to disk.

Many outboard modeling units (for example, the Line 6 Pod XT, the DigiTech GNX4, and the M-Audio Black Box; see Fig. 2) have USB outputs that allow them to function as audio interfaces in their own right. Alternatively, you can plug their analog outputs into the line inputs of an audio interface.

Perhaps the biggest advantage of recording through an outboard modeling processor is simplicity. Just as if you were going through an amp (or an amp with effects pedals), what you hear is what you get. Find a tone you like, record the track with it, and you're done. Because your signal is being processed before it hits the computer, latency is not as big an issue as it is when using a plug-in modeler.

Session Sense

If you're a guitarist who records in studios besides your own, going direct through an outboard modeler is an excellent and lightweight option for toting your sound around. Unless you really trust the engineer or producer on a particular session, you probably don't want to record an unprocessed DI part and let them add the modeled amp and cabinet tones later, when you're not around. With an outboard processor, you can have your best sounds saved as presets and ready to go for any type of session, and you'll have more control over your final guitar tone.

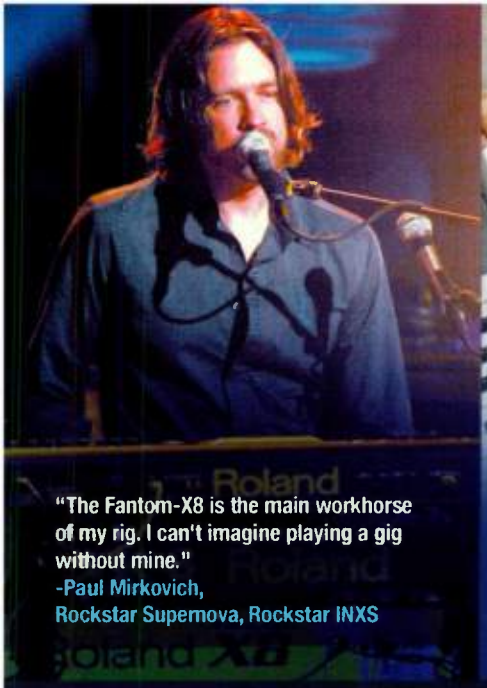
One component of a guitar's sound that probably

FOUR-STRING DI

Electric bass is recorded direct even more frequently than guitar (see "Sweet & Low" in the June 2004 issue of EM, available at www.emusician.com). Bassists recording direct usually go through quality analog preamps and compressors. However, more and more products are becoming available that feature bass-amp and bass-cabinet modeling. If you've got a guitar modeler, it may also have presets designed for bass recording and processing.

FIG. B: IK Multimedia's recently released Ampeg SVX is a modeling plug-in that emulates a variety of Ampeg bass amps.

For example, Guitar Rig 2 and AmpliTube 2 offer bass presets in their standard factory collections. IK Multimedia recently released Ampeg SVX, a dedicated bass-modeling plug-in that provides a large selection of modeled Ampeg bass amps and cabinets, along with effects processing and more (see **Fig. B**). Some examples of outboard bass processors include Tech 21's SansAmp Bass Driver DI, which lets you dial in a variety of bass-amp tones (using analog circuitry); Line 6's Bass Pod XT; the Korg AX10B; and the DigiTech BP200.



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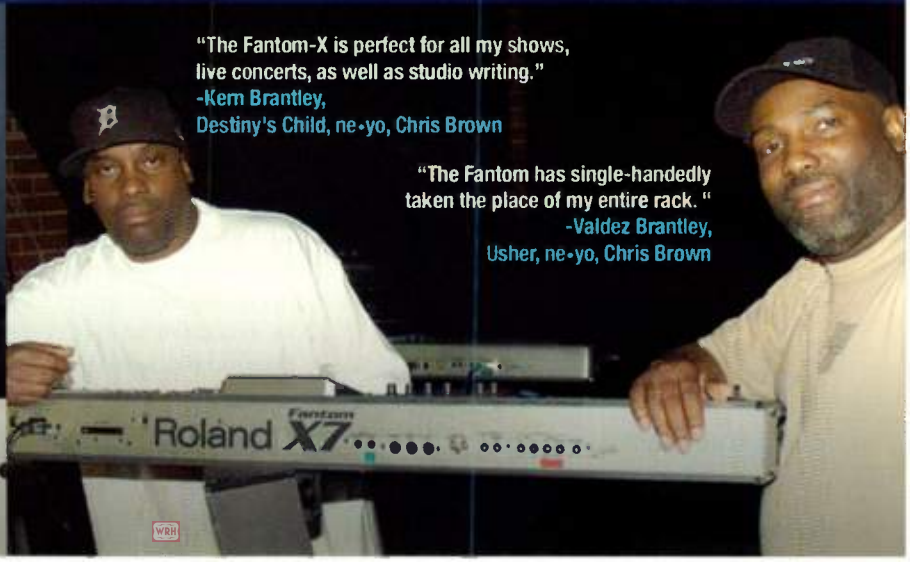

shouldn't get recorded through an outboard processor is reverb. Although your modeling processor likely has many reverb options, it's best to record tracks dry and add the reverb later—even if you won't be present when it's applied. At the time you record your part, it's not always easy to tell how much reverb will sound appropriate in the mix. So it's best not to commit when tracking. This is true not only for DI tracks, but also for miked, guitar-cabinet recordings. Unless you've got a really unusual reverb that's essential to the sound you're going for, I'd recommend turning it off when tracking.

To some extent, the same is true of delay, especially for sounds that use relatively long delay times. It's hard to judge how to set the mix control (wet-to-dry ratio) of such a delay until it's heard in the context of the final mix. What's more, delay plug-ins have the ability to sync to the sequencer's tempo. A tempo-synced delay will often sit better in the track than a manually set (or tap tempo) delay from your outboard processor. (Some modeling processors, like the Black Box and Roger Linn's AdrenaLinn II, can be synced to the host tempo using MIDI Clock.)

DOUBLE DOSE

Here's a pseudo doubling trick you can use when recording with a direct box. Patch your guitar signal into the DI's instrument input and through its XLR output, and connect the DI's ¼-inch pass-through output to an outboard modeling processor that's connected to another line input on your recorder. Record both signals to separate tracks, panned left and right, and reamp the unprocessed track (from the XLR out) with a modeling plug-in. You can get some really fat sounds without having to double the track (see **Web Clip A**).

If you have the time, track count, and inclination, recording a doubled rhythm guitar part (especially in rock and pop songs) is a great method for getting thick sounds. Many contemporary recordings feature doubled rhythm guitars. After you've recorded the first track, the trick is to make sure you hear it really well in your monitor mix when recording the second, so that your doubled performance is as close as possible to the original.

Plug and Play

Just as an outboard modeling unit is best suited to processing your guitar sound on the way into the recording device, an amp-modeling plug-in in a sequencer lends itself to being used as an insert effect. Why? Because most digital audio sequencers make it easy for you to insert your modeling plug-in into one of their virtual mixer channels, and monitor the processed sound while recording the guitar to disk unaffected. When you play back the track, it also goes through the plug-in.

FIG. 3: The Hughes & Kettner Red Box Classic lets you take a direct out or speaker out from your amp and process it with a cabinet emulation before sending it into your DI input.

Here's how you do it. Connect your guitar to your sequencer using a DI box, audio interface, or mic-preamp instrument input as described earlier. Create a track to record on, and instantiate the modeling plug-in in one of the track's insert slots. If necessary, set your sequencer's input monitoring options to monitor through effects, and you're ready to go.

This approach is doubly advantageous: you get both an amp-modeled sound as you're playing and the flexibility to change sounds later on. Some sequencers, such as Apple Logic Pro 7, Cakewalk Sonar 5, and Steinberg Cubase SX4—and many personal digital studios—come with their own amp-modeling effects. Adding tone through a plug-in after a track has already been recorded is similar in many respects to the process of reamping.

Tardy Signals

On a sequencer, the most common obstacle when recording a guitar track while monitoring through a plug-in is latency. It can be an issue because your guitar sound has to pass through the computer's processing before you hear it. This introduces a slight delay, depending on how you've set your record buffer. If it's set too high, the delay may be long enough to throw your timing off (you play a note and hear it back a split second later). If it's too low, and you're working on a project with lots of tracks and plug-ins that are already taxing your processor, you might experience CPU overload during recording or playback.

Another downside of monitoring through your plug-in is that if you want to hear a purely processed guitar sound, you can't take advantage of any low-latency monitoring features that your interface or sound card might offer. That's because such a setup typically lets you hear your input *before* it goes through your audio system's processing, including any plug-ins.

According to Line 6, its recently released GearBox plug-ins are designed with a unique architecture that


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provides the advantages of recording unprocessed while monitoring through a plug-in, but also gives you low-latency monitoring (when used in conjunction with the included TonePort DI USB interface or certain other Line 6 hardware units). GearBox plug-ins are available in AU (Mac) and VST (Win) formats.

On the Side

Most sequencers will allow you to commit the sound of your modeling plug-in to disk as you record it, but it typically involves some extra signal routing within the program. Even if you decide to do this, I'd recommend sending your signal to a second track unprocessed to keep your reamping options open.

One advantage of printing your track processed is that you won't have to leave the plug-in active after you've recorded the track. Considering that modeling plug-ins are often CPU hogs, that could save you a nice chunk of processing power.

There's another solution to the CPU usage problem: many sequencers offer a function that lets you "freeze" a track, temporarily rendering it to disk after it's been processed by any inserted plug-ins. Doing so eliminates the CPU hit that your system takes from the track's plug-ins, but allows you to still hear it processed. If you decide later that you want a different sound, just unfreeze the track, and you can change it at will. If your sequencer doesn't have a freeze function, you can always bounce a track to disk and disable the original track.

Alternate Means

An increasing number of products have been introduced recently that allow guitarists to plug their instrument into a computer using USB. For DI parts that will be reamped with plug-ins or actual amps, a direct USB connection is an appealing option. As mentioned, many outboard modeling processors have USB outputs.

These days you can even get guitars with USB capabilities. Certain models of the iGuitar line from Brian Moore Guitars feature class-compliant USB audio outputs, which means they work without any drivers. Just plug into your computer and play. Another, less expensive alternative is a ¼-inch-to-USB adapter cable, such as IK Multimedia's Stealth Plug, which should be available by the time you read this.

Amp Direct

If you have a modeling amp such as the Vox ValveTronix, Line 6 Vetta, or even the inexpensive Roland Micro Cube (which features Roland's COSM modeling technology), you can take a direct line out to your input device's line in (the Vetta also has a digital output). On many such amps, plugging into the direct out or headphone out will mute the speaker, making the amp function essentially as an outboard processor. Some tube amps, such as those made by Mesa/Boogie, also have direct outputs that are designed to be used when recording. Keep in mind that when using an amp's direct output, you'll be missing the sound

of the speaker, so you might want to add a speaker simulation for additional realism.

You could do that with a modeling plug-in, or you could use one of a number of outboard products designed to provide you with a speaker-simulated DI sound. For instance, the Hughes & Kettner Red Box Classic (see Fig. 3) lets you plug your amp's line out or speaker out into it, and then sends a balanced signal, featuring a simulation of a 2 × 12 or 4 × 12 cabinet, to your sequencer or multitrack.

When plugging your speaker output into a simulator, remember that you must always keep a speaker load on your amp, or you might damage its output transformer. The Red Box Classic has a Speaker Thru output, which you connect to your speaker input jack after you've plugged the amp's speaker out to the Red Box's speaker input. Doing so presents your speaker with its accustomed load, avoiding any possible damage. The disadvantage of this arrangement is that you'll still hear the speaker, which will negate one of the benefits of direct recording—silent operation to avoid leakage.

If you're willing to spend considerably more, you can get processors that let you record direct from your amp with the speakers muted. For instance, the Palmer PGA-04 and the Sequis Motherload provide speaker emulation and send a "dummy load" to your speaker

FIG. 4: Fretted Synth Audio's FreeAmp 2 is a fully featured freeware amp-, cabinet-, and effects-modeling VST plug-in for Windows.

input, thus avoiding any potential damage. Another advantage to running an amp with the speakers muted is that you can crank it up really loud, which, with a tube amp, will likely net you a sweeter tone.

A hybrid approach that will give you a real cabinet sound with the flexibility of reamping starts with recording your part clean through a miked amp. Then you can reamp it through a modeling plug-in, adding whatever amp and effects sounds you want.

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Where Are the Models?

Even if you don't have a modeling device or plug-in, you can still record some believable DI guitar parts, although it takes more doing and will yield only clean sounds. One trick is to add a tad of room reverb to the track, to give it a bit of space (see **Web Clip 2**).

Adding some compression can also help your track sound more authentic. Try experimenting with EQ settings too. Applying a highpass filter to cut out some unneeded lows will often improve your track. It will also help the track sit better in the mix. And, as mentioned, a tube preamp can help.

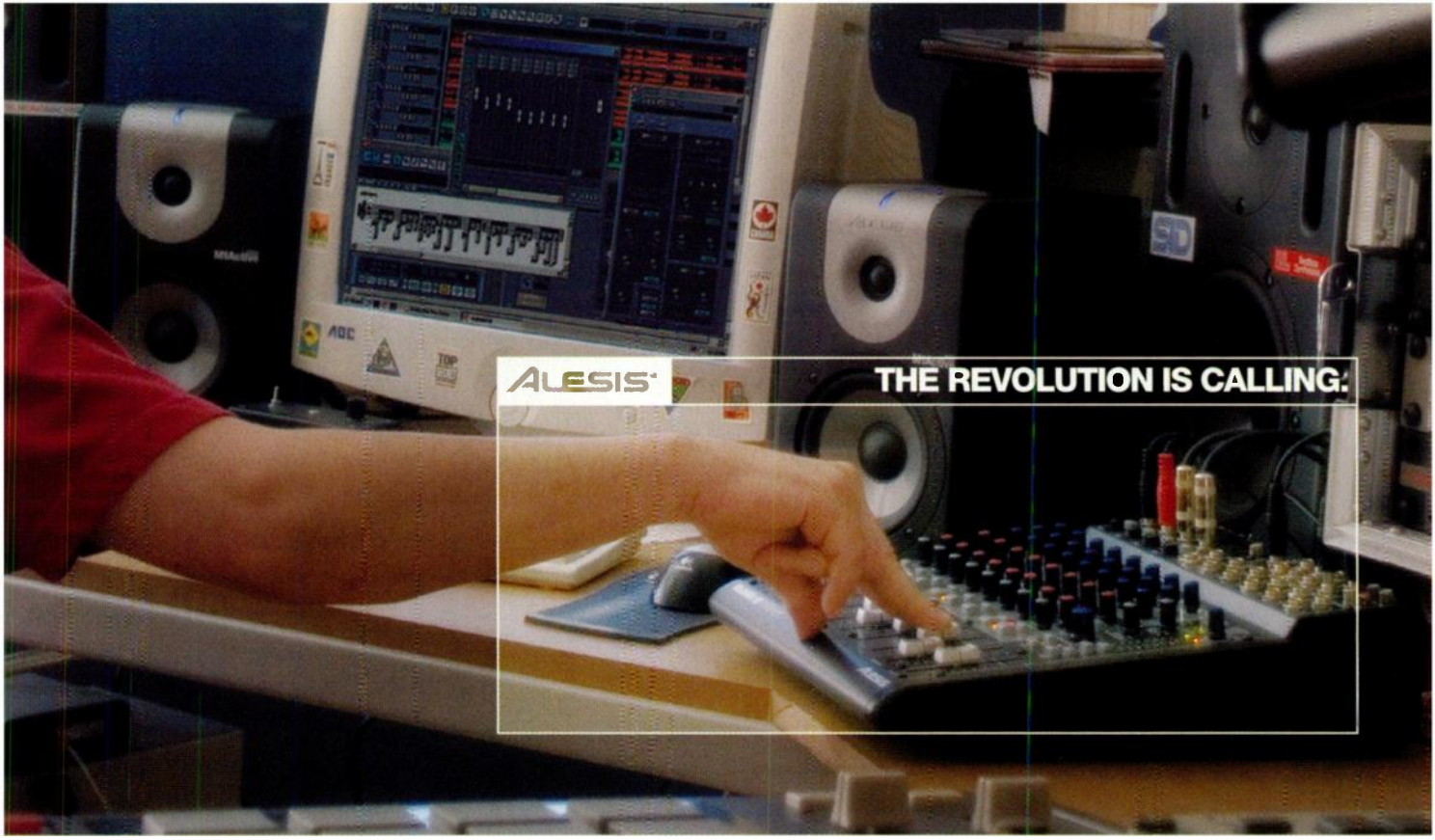
But if you record on a computer, there is no reason to be without amp-modeling software. Even if you don't want to spring for a commercial modeling plug-in, you can find pretty-good-quality freeware modelers for both Mac and Windows. For instance, MDA Combo, which runs on Mac or Windows, gives you speaker simulation, overdrive, and several other sound-shaping parameters. Windows users might also want to try Fretted Synth Audio's FreeAmp 2 (see **Fig. 4**). This VST plug-in offers amp and cabinet simulations, overdrive, chorus, reverb, delay, tremolo, and more.

Go Directly There

Recording direct is a useful option that should be considered whenever you're tracking guitar. It may not always provide you with the ultimate sound, but it gives you versatility, variety, maximum isolation, and convenience. In many situations, it's simply the best way to go. EM

EM WEB CLIPS

Mike Levine is an EM senior editor. Thanks to Mark McCrite, Peter Janis, John Bednar, Orren Merton, and Michael Cooper.



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SIX-STRING STRATEGIES

By Michael Cooper

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The acoustic guitar is arguably one of the most challenging instruments to record, due to the large number of variables that affect its sound. In a sense, the instrument carries its own room—the body—around with it. The size, shape, and internal bracing of the body, as well as the type of wood used in its construction, have a profound effect on its acoustic properties and thus the guitar's overall sound.

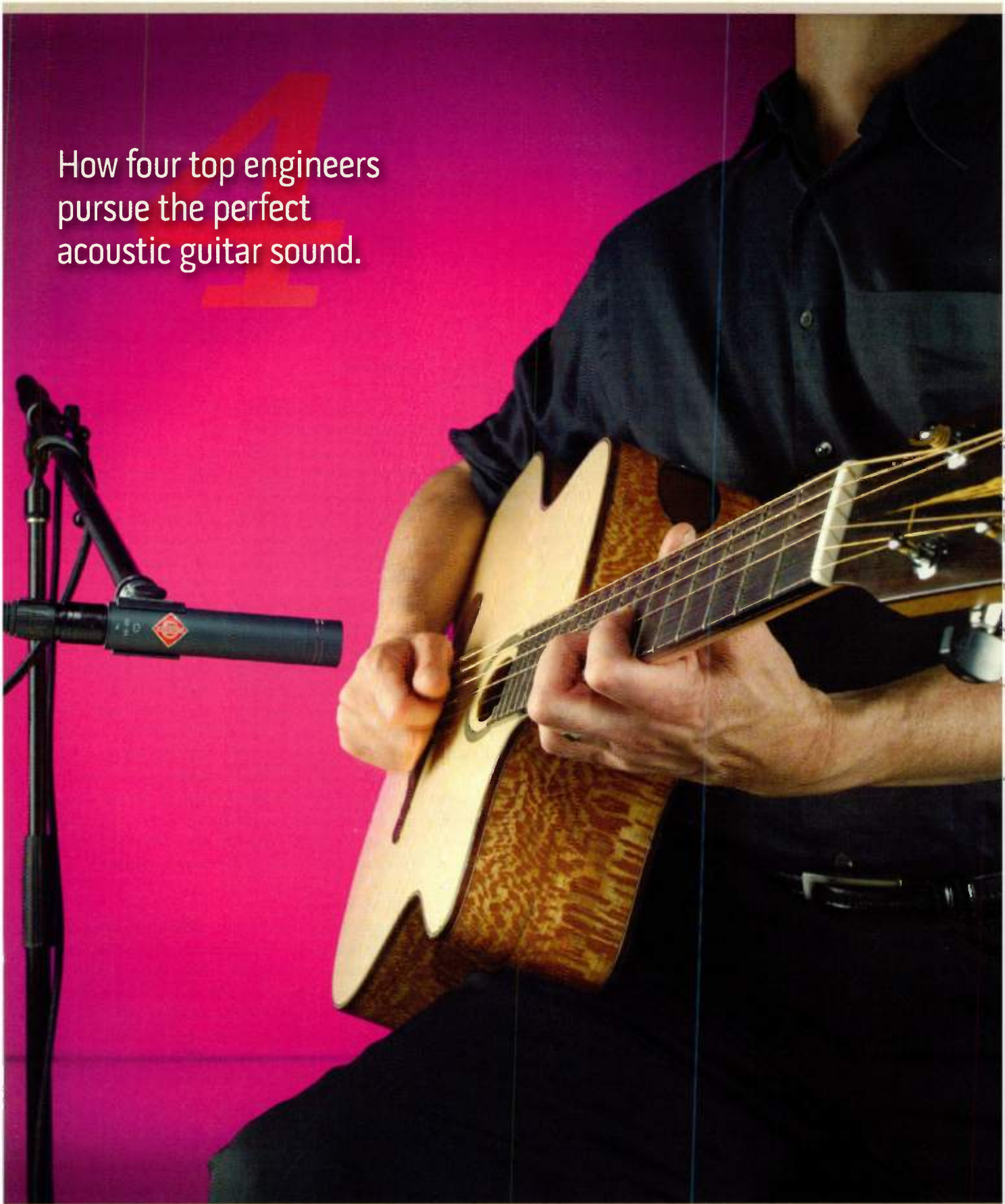
Other factors greatly influencing an acoustic guitar's timbre include the neck construction, choice of strings, and type of pick used. But above all, it's a player's unique techniques that shape the sound of the instrument: have two musicians play the same guitar, and it will likely sound different, sometimes drastically so.

All of these factors come into play before the recording studio's acoustics and the engineer's mic selection and placement are even considered. If that were all the variables involved, recording acoustic guitar would be challenging enough. But you also have to consider the signal path, which includes selecting the best mic preamp for the job, and deciding whether or not to equalize and compress the signal going to disk or tape.

For this article, I interviewed four of the music industry's most accomplished recording engineers—Brian Ahern, Richard Dodd, Jay Newland, and Dave Way—to glean their expert advice on recording acoustic guitar (see the sidebar "Acoustic Guitar Gurus"). We talked about mic choice and placement, polar pattern selection, and signal processing (including the use of DI boxes). I asked the engineers what their favorite pieces of gear were for recording the instrument, and what equipment they would recommend for recordists on a limited budget. We also occasionally touched on the subject of mixdown considerations to keep in mind while recording.

The responses to my questions were surprisingly varied and not always what I expected. No matter how deep your expertise in recording acoustic guitar, you're bound to find some fresh tips and perspectives in the discussion that follows.

How four top engineers pursue the perfect acoustic guitar sound.



Too Much of a Good Thing

Before you even consider mic choice and placement for recording acoustic guitar, you must decide whether to track using one or two—or more—microphones. The key here is to think ahead to what might be needed at mixdown. The decision of how many mics to use to record the instrument often comes down to balancing the need for clarity against how big the guitar track needs to sound. In most cases, clarity wins out and a mono recording is the best way to get there.

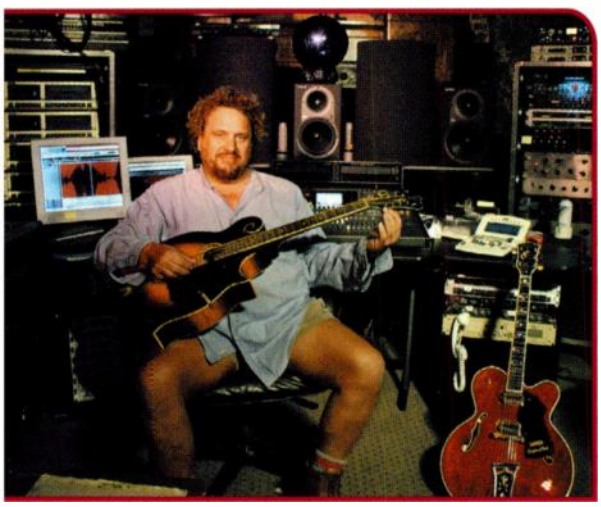
"I prefer to use one mic," says Richard Dodd. "Unless the acoustic guitar is the only instrument in the song, it's very difficult and a waste of time in the mix to have two mics. If you've got one that sounds great, you don't need another one."

Jay Newland is of the same mind. He records acoustic guitar "mostly in mono, occasionally stereo if it's the only thing there."

Dave Way also concurs. "More often than not, it's going to be one [mono] microphone," he remarks. "The times when I would use a stereo mic would be if the guitar is going to be either the only instrument in the track or a really big part of it to where one lone mic and maybe a vocal is not going to be as interesting as something that's slightly bigger and more open. But most of the time it's going to be one mic.

"I find, as a mixer, nothing gets more dense and complicated than everything miked in stereo," Way continues. "Often in a mix, I'm either picking one channel out of the left and right or, more often than not, summing a stereo track to mono and panning it in one position because, in a dense mix, lots of things in stereo just sound like *moosh*. If I know going into a recording that it's gonna be in a dense track, one microphone is a good friend."

It may sound counterintuitive, but the pinpoint imaging provided by mono acoustic guitar tracks can often make a mix sound wider—especially if the same part is performed twice and recorded to separate tracks. Dodd, Way, and Newland are all fans of double-tracking the same acoustic guitar part and hard-panning the resulting tracks opposite each other in the mix (when appropriate for the production). The result usually sounds much wider and bigger than what any stereo-miking technique could produce. But that doesn't mean that piling on even more guitar parts will add more size.



Brian Ahern

RICK CLARK

CAROLYN DODD

Richard Dodd

"Often in a pop or pop-rock record, you're gonna have a double-tracked acoustic guitar, sometimes maybe even triple tracked," Way notes. "I've mixed tracks where there are six acoustic guitar parts all playing basically the same thing, but that gets to be overkill. You end up sacrificing clarity for the sake of trying to make something sound huge. Things can only sound huge in context to something that doesn't sound huge."

Brian Ahern offers an interesting twist on the double-tracking technique. He remembers one production where he added a digital delay to each of the two hard-panned acoustic guitar tracks. He panned each delay toward the opposite side from its dry counterpart "so that they kind of danced with each other back and forth. I'd pan guitar A to the left and pan the effect from that guitar halfway to the right. That's the only time I've done that," Ahern recalls. He goes on to note that he made sure the effect was "dull-sounding so it would not distract, it would just create a mood. I always roll the top off my echoes. I find them distracting otherwise."

Put It There

When recording acoustic guitar with one mic, Way generalizes his approach to mic placement by saying, "More often than not, you're gonna be down by the sound hole. I usually like to place the mic about 10 inches, maybe 12 inches from the guitar, if it's the only instrument I'm recording at the time. You might not be able to get as far away if you're in a really live room." He also notes that a guitar played with fingers instead of a pick usually calls for closer miking to compensate for the instrument's resulting quieter sound.

"If I'm looking for something that's brighter," Way continues, "the mic will probably be pointed more towards the bridge than if you're trying to get something warm." If the guitarist is playing mostly the

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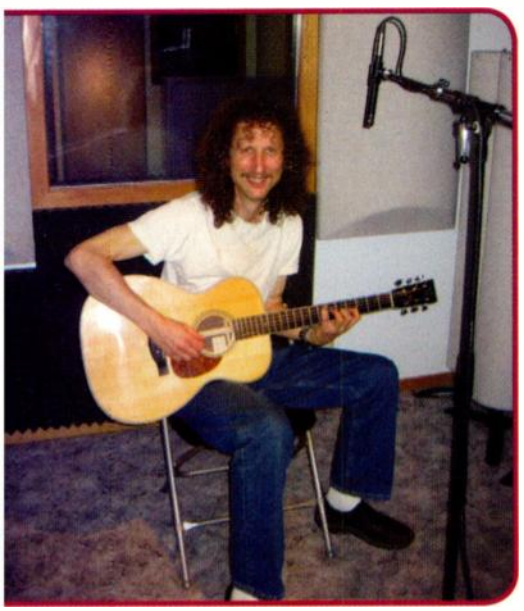
higher-pitched strings, Way says he might point the mic more at those strings to avoid having to "carve out the low frequencies with an EQ."

Way stresses that these are all generalizations. "I'm always trying to put it in context before I make any of these decisions. There's never any rule, because the music always dictates what you're going for. You want to know what you're trying to achieve before you just blindly put it up." That said, he notes "there are times when you have no idea what they're going to play and you don't have the ability to find out ahead of time," such as when recording a live event. He also emphasizes that initial mic placement should always serve as just a starting point. "It's pretty rare that you're just going to put a mic in front and not want to go out and move it even just half an inch," he says.

Ahern generally likes to mic acoustic guitar "as close as I can to where the strings are plucked." To avoid a boomy sound with such close mic placement, he will use an omnidirectional mic. In fact, he typically prefers omni mics because they don't have all the design "machinations" that directional mics must have in order to create their null points, such as ports that change the incoming signal's phase. "A bad omni mic sounds better than an expensive cardioid," Ahern declares flatly.

Dodd takes a markedly different approach to miking acoustic guitar from Ahern and Way. He goes so far as to say that miking the wide part of the guitar's body is "mostly employed when the instrument sucks." He maintains that a good starting point for mic placement is roughly four to six inches away from the instrument, angling the mic down from above the level defined by the top of the sound hole. He positions it slightly past the end of the fretboard and aims roughly at the low E string. But he ironically points out that "wherever you put a microphone, as soon as you turn your back and walk away, that's the last you hear of that sound, because the player is going to settle himself in, lean back, and it's all going to be different anyway."



JANET HUERTA

FIG. 1: A small-diaphragm, cardioid condenser mic is placed a foot in front of the guitar and about 18 to 24 inches above the sixth fret.

Magic Carpet Ride

It bears mentioning that yet another approach besides those discussed earlier may be necessary when recording overdubbed acoustic guitar in a heavily damped project studio, such as one that is carpeted or makes extensive use of acoustical foam. In such a situation, I have achieved great results by placing a small-diaphragm, cardioid condenser mic—one that responds very quickly to transients, such as a B&K 4011—roughly 18 to 24 inches above the guitar's sixth fret, and about a foot in front of the guitar on the horizontal plane, angled so that it points at the strings (see Fig. 1). This mic placement would likely be too distant in a room with very live acoustics, when other instruments are playing with the guitar in the same room, or with a mic that has a soft high-frequency response.

JANET HUERTA

FIG. 2: Recording with several mics simultaneously gives you the option of picking the ones that work best for the song when it comes time to mix.

For neophytes who are still learning how to record, Dodd advises putting up several mics simultaneously in different positions relative to the guitar while recording (see Fig. 2). They can then choose which mic(s) and position(s) sound the best during subsequent playback.

Newland has similar advice for acoustic guitarists who record themselves. He recommends they move the mic up and down the neck of the guitar while wearing headphones until they find the position that sounds the best. Newland generally avoids placing the mic near an acoustic guitar's sound hole.

Pro Choice

Because every microphone has a different sonic signature—which, especially with a directional polar pattern chosen, varies according to the distance the mic is placed away from the source—mic choice is usually decided on together with mic placement. All four engineers were careful to state that they always make spontaneous mic choices based on the live sound being created by the guitar and player. They also said there is no inherent difference in their approach to miking an acoustic guitar fitted with a capo as opposed to one that is not.

Dodd, whose credits cover a wide range of both pop and country productions, takes essentially the same approach to miking acoustic guitar no matter what the musical style, noting that "the player makes more of an

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adjustment than a microphone ever could." That said, all four engineers revealed a general preference for using small-diaphragm mics to record acoustic guitar.

"Often my favorite mic to start with would be like a KM 54 or an M 582, if I'm going for a nice, well-rounded kind of featured acoustic guitar," says Way. (Both mics are vintage, small-diaphragm Neumann tube condensers.) "The other microphone that I love a lot of times is the AKG D 19," a '60s-era dynamic mic used by the Beatles on drums, piano, and acoustic guitar. "It's great when you want something that has a little ratty attitude. It's sonically interesting rather than beautiful. You used to be able to get them for about $200 or $300, and now I've seen them as high as $600 on eBay."

Newland also generally prefers small-diaphragm condensers for recording acoustic guitar, citing the Neumann KM 84 and AKG C 460 B. "That said, there are times when it seems like a Neumann U 47 [a large-diaphragm condenser mic] is the right thing to put up," he qualifies. "A lot depends on the instrument."

The artist's personal preference can also come into play. On Norah Jones's second album, *Feels Like Home* (Blue Note, 2004), Newland used a Royer R-121 ribbon mic placed 6 inches away from a fingerplucked acoustic guitar. "We really wanted to get the *wood* in the guitar," he explains. Because the R-121 exhibits a significant amount of bass-boosting proximity effect and an attenuated high-frequency response, Newland recalls he had to roll off a little bottom-end EQ and add some highs. However, he mostly compensated by moving the mic farther down the neck, away from the sound hole, when he wanted a sound with less bass. He also points out that aiming the back side of an R-121 at the instrument yields a less bass-heavy sound. In any case, Newland preferred to roll off bottom-end EQ on the track as needed, rather than moving the mic farther away and picking up too much of the room sound.

Ahern likes to record acoustic guitar using a B&K 4090, in part because its fixed omni pattern and small profile facilitate very close miking. The larger B&Ks, such as the 4011, sound too pristine for his taste. "The 4090s have a little color to them, what I call 'armpits,'" Ahern says. He recollects that he used an AKG 224E on all of the early Emmylou Harris records he engineered, with great results. He also cites the Neumann M 50 microphone as "arguably the best ever made. You can record an acoustic guitar 15 feet away [with an M 50], and it sounds like it's in your face." For recordists on a budget, Ahern recommends the Shure KSM27 condenser mic on acoustic guitar.

Dodd's favorite small-diaphragm mics include the Neumann KM 140 and KM 56 (the latter is a vintage, variable-pattern tube mic), as well as the workhorse Shure SM57 dynamic mic. He'll use a mic with a wider frequency response—one that will capture more of the instrument's sound—if he doesn't know ahead of time how the guitar will need to fit in the mix. "It's easier to get rid of it than it is to make it up," he explains. But if there will be a lot of electric guitars and drums for the acoustic guitar to compete with in the mix, Dodd notes, there is "no reason why an SM57 or 58 won't work. The mic's natural tendency to be 'middley' is going to save you some trouble later."

Jay Newland

Newland has also used the SM57 to record acoustic guitar. "In a rock context, where you have some pretty slammin' stuff going, put a 57 on there and compress it a little bit, and it'll give you that nice midrange rip," he says. "Sometimes the unusual thing is the right thing. Most of the time, I do sort of the standard stuff. You know, the small condenser or whatever. But every once in a while, you just need something weird."

Who's Sorry Now?

Dodd makes no apologies about using the inexpensive SM57 to record acoustic guitar. "A 57 only sounds like an inexpensive microphone when it's paired with an inexpensive preamp," he declares. "You pair it with an extremely sensitive, musical preamp, like a Telefunken V76 or Neve 1073, and the balance of the two is a wonderful thing to be heard."

Dodd's philosophy on mic choice should hearten those on a limited budget. He notes that using a very expensive mic can sometimes "adversely affect the performance" by putting added pressure on the performer. If, on the other hand, you "up the ante of the performance by making the person feel comfortable, that will improve the sound more than any finessing of microphone choice or position," he says. "It's got to be in the ballpark. But you can have the best microphone and the best mic pre, and if the player isn't comfortable and can't perform, you've got nothing."

Dodd asserts that fast setup is also far more important than choosing the absolute best mic. "It's better to have the wrong mic ready than the right mic after the

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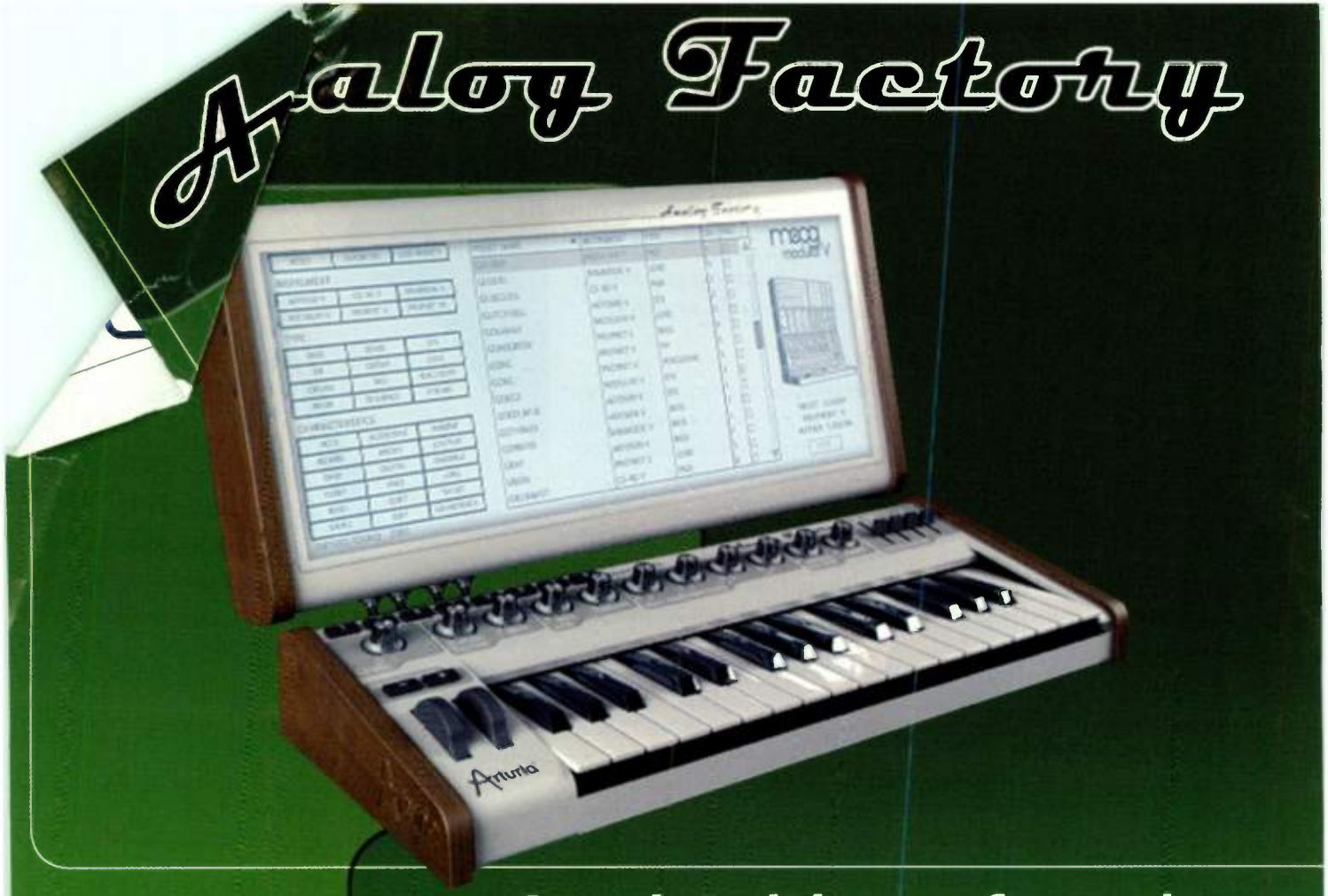

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performance has peaked," he insists. "'Keep playing, I'm just gonna try this . . . keep playing, I'm just gonna try this . . . keep playing, I'm just gonna try this . . .' Okay, now they're all played out. Now you've got a great mic, and nothing to record." He adds that the instrument, room, performer, musical arrangement, and choice of strings and pick "have more effect on the acoustic guitar sound than the choice of any given set of good microphones."

It Takes Two to Tango

For all the benefits of recording acoustic guitar in mono, there are times when two or more mics are called for. This is often the case when the guitar is the only instrument in the mix and needs to sound very big.

In such a situation, Dodd sometimes likes to use one mic placed close to the instrument and a stereo pair placed farther back to capture room ambience. (Of course, distant miking dictates that you record in a room with desirable acoustics that will enhance the recording.)

Way usually takes a different approach when miking an acoustic guitar in stereo. "Often the first thing I go for is a stereo microphone like an AKG C 24 or a Shure SM2, so that my phase problems are going to be minimal," he says. Way suggests the Sony ECM-999PR, a discontinued stereo electret condenser, as an inexpensive alternative to the higher-priced mics. "I really like the high end" of the ECM-999PR, he explains. "It's not full bodied at all, but it's very sparkly sounding. When I'm looking for a guitar sound that, in the mix, I would be rolling off bottom on anyway, and I want it to be in stereo, I'll use that mic. It does a really nice job." He'll sometimes add another microphone as an ambient mic to complement the Sony's sound.

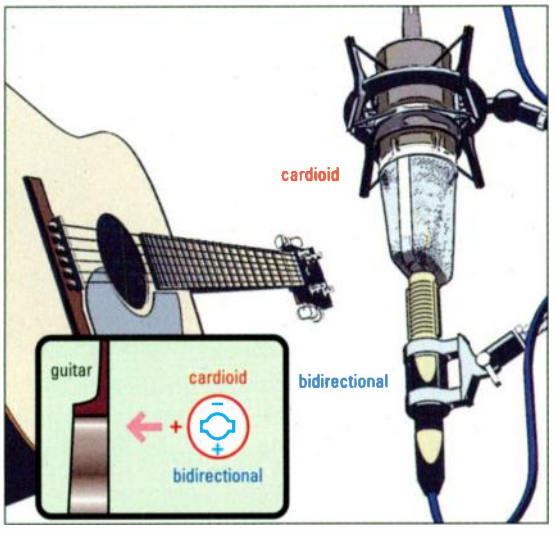

FIG. 3: The mid-side (or M-S) technique uses two microphones to create a stereo image. A cardioid mic is pointed directly at the source, and a coincident, bidirectional mic (positioned so that the capsules of both mics are almost touching) is aimed 90 degrees off-axis to the cardioid mic. Polarities in the above illustration denote the receptive side(s) of each microphone. The two mic signals must be processed using specialized equipment in order to create and monitor the effect, which offers good mono compatibility.

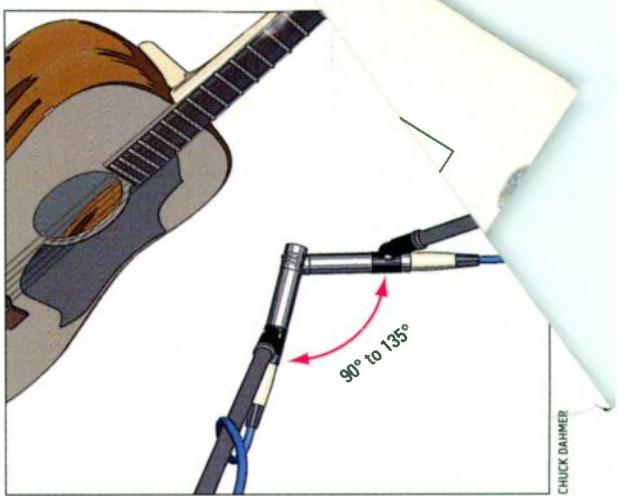

FIG. 4: XY mic placement uses two coincidentally placed mics (typically with a cardioid pattern) positioned to create an angle of 90 to 135 degrees between them. By positioning the capsule of one mic directly over that of the other, sound arriving at the mics in the horizontal plane will be picked up at virtually the same time, reducing phase cancellations. The XY technique is highly mono compatible.

Ahern has a predilection for using ribbon mics to track acoustic guitar in stereo, in part because of the coloration they add to the recording. "I don't like flat recordings," he says without hesitation. "Sometimes I'll record Emmylou Harris's guitar with two Varacoustics [vintage RCA ribbon mics], one up by the 12th fret and the other down by the fingers of the right hand. That's a pretty interesting sound." In this configuration, he'll typically place the ribbon mics about six inches away from the guitar. He also uses beyerdynamic M 160 and M 130 mics in a mid-side configuration, placing the mics near where the fingers are plucking the guitar (see **Fig. 3** for an explanation of mid-side mic placement). The vintage, '70s-era Reslo ribbon mic is another of Ahern's favorites on acoustic guitar. (For more tips on using ribbon mics on acoustic guitar, see "Ribbon Mic Summit" in the August 2006 issue of EM, available online at www.emusician.com.)

Newland usually leans toward using an XY mic configuration for stereo-miking acoustic guitar (see **Fig. 4**). "If I were going to use a pair of mics, nine times out of ten I would do XY," he affirms. "I like the punch of it. You get a little more centeredness," he says when describing the sound of XY miking compared with that achieved by using a spaced pair of mics.

Newland is not afraid to get radical, however, when a gigantic sound is called for. On *Beyond the Missouri*

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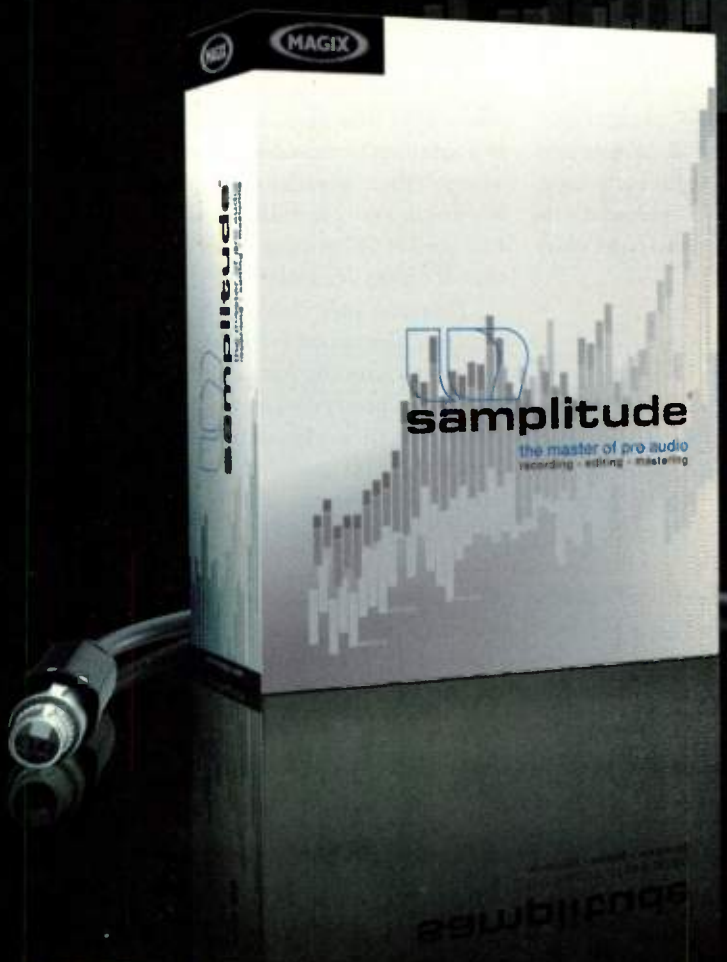

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MAGIX

Sky (Universal, 2003), a DVD collaboration between Pat Metheny (playing acoustic guitar) and Charlie Haden (bass), Newland notes he "went for this monstrous, huge acoustic, stereo sound. Not stereo left and right, but this wider image of the guitar. I think we had eight tracks for the acoustic on a single pass. There would be a large-diaphragm condenser, a small-diaphragm condenser pair in XY, a spaced pair, an internal mic on the guitar, and a custom DI." Newland used all of those mic signals to varying degrees. "If there was a big finger squeak and you wanted to get rid of that, you could kinda cheat it and push the level of the DI for a bar." The large-diaphragm condenser and small-diaphragm condensers (in the XY configuration) were all cardioid and placed as close together as possible, roughly six to eight inches away from the guitar's 15th fret. The spaced pair was placed farther back and generally used more sparingly in the mix than the other mics.

Pickup Lines

Dodd and Way generally regard using a DI box to record acoustic guitar as a last resort to solve some sort of problem with the miked signal(s). Ahern and Newland see limited use for a DI signal when trying to achieve a creative effect. (See the sidebar "The Direct Route" for information on using DI boxes to record acoustic guitar.)

Dodd is adamant that a DI is useful "in a live situation only. It's such a ridiculous thing to put a DI on an acoustic guitar [in the studio]," he says. "You only do it when you have to. There are reasons to use a DI—in a live situation, it's very convenient and very controllable—but sound quality isn't one of them. It sucks. You don't need to use it, either."

Way agrees with Dodd, saying he won't use a DI box "unless it's absolutely necessary. That's just not a sound that I particularly like." When pressed to mention DI boxes he likes, Way countered that the last time he used one on acoustic guitar was "not in this decade."

Ahern will use a DI box on acoustic guitar "only if the artist wants a special effect." On one Emmylou Harris record, he recounts putting a Sunrise pickup on the acoustic guitar, boosting the signal with an old Music Man tube amp, and routing the amp's output to a spinning loudspeaker, in order to achieve a Leslie-speaker effect. When he must use a DI box, Ahern usually reaches for custom-built units. But he also notes that the API 512 preamp's instrument input "cannot be beat. It's often overlooked."

Newland says that for a production that isn't "organic," one option is to pan mic and DI tracks from the same recording pass opposite each other. "That gives you a pretty cool stereo effect," he notes. When asked what DI box he could recommend, his response was that "for the money, the Countryman is one of the best DI boxes you can get." He also lauds the Universal Audio 6176 Channel Strip as having "a great guitar direct input."

Newland notes that the DI track can be a saving grace in some instances where bleed from a vocal outtake forces you to ditch a miked guitar track that was

ACOUSTIC GUITAR GURUS

Brian Ahern's discography spans 35 years and includes engineering, mixing, mastering, and production credits with many of the biggest icons in country music. Artists Ahern has worked with include Anne Murray, Glen Campbell, Emmylou Harris, Guy Clark, Johnny Cash, Linda Ronstadt, Béla Fleck & the Flecktones, George Jones, Tammy Wynette, and Ricky Scaggs. Ahern recently accepted the Canadian Country Music Association Hall of Fame Award.

Richard Dodd's recording and mixing credits know few musical boundaries and include work with artists as diverse as Johnny Cash, George Harrison, Green Day, Joe Cocker, Roy Orbison, Electric Light Orchestra, Boz Scaggs, Sheryl Crow, and John Hiatt. Dodd has also produced Uriah Heep, Little River Band, and Steve Earle, among others. His mastering credits include Tom Petty and the Heartbreakers and Jars of Clay. Dodd has earned Grammy Awards for Best Engineer (for Tom Petty's *Wildflowers;* Warner Bros., 1994) and Mixing (for Delbert McClinton's *Nothing Personal;* New West, 2001).

Jay Newland's lengthy discography includes engineering, mixing, mastering, and production credits for a who's who of jazz music. Newland has worked with Milt Jackson, Pat Metheny, Dave Brubeck, Charlie Haden, Michael Brecker, Jaco Pastorius, Etta James, and Keith Jarrett, to name but a few. His work also includes forays into the pop world, collaborating with such artists as newcomer Missy Higgins and the late Harry Nilsson. Newland has won eight Grammy Awards, including three for Norah Jones's 18-million-selling debut album, *Come Away with Me* (Blue Note, 2002), on which he served as producer, engineer, and mixer.

Dave Way has lent his magic touch to many a Top Ten single and album, in a dizzying variety of musical genres. He has engineered tracks for Michael Jackson, Dixie Chicks, Sheryl Crow, and, perhaps most notably, Macy Gray, on her Grammy-nominated, triple-Platinum album *On How Life Is* (Epic, 1999). Way has also mixed big-selling tracks for Pink, Christina Aguilera, Fiona Apple, Babyface, Savage Garden, Spice Girls, and Taylor Hicks (including Hicks's 2006 single "Do I Make You Proud," which Way also coproduced for BMG).

performed simultaneously. "There are times where the intention was for the voice and the guitar to go down at once, then you end up replacing the voice," he says. "But there was something very cool about the guitar track, rhythmically, that was cut with the band. You still have the DI track. The music has to win out at that point."

All four engineers stated that there was no difference in their approach to using a DI on an acoustic guitar fitted with magnetic versus piezoelectric pickups, with the exception that Ahern noted piezoelectric pickups usually require a healthy dose of EQ cut in certain frequency bands to sound decent. I have observed that the Demeter VTDB-2 Tube Direct box seems to load either type of pickup less than competing models (due to the VTDB-2's sky-high input impedance), resulting in the capture of a significantly wider range of frequencies. Also, musicians who are interested in using a DI with their Taylor guitar (300 Series model or higher) should check out the Taylor Expression System, which consists of three internally placed transducers and associated active electronics. It is by far the most natural-sounding DI system I have heard.

Amped Up

Despite the DI box's occasional creative and problem-solving uses, a good mic and mic preamp will always deliver a more organic, natural, and realistic acoustic guitar sound. If you can't afford both a high-end mic and an expensive preamp, Dodd recommends that you put most of your money on the preamp.

"You can maximize [the quality of] an inexpensive microphone with an expensive mic pre," Dodd notes. "But you can lose so much the other way around."

When recording to Digidesign Pro Tools, Way prefers to use a tube preamp. "When you're going to digital, especially with something as transient as acoustic guitar, to be able to soften up those transients is definitely something you want," he explains. "Otherwise, it can sometimes just hurt your ear, especially when you put a compressor on it."

Ahern says, "The best all-around preamp for guitar is a [Neve] 1084. If you can't afford one of those, get a Peavey VMP-2 tube preamp. It sounds really close to a 1084. Available cheap, and they sound amazing. Nobody paid much attention to it when it first came out, but it was a fine piece of work, and everybody should own

THE DIRECT ROUTE

Most acoustic guitar pickups provide an unbalanced, high-impedance, instrument-level signal unsuited for direct interfacing with line-level gear such as a workstation I/O box, mixer, or A/D converter. A DI (or direct injection) box converts the pickup's unruly audio signal into a balanced, low-impedance signal that professional audio gear can handle with grace. While most direct boxes (or "DIs" for short) provide mic-level signals at their outputs (requiring connection to a mic preamp for additional gain boost), a few models feature onboard gain-boost facilities that bump up the output level enough for direct connection to line inputs.

Recording the output of an acoustic guitar's pickup with a DI box is easy: connect an unbalanced cable from the instrument's output jack to the DI's unbalanced input, and then patch the DI's balanced output into the input for an outboard mic pre, I/O box, console mic input, or A/D converter for routing to your recorder or DAW (see Fig. A).

If you are also using a mic or multiple mics to record the acoustic guitar, it's best to record each signal to a separate track so that you can independently (and perhaps dynamically) adjust the levels of the direct and miked tracks at mixdown. Be sure to delay the DI's track so that it is in phase with any miked tracks of the same acoustic guitar performance. It takes roughly 1 ms for the guitar's sound to reach a mic positioned 1 foot away. Delaying the DI signal by the same amount (1 ms for every foot the mic is distanced from the guitar) puts it in phase with the mic signal(s). Alternatively, DAW users can simply nudge the DI track later in time so that its waveform peaks line up with those for companion mic tracks. Aligning mic and DI signals in this manner reduces phase cancellations and comb filtering that would otherwise thin out bass frequencies and make higher frequencies sound less smooth.

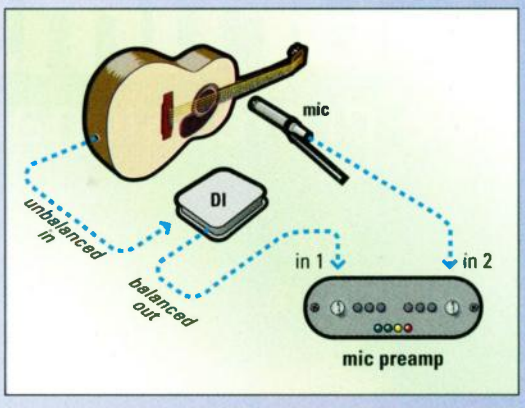

FIG. A: An acoustic guitar being recorded simultaneously with a microphone and DI box, routed to separate channels of a mic preamp.

For further information on using DI boxes, see "Direct Action" in the November 2001 issue of EM, available online at www.emusician.com.

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one." (The Peavey VMP-2 is a discontinued dual-channel model that occasionally sells on eBay for around $650.)

Busy Signal

Way and Dodd have no qualms about applying analog signal processing such as EQ and compression to acoustic guitar while tracking. In fact, they both stress the importance of getting the sound right in the analog domain in order to avoid the use of DAW plug-ins as much as possible. The big caveat is that you have to be sure the processing you want to apply is appropriate for the production before you cast it in stone. Prudence dictates playing it safe if you're just beginning to track and have no idea what will be needed at mixdown.

"The engineer's job is a bit like the Hippocratic oath: first, do no harm," Dodd states. He won't hesitate to apply EQ and compression, however, when he knows it's needed or will add a nice touch. "If I'm sure, I'll commit," he says, adding, "If this is the last overdub, you should be doing it the way it's meant to be on the record. What's the point in not doing that? Leaving too many options [open for mixdown] is a bad thing." But he qualifies his statement by noting that "the EQ usually only comes into play to deliberately change something you already like and seldom to try and salvage something."

Way agrees. "I try to move the mic before I put in EQ," he notes. When EQ is needed, however, he says that "more often than not, you're either rolling off a little bottom, adding a little top, or both."

Like Dodd, Way is "not afraid to put in the compressor. Ultimately, if I'm going for a sound, I'll go for it," he says. A typical way in which Dodd and Way might use compression while tracking acoustic guitar would be to back the mic farther away from the instrument and compress the signal to bring out the room tone (if it's favorable). Way also notes that compression "can affect how the guitar player plays. If he gets that hyperreal sound in his headphones where he can hear every little detail, that can actually inspire a great performance."

Unlike Dodd and Way, Newland usually shies away from compressing acoustic guitar while tracking, preferring to capture "the full signal." He will, however, conservatively apply EQ while recording "if it's obvious it needs a little bit no matter how you look at it." He notes that you can always add a little more EQ during mixdown if needed. He has no compunction about using EQ and compressor plug-ins, citing Waves Renaissance EQ and Renaissance Compressor as favorites.

"The Renaissance Compressor is probably the No. 1 plug-in that I have, period," Newland enthuses. "It's great on guitars. It's a totally flexible, great piece of gear. I've got no problem slapping one of those things on in addition to whatever analog processing I'm using. I'm totally into the concept of a little bit of the old, a little bit of the new."

I have also found Renaissance Compressor to be my go-to compressor, when uncompromised transparency, minimal

coloration, and preservation of depth are required for acoustic guitar tracks.

Ahern will use compression and EQ while tracking acoustic guitar "if the player is looking for a specific trashy, rock 'n' roll sound." But he notes that his favored UREI 1176 "may sit there for a year and never get lit up." As for equalization tips, he revealed that "if you use a felt guitar pick, the plectrum noise becomes suppressed, and then you can crank up the high frequencies without any danger of clicking and banging."

When asked about their favorite compressors and equalizers for tracking acoustic guitar, the four engineers cited mostly vintage, high-end models. However, when pressed for suggestions on an inexpensive compressor to use, Newland said of the FMR Audio RNC Compressor, "That thing, for $200 or so, is one of the most kick-ass pieces of equipment available."

Good Is Good

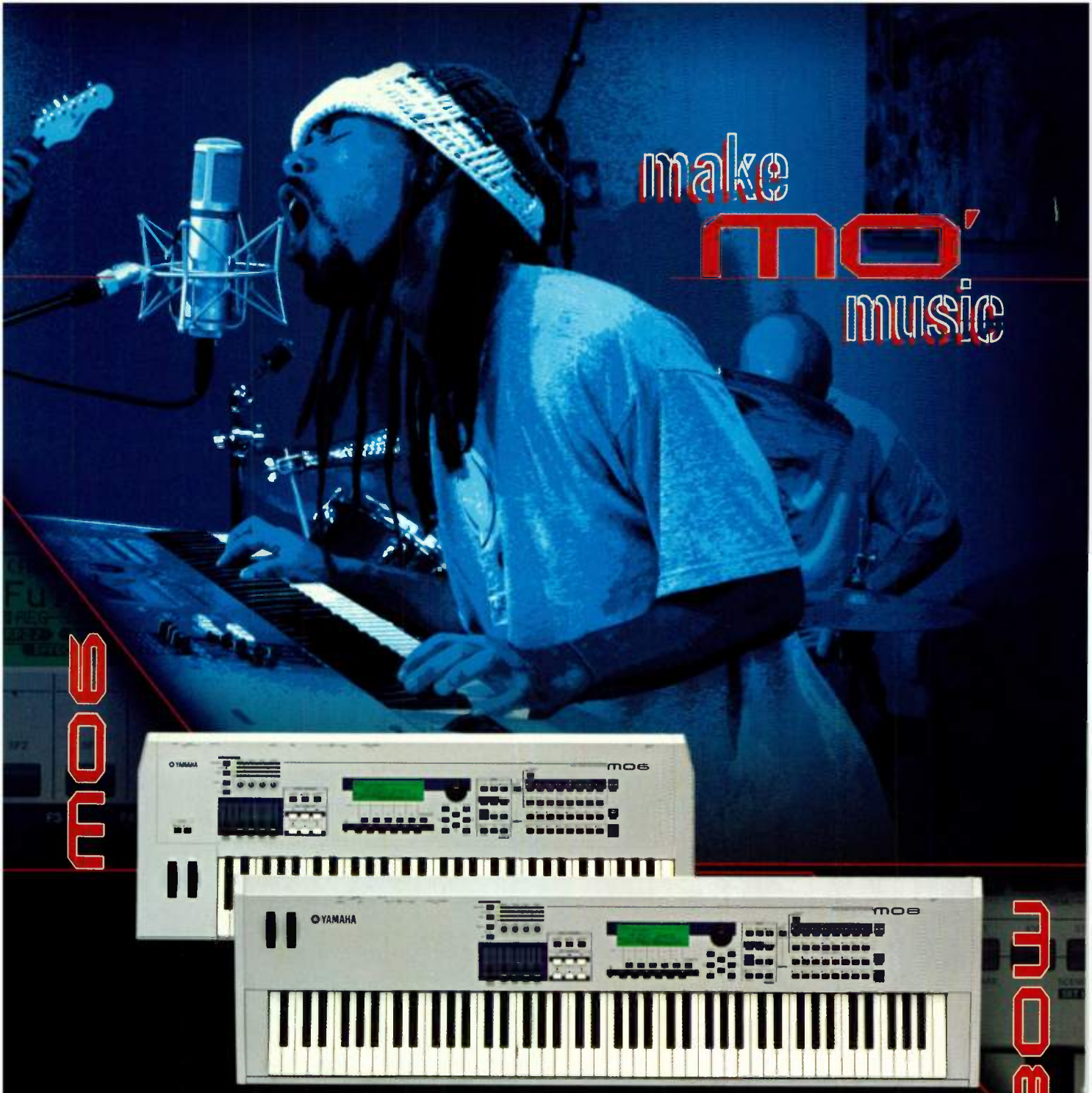
As we wrapped up the interviews, Newland and Ahern both offered advice to those who are new to recording acoustic guitar. "If I didn't know what mic to put on it and was recording a great player," suggested Newland, "I'd say, 'What works for you?' I would certainly ask the player what they've had success with. I think that people learning to record can learn a lot by asking other good players what helps. It's a good way to get another perspective and some extra knowledge."

Ahern brought the subject of recording acoustic guitar down to earth by cautioning not to get caught up in equipment specs and theory while tracking. "I always ignore math," he said flatly. "What sounds good *is* good. Period. Just turn the buttons till it sounds right." EM

EM contributing editor Michael Cooper (www.myspace.com/michaelcooperrecording) is the owner of Michael Cooper Recording in Sisters, Oregon.

RAFAEL SERRANO

Dave Way



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Monitoring for Success

By Charles Dye

Tools, tips, and techniques for productive listening during mixdown.

One of the ironies of mixing is that the more successful your work, the lousier the speakers on which it will be listened to. As a mix engineer, you have absolutely no control over that. But the one thing you do know is that your mixes will be heard on a wide spectrum of systems, including some of the worst. And knowing this allows you to wrest back control over your sound.

When I mix a song, my goal is to create a great-sounding mix: a big sound with expansive width; lots of depth and separation; full, rich lows; well-defined mids; and clear, open highs. But because I know my mix won't always be heard on the best of speakers, I work hard to ensure that all of the recording's key elements—the rhythm, harmony, and melody—retain a consistent musical balance no matter the speaker.

Many techniques with EQ, compression, and effects can be used to accomplish these goals, but processing is not the only tool. Shaping the mix actually begins with how it's *listened* to. And there are a number of monitoring techniques you can use to achieve a big, powerful mix that will sound great on any speaker, regardless of frequency response.

Getting out of the Woods

Have you ever noticed how when you listen to a mix in the morning, you're able to hear things you hadn't noticed the night before? It's an empowering moment, because all of a sudden you have a fresh perspective. Problems become clear in an instant and you can correct them with just a few changes. It's both relieving and frustrating. Why couldn't you do that last night?

This is the forest-for-the-trees effect. When you get to the studio in the morning, you're able to hear the big picture, and the problems are suddenly obvious: the vocals are hideously drenched in reverb, or the snare is just *way* too loud. So, how was it possible to be so amazingly deaf to those deficiencies during the previous night's session, yet so attuned to them the next day?

The answer is that you're listening with a clear head and your ears are no longer prejudiced from hearing the same thing over and over. But wouldn't it be great if you could re-create this eureka moment many times a day while mixing? Well, you can, and it can really speed up the process.

A Forced Perspective

During a mix session, your goal is to keep your perspective the whole time and to come up with a mix that sounds good on all speakers. That might seem like it's asking a lot, but both those things are possible. How? By emulating how others will hear the mix, and forcing yourself to listen in as many ways as you can: on different speakers, at different volumes, and from different locations in the room. This will make it possible to create a mix that will translate well to many speakers, while also continually giving you a fresh perspective—waking up your ears to problems you couldn't hear minutes before.

Certainly, it's important to have high-quality monitors and an acoustically well-treated control room. If you can't accurately hear what's going on, decision making will be difficult. And this is especially true while recording. But accuracy in mixing is a very relative thing, because nearly everyone will hear your mix anywhere *but* in a studio.

JON KAGI

FIG. 1: Charles Dye at Supersonic Studios in Miami, with three sets of monitors set up for mixing: Genelec 8050s, Yamaha NS-10Ms, and Sony SRS-88s. He doesn't like to use large, soffit-mounted studio monitors, such as those in the background.

FIG. 2: The Yamaha MSP10 Studio, a biamped active monitor (now discontinued), is Dye's top choice to fill the studio monitor role among the troika of speaker pairs he uses when mixing.

I have at least three different sets of close-field monitors that I switch between when working on a mix (see **Fig. 1**):

- High-end, self-powered studio monitors with extended lows
- Consumer-grade bookshelf monitors
- Real-world lo-fi speakers

Although I frequently work in studios that have large, soffit-mounted monitors, I don't usually find them as useful as close-fields when mixing.

At the High End

My powered reference monitors are the most accurate speakers I use when mixing, reproducing the highs and lows as they actually are in my mix. They're a great choice because they include not only the speakers, but the entire system including the amplifier and crossover as well. Quality powered monitors have matched components that sound their best together. Such monitors are one of my most important investments.

Models like the Genelec 8050 or 1031, Mackie HR824, and Yamaha MSP10 Studio (which I use; see **Fig. 2**) are all excellent. But there are many comparable brands and models, and self-powered isn't the only way to go. The real key is to have a pair of accurate monitors with extended lows; generally a pair with 8-inch woofers is needed for good low-end reproduction.

Oddly enough, I use these monitors the least—usually only while I'm working on kick and bass—but their importance can't be overstated. Having monitors

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that can properly reproduce your mix's low-end balance is essential when shaping this critical area. Lows can be difficult to judge, and without accurate monitors you can get unpredictable results. (Some people use systems with subwoofers for judging lows, but because I don't, I can't offer any concrete advice about using them.)

At the same time, however, great-sounding speakers can be quite flattering to your mix and potentially fool you into believing that things sound better than they actually do. This is not to discount these speakers, but I simply don't stay on high-end monitors for too long while mixing.

Off the Shelf

Once I've got my low end sorted out, I'll spend a lot of my time on smaller, less expensive, and more consumer-oriented monitors like Yamaha NS-10Ms. Their predecessor, the NS-10, was actually designed for home use, but it became popular with recording and mixing engineers in the '80s, and Yamaha released the NS-10M in response. Neither model is manufactured anymore, but they're readily available on eBay, and Yamaha's current HS50M is intended, in part, to fill the void left by the discontinuation of the NS-10M.

Now, NS-10Ms are not great-sounding speakers, but I find that their limited frequency response more accurately represents how the listener may hear my mix. And experience has shown that mixes I do on NS-10Ms usually translate very well to many other speakers.

The cliché goes something like "If you can get your mix sounding good on NS-10Ms, it'll sound great anywhere." And it's true, because NS-10Ms don't complement your mix with pretty top end and full lows. Instead, they make you work hard to get great sounds, and the results are definitely worth it.

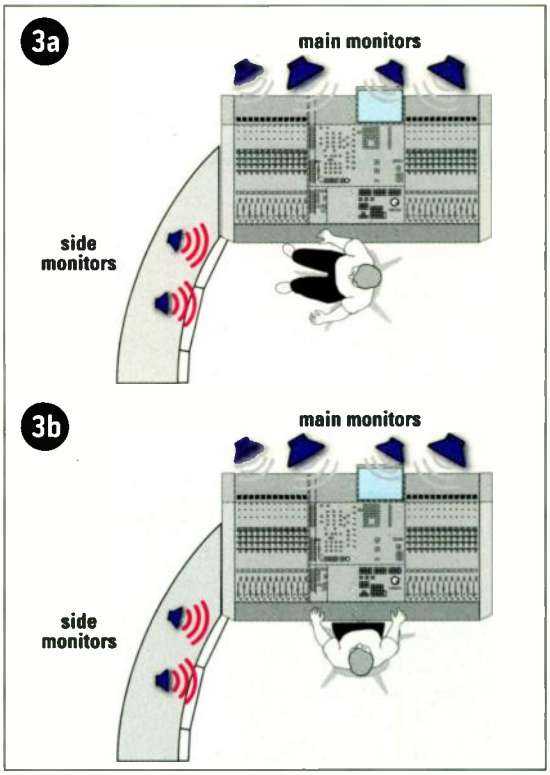
The strength of NS-10Ms is their pronounced high mids, which force you to focus on this important frequency area where so many instruments have their definition. The vocals, guitars, keys, kick, snare, hat, toms, and even bass all have upper mid frequencies that need to be managed in a way that will allow each of them to speak without distracting from the others.

And whether you use NS-10s, HS50Ms, or another similar monitor, the point is to use speakers without extended lows and highs to better craft the mids of your mix.

In the Real World

In the end, the speakers I spend most of my mix time on are a small pair of self-powered computer speakers. The pair I currently use are Sony SRS-88s, but the actual model is less important than the fact that they're extremely "real world," sort of like modern-day Auratones.

This type of monitor doesn't have any low end or high end to speak of, and you can't listen to them loud. So, all of a sudden your mix becomes totally about the music (as it should) and very little about pristine sound. As a result, these lo-fi monitors are perfect for balancing the instruments and doing automation.



FIGS. 3a and 3b: Placing an alternate set of speakers to the side of the mix position lets you swivel to face them so that you hear your mix interacting acoustically with a different part of the room (3a). If you listen to those same speakers while facing the main mix position, you're hearing them mainly out of one ear, giving you a simulated mono effect (3b).

As lo-fi as they are, these speakers work surprisingly well for EQ'ing, for the same reason the NS-10Ms work well in the high mids. Because these speakers have a pronounced woofy quality without much low end, they cause me to clean up my low mids on upper-register instruments. At the same time, they're great for making sure the kick and bass have enough low mids to cut properly on small speakers.

And because these speakers don't have much high-end response, they allow me to hear if instruments that are bright enough on high-quality monitors have enough definition in the upper mids to translate well to cheap speakers. Seriously, lo-fi speakers are really useful.

Mono a Mono

Besides using different speakers, there are other techniques you can employ to freshen up your view of the mix. The first is listening in mono. Many engineers swear by it, and there is a lot of validity to the technique. It works well when checking for phase problems—by listening for instruments that lose level. And it's a great way to

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force yourself to listen only to the balance and frequencies of your instruments, without being distracted by the exciting stereo panorama moving around you.

I don't like listening in mono; it takes the fun out of mixing, and for me, it's just not that useful. However, I will frequently listen in what I call *manual mono* (which I'll explain in the next section). And this brings me to the next technique: you may have noticed that when you take your mix out of the studio, even simply to a boombox in the next room, you hear things you hadn't noticed only minutes before. It's because the change in environment also gives you a new perspective.

I've found it's possible to actually re-create this effect in real time, while still tweaking the mix. Moving around from speaker to speaker as we've already talked about can jerk you into a whole new reality that gives you a fresh perspective of your mix almost every time you switch. But one thing isn't changing, and that's the room you're in.

Altered Space

When I have three sets of speakers set up in front of me, even though their sound changes as I switch between them, what remains constant is the impact of the room acoustics. This can cause three extremely different speakers to share common sonic characteristics. And that doesn't help the whole perspective issue.

But I have a solution for this problem. Instead of placing my lo-fi speakers in front, I place them directly to the left of my mix position. This causes two things to happen. First, with the speakers to the side I will sometimes turn and face them, which changes the entire color of the room (see **Figs. 3a and 3b**). The speakers now interact quite differently with the space, making it seem like I'm hearing the mix back elsewhere. I also listen to them while facing forward (toward the console and the other speakers). With only my left ear facing the speakers, I essentially perceive the mix in mono, and this is what I refer to as manual mono. It most accurately represents the way many people will really hear my mixes: as incidental music that's mainly in the background.

Does It Hurt Yet?

Ever notice yourself reaching for the volume knob when your mix doesn't quite sound right? When you do, it should wake you up to the fact that the real problem is the mix itself. This is when you should turn the volume down instead.

Mixing loud is the best way to get a result that will sound wimpy and flat when played back on small speakers at a low volume. Why? Because the number 1 rule of monitoring is, "Everything sounds great loud." Although blasting the playback works perfectly to make the drummer feel good after a take, it's pretty useless when you're trying to create a mix with any detail.

One reason is that our ears have built-in compressors, and at high volume they begin to distort. The louder the speakers get, the more your ears compress the sound, essentially impairing your ability to accurately judge dynamics. It can sound good at the time, but when you listen to it at more realistic volumes, your mix will be very different.

The frequency response of your ears changes as you turn up the volume, but you can use that to your advantage. By changing the level, you can alter the way your mind hears the mix. Like switching speakers, you can once again give yourself a fresh perspective.

Although listening loud for extended periods isn't generally a good practice while mixing, I do occasionally push the volume up, because louder levels help me more accurately judge low end. This is because the lowest frequencies are something you almost feel more than hear—literally in your chest. And you really can't sense this until you get some air moving.

Try a Little Quietness

Not only is an extended period of high-volume listening hazardous to your health, it will also temporarily burn out your ears. It will cause the highs to seem not bright enough, and instruments not loud enough. That is precisely why and when your mix will get out of whack—leaving you chasing sonic ghosts.

I find it's best to spend most of my time mixing at low volumes, what you would probably consider very quiet. I listen so quietly that even a whisper from the studio couch can be distracting. It might be difficult to get used to at first, but the longer you do it, the more you will realize how much easier it is to hear mix problems this way.

Try listening as quietly as you can. If you've never done it, I promise you'll hear detail you haven't heard before. And you'll soon learn your ears will have more productive hours in a day because of it. Listening loud can be exciting, but your emotions will often cloud your sonic judgment. Save it for the final playback.

TIPS FOR PRODUCTIVE MONITORING

- Switch between different types of monitors to get a variety of perspectives. Don't spend all your time listening to your expensive monitors.
- Set up one pair of monitors off to the side of your mix position. You'll get different room interaction than you do with your mix-position monitors, and you can listen mainly through one ear when facing forward for a pseudo mono effect, or swivel and hear them full on.
- Don't mix with your monitors at loud volumes except for when you need to judge low end.
- Vary the levels at which you listen, and spend a lot of time with the volume turned low.
- Occasionally walk away from the mix position and listen from another part of the room or from right outside the door.
- Take short breaks to regain your perspective.

Something Completely Different

Here are several low-tech but time-tested techniques for reinventing your listening environment. One of the simplest and most revealing is to get out of the hot seat and listen while relaxing on the studio couch. Or, if your studio doesn't have a couch, simply stand off-axis elsewhere in the room. The mix position has a tendency to put you in a very reactive state, constantly wanting to fix every problem you hear. But when you move away from it, the fact that you can't reach the knobs frees your mind to hear the mix in a new way.

Another method is to leave the environment completely. Get out of the studio for 15 to 30 minutes, just long enough to clear your mind completely of whatever you were zoned in on. When you come back, you'll be able to hear the whole mix, not just that hi-hat you were tweaking. Or, simply remove the environment from the equation altogether—by listening on headphones. It's a completely different way to hear your mix, and with all the iPods out there, it's how many others will hear it too. You can even try listening with iPod-style earbuds.

The next two suggestions may seem quirky, but they're effective. They're known as the "finger test" and the "door test." For the former, you listen to the mix with your fingers in your ears. Really. And for the latter, you step outside the control room and listen back to the mix with the door closed or just slightly ajar. The effect for both is similar. By eliminating the top end in the mix, you can tell if all the instruments are well balanced. That is because all you can hear are the fundamental frequencies. It's not something you need to do very often, but a finger or door spot check can be very illustrative.

To review, listen to your mix on different speakers, at different volumes, and in different environments, and change between them often. The idea is to hear something unexpected, and there is no better way to do that than by changing your aural point of view. Of course, literally taking your mix out of the studio (like listening on your car stereo) can be helpful too, but the advantage to the techniques I've described is that you not only get a fresh perspective, but you also can tweak your mix while you're listening. And if you can get it sounding good in every situation, then you've got a great-sounding mix. EM

Grammy-winning engineer, mixer, and producer Charles Dye (Lauryn Hill, Shakira, Ricky Martin, Jon Bon Jovi) adapted this article from his DVD/DAW session-based mixing course Mix It Like a Record (www.likearecord.com).

The Low Road

By Steve Skinner

Recording synth-bass parts that enhance your mixes.

The three ingredients that are essential for creating a convincing synthesized-bass part are programming, playing, and mixing. I describe how to program a killer bass sound in this month's "Sound Design Workshop" column (see p. 70). Here, I'll discuss playing bass tracks and placing them properly in a mix.

I'll assume that you are playing or programming your bass line into a MIDI sequencer. If you have some keyboard chops, I highly recommend that you play the part yourself rather than step record or draw it in. There's nothing wrong with using both hands to play the part, and you can slow down the tempo of your sequencer to play more-difficult passages. Dance and industrial bass lines are exceptions, because they need to have the machinelike precision of a step-entered or quantized part.

First Things First

It's best to have your basic drum and keyboard parts before you record the bass line. If you also have a reference vocal, that's even better. A good bass line works around the vocal and pops through in the spaces between vocal phrases.

Once you have played your part into the sequencer, solo it and fix any obvious mistakes. Get rid of extraneous notes and spend the extra time necessary to ensure that the note lengths are just right; an undesired gap between notes can create an unsettling effect. When notes of the same pitch overlap, the second note will be stopped by the first note's MIDI Note Off message and will thus be truncated. That can happen, for example, when a bass line is quantized and the first note has been played early. If notes of differing pitches overlap, you may get a moment of mush as the two notes play together. To fix that, either shorten the duration of the offending notes or set your synth's keyboard mode to monophonic.

FIG. 1: Waves Renaissance Bass with a moderate bass-enhancing setting.

Whether to quantize, and by how much, is a matter of personal choice and differs by genre. With dance music, quantizing is the rule, whereas with hip-hop and rap, it isn't. If you are using a bass sound that has a strong attack, line it up with the kick drum, regardless of whether it is quantized. That will avoid a flamming effect in the low-middle range, which weakens the impact of bass and kick.

Experiment with portamento in your bass line and practice using your keyboard's pitch-bend wheel. Also experiment with the retrigger setting if your synth has that. With retriggering on, each note you play has a new attack; with retriggering off, notes that are overlapped do not have a new attack. That allows you to create gliding effects and to separate notes by changing your playing style. Portamento can often be set to kick in only when you play legato. You can program very expressive bass lines by alternating legato and staccato playing, by using the pitch-bend wheel, and by keeping a hand on the filter cutoff frequency.

Recording the Bass

If you are using a hardware synth, you need to find the best way to get its sound into your digital audio sequencer. The easiest way to do that is to plug the synth into one of your audio interface inputs and set that input at a –10 dB level; that may not give you the best possible sound, though. If you have a good standalone mic pre with a DI (direct injection) input, experiment with plugging the synth into it and running the mic pre output into a line-level (+4 dB) input of your sequencer.

I like to add a little hardware compression to the bass during tracking. Try 2 dB of gain reduction, hard-knee compression with a compression ratio in the 3:1 to 6:1 range, and a 20 to 40 ms attack. The slow attack allows the initial kick of the bass sound to come through, whereas the rest of the note is smoothed out and less woofy.

If you are using a software synth, bus the signal to your bass track and record it, leaving compression and EQ for the mix. Alternatively, you can run the sound out an

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analog output, through a hardware compressor, and back into an analog input of the audio interface. This gives you that analog sound, but you lose a little presence as a result of the digital-to-analog-to-digital conversion.

When setting up a compressor, set the gain so the volume is the same whether the bypass button is in or out. You can't tell if the sound is improved by the compressor unless the levels are matched.

FIG. 3: Bomb Factory BF76 showing my favorite second compressor setting.

Effects

I rarely use time-based effects such as chorus and delay with synth bass. Those effects might sound cool when the bass is soloed, but they make the bass sound mushy and can cause big problems in a mix. On the other hand, you might add some delay to a Jaco Pastorius–style bass sound. Pastorius used lots of delay, with pitch modulation, when he played flowing, melodic parts.

One of my favorite effects plug-ins is Ohm Force Frohmage, which you can download for free from www.ohmforce.com. It's a funky-looking filter plug-in with enigmatic markings. Without worrying about the details, I set the small knobs to around 12 o'clock and automate the large cutoff-frequency knob. Start with a bright, annoying bass sound; put Frohmage on the track; and automate the cutoff frequency to get a really expressive bass sound. This technique is most effective when the bass is in a featured role.

FIG. 2: Waves Renaissance Channel showing my favorite bass EQ and compression settings.

Mixing

I've become very attached to using Waves Renaissance Bass (RBass) in my mixes; it's usually the first element in my bass signal chain. RBass boosts the first few overtones of the bass without boosting the fundamental. The ear infers the fundamental from those overtones, and you get a more apparent bass without adding a lot to the overall volume of the mix. It's also easier to find the best bass mix level with RBass (see **Fig. 1**).

Fig. 2 shows Waves Renaissance Channel with my favorite bass EQ and compression settings. I place this after the RBass. The 350 Hz dip in the EQ takes out the unclear part of a bass sound that lies between the bottom and the definition; the boost at 1 kHz increases that definition. Experiment with the frequency of the boost—different bass sounds will be defined by different frequencies. Be careful that the frequency you boost does not interfere with guitar and lower keyboard parts.

I use roughly the same compression settings for mixing as for tracking (see Fig. 2). Set the threshold so that the gain reduction is no more than 3 or 4 dB. Experiment with the attack time to see how it affects the attack portion of the bass sound. Experiment with EQ and compression while the whole mix is playing and with the bass soloed.

I often insert another compressor, Bomb Factory BF76, at the end of the signal chain (see **Fig. 3**) because it adds extra punch to the sound. **Web Clip 1** is a Minimoog bass line without processing. **Web Clip 2** is the same track with processing. The difference is subtle, but you can hear how the bass sits more evenly in the mix with effects processing.

Finally, for the bass to punch through a mix, you need to clear the other instruments out of its way. Put highpass filters that have 24- or 36-dB-per-octave slopes and cutoff frequencies that are around 100 Hz on any tracks other than the bass or kick that have a residual low end. The bass and the kick drum should be the only parts that have really low frequencies. My thanks to producer-engineer Carlos Alvarez for that tip.

The hardest part of creating a good mix is getting the vocal level right; the second-hardest part is getting the bass level right. Because rooms vary widely in their bass response and different areas of the same room will sound quite different from each other, it's important to check the mix in as many environments as possible: in the living room, the car, and, my favorite, through an iPod with Bose Noise Cancelling headphones. The bass response of earbuds varies depending on their position in your ears, so I don't like them for checking mixes.

A bass sound is ultimately measured by how it works in relation to the rest of the track. So pay careful attention to the rest of the mix along the way. EM

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Steve Skinner has worked as an arranger-programmer for Celine Dion, Jewel, R. Kelly, Chaka Khan, Bette Midler, and Diana Ross. He arranged the musical Rent *and coproduced the cast album.*

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Synth Bass-ics

By Steve Skinner

How to create convincing electronic bass sounds.

The key to programming convincing synth-bass sounds is simplicity. I use one oscillator with a medium-width pulse wave and a lowpass filter whose envelope has a quick attack, a medium to quick decay (depending on the tempo), and a quick release. The volume envelope's sustain level and decay time depend on the style of music. For R&B and hip-hop, I use maximum sustain; for ballads and gentler music, I use less sustain to leave room for the other instruments.

My favorite synth for bass sounds is my '70s Minimoog with its MIDI retrofit. The inconsistencies in analog circuits produce subtle variations in the sound that keep the ear interested. As an alternative, I frequently use GForce Virtual Instruments ImpOSCar or Access Music Virus Indigo virtual synth plug-ins. Because the programming of the bass sound is fairly simple, the choice of synth is quite important. **Web Clips 1, 2, and 3** realize the same bass line using similar settings on the Minimoog, ImpOSCar, and Virus Indigo.

Simple Settings

Trying to concoct massive bass sounds by layering multiple synthesizers almost never works. The multiple attacks blend to make one fuzzy attack, and because of the phase differences among the synths, the low end is inconsistent. I use one synth with one oscillator. I have combined multiple bass tracks by recording four passes of my Minimoog playing the same MIDI part with the same sound. In that case, the attacks lined up and the analog oscillators' subtle differences in tuning were enough to spread the sound just slightly. The bass sound in Devo's "Whip It" was created with multiple Minimoogs linked together.

A bass sound that works well in a mix usually doesn't sound good by itself, and a bass sound that is complex enough to sound good by itself often takes up too much room in a mix. Therefore, it's a good idea to program the bass sound while the track is playing. When I record MIDI, I play the part with a sound that I know is close, and then I tweak the sound as the MIDI sequence plays back. I may even change the bass sound during the final mix.

Different styles require different bass sounds. For hip-hop and reggaeton, the sound starts with a sine wave or something close to it. For example, a square wave followed by a 4-pole lowpass filter with a low cutoff frequency is similar to a sine wave but has more definition. Experiment with resonance as well (see **Fig. 1**). For pop, rock, and ballads, a sine-wave-based sound sometimes works, but you might also try a pulse wave with the filter more open. For those styles, use a slow decay.

For dance tracks, use a brighter sound with a more open filter and even multiple oscillators tuned in octaves. To avoid phase problems when using virtual analog synths, use little or no detuning between the oscillators. When using a true analog synth, sync the oscillators. When the bass sound needs to be bright and function as a midrange part as well, use four to six oscillators tuned in unison and pan them widely. In dance music, the bass usually has a quick decay.

Not-So-Simple Settings

For a more fun bass sound, try increasing the filter resonance and modulating the filter cutoff frequency with a MIDI controller (mod wheel, pedal, or some such). Work the filter modulation with one hand (or foot) as you play the part with the other. (See "Making Tracks: The Low Road" on p. 66 in this month's EM for a description of how to use a filter plug-in to add expressiveness.)

There are times, of course, when a simple sound doesn't work. You can stack oscillators, combine synths, and open your filters wide with a mix that has little more than the bass line. Experiment with hard-syncing the oscillators, and then modulate the pitch of the synced oscillator with an envelope, LFO, or mod wheel. Try using samples meant for something else—electric piano or pitched drum, for example. In short, be as creative as the sonic space allows. EM

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FIG. 1: The ImpOSCar settings as heard in Web Clip 2 have a deep, square-wave sound. Only one oscillator is on.

Steve Skinner has worked as an arranger-programmer for Celine Dion, Jewel, R. Kelly, Chaka Khan, Bette Midler, and Diana Ross. He arranged the musical Rent *and coproduced the cast album.*

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Get Real

By Jim Aikin

How a sampler does its magic.

Until the late 1970s, synthesizers were used primarily to create new and unusual types of sounds. But by 1980, keyboard players were clamoring for synths that could stand in for traditional instrumentation such as string and brass sections and electric bass.

The synthesizers being built at that time weren't especially good at imitating real-world sounds. To fill that need, companies such as Akai, E-mu, Ensoniq, and Fairlight developed the first samplers. A sampler can sound exactly like a string section, a solo French horn, or whatever else you need, because it plays samples (digital recordings) of the real thing.

Problem solved? Not exactly. For all its power and promise, sampling technology is far from perfect. In this column, I'll describe what can go wrong when you play the keyboard of a sampler.

Because most samplers today are capable of ultra-high-fidelity recordings, I won't discuss bit rate or sample resolution, which were major issues in samplers in the 1980s. Unless you're using a very old piece of gear, your sampler or sampling software can record with at least 16-bit resolution at the standard CD sampling rate of 44.1 kHz, so achieving good sound quality won't be difficult.

Terminology

The word *sample* refers to a complete digital recording (for example, a recording of a trumpet playing a single note). The same word is sometimes used to refer to a single numerical value that makes up the recording. To avoid confusion, we'll refer to such values as *sample words*.

Hardware-based samplers, including modern workstation keyboards like the Yamaha Motif and Alesis Fusion, have audio inputs to allow them to record new samples. Only a few software-based samplers, however, are capable of recording directly. Software instruments are generally used in a computer audio environment, in which some other program does whatever recording is needed. Calling an instrument that can't sample a sampler isn't quite correct. Nevertheless, the word is used when referring to software programs that can play back any digital audio file you choose, typically under MIDI control.

A *sample-playback synthesizer*, in contrast, makes sound using a fixed library of samples. It does not allow you to load your own.

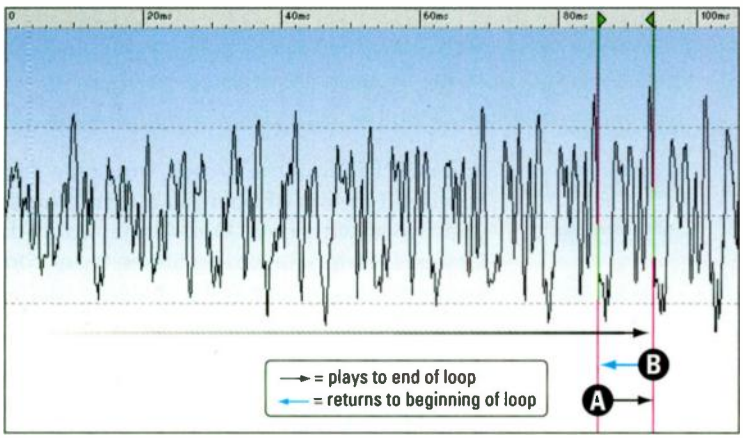

FIG. 1: When a sample is looped, playback starts at the beginning, proceeds to the loop end point (b), jumps back to the loop start point (a), and continues playing the looped range until the key is released.

Tailspin

Early samplers had extremely limited amounts of memory compared with modern instruments. To fit a number of samples into memory at the same time, each sample had to be kept very short. But musical notes are often many seconds in length; for notes to be sustained as long as necessary, samplers *loop*.

Looping is also used to repeat sampled drum beats, but that's a more recent musical development. The process here consists of repeatedly playing back a sample that contains a single note, such as a note played on a trumpet or an organ.

When a key is pressed, the assigned sample starts playing from the beginning. When the *loop end point* is reached, playback jumps back to an earlier spot called the *loop start point* and continues as before. Each time the end point is reached, playback jumps back to the loop start point (see **Fig. 1**). This continues until the key is lifted and the amplitude envelope generator shuts down, allowing the note to die away.

Many samplers offer even more-complex looping schemes. For instance, loop playback can occur in a back-and-forth fashion rather than jumping from the end back to the loop start point. Back-and-forth looping, or *bidirectional looping*, can sound smoother than ordinary looping because there's no jump.

No matter how it's done, looping is not a perfect way to create a sustained sound. The audio in a loop is static

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and unchanging, while the tone of real instruments is not. And finding loop start and end points that create a smooth loop is not easy (see **Web Clip 1**).

Modern computer-based samplers can play looped samples, but many of them are capable of dispensing with looping entirely. Because the computer has lots of memory and can play audio directly from its hard drive, the sampler may *stream* (play back directly from the hard drive) samples from the drive as the keyboard is played. This works well with sounds that die away naturally, such as piano notes, but not so well with sounds like string orchestra, which may need to sustain a single note for an unknown length of time. The latter will usually be looped.

Transposition

When you run your fingers up and down the keyboard of a sampler or sample-playback synth, you'll probably hear a chromatic scale. When you play higher on the keyboard, the sample plays back more quickly, which raises its pitch. When you play lower, the sample is played more slowly, lowering the pitch. What could be more natural, right?

But getting a digital recording to play "faster" or "slower" isn't simple or natural at all. Unlike a turntable, which can speed up and slow down and thereby alter the pitch, the sampler's audio output runs at a fixed sampling rate—for example, at 44.1 kHz. That rate, which is the speed at which the sample words are sent to the audio output device, can't change. If you want to play the sample more quickly, you can't jam the sample words within the recording closer together, so you have to skip some of them entirely. If the sample is to be played more slowly, the existing sample words can't be spaced farther apart. Instead, new sample words must be found so that the recording as a whole will last longer.

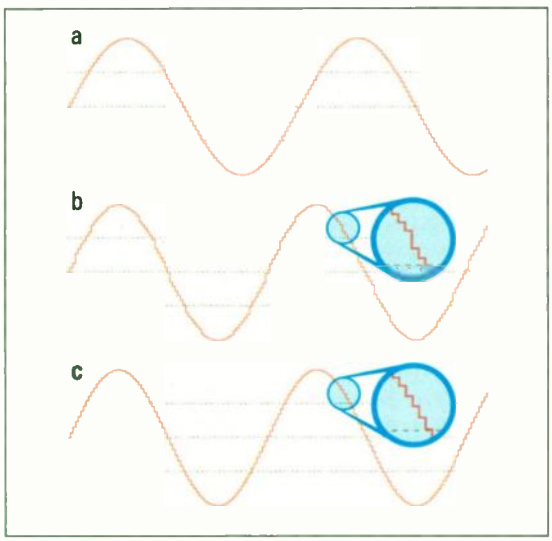
First-generation samplers performed this type of transposition in the simplest possible way: if the sample needed to be made shorter, some of the sample words were simply dropped. If the sample needed to be longer, certain sample words would be repeated.

This method, called *drop-sample* transposition, distorts the shape of the waveform, as seen in **Fig. 2**. This distortion is one reason that older samplers have a distinctive sound.

Modern samplers perform mathematical *interpolation* to make the waveform longer or *decimation* to make it shorter without distorting its shape noticeably. For every sample word that's added or dropped, the sampler adjusts the values of one or more adjoining samples to keep the waveform smooth. The distortion caused by changing the pitch of a sample is no longer an issue that we need to worry about.

Transposition causes other problems, though. If you sample a single midrange note on a guitar or almost any other instrument and then transpose it up or down over several octaves, you'll discover that the farther from its original pitch the note is played, the less realistic it sounds (see **Web Clip 2**). To understand why, consider the *snap* of a guitar pick hitting a string. That snap is part of the sampled sound. If a sample of a guitar string being plucked is transposed down an octave, the tone of the string may still be acceptably realistic, but the snap will be too low pitched and will last too long. It will sound as though it were made by a large, soft pick. If the sample is transposed up by too many half steps, the snap sound will become high pitched and brittle, as though the string were being picked by a thumbtack. Again, the sample won't sound much like a real guitar.

Some sounds survive transposition better than others. Mallet percussion, for instance, can often be transposed over a couple of octaves before it starts to sound unacceptable. The human voice, on the other hand, can be transposed by no more than a few half steps before it becomes comical. That is because all of the frequency components of the sample (its *partials*) are transposed up or down by the same amount.



CHUCK DAHMER

FIG. 2: Fig. 2a shows a 500 Hz sine wave. Fig. 2b shows the same sine wave transposed up to 583 Hz by dropping samples. Fig. 2c shows a 583 Hz sine wave created using interpolation; it has no dropped samples and is therefore smoother.

Trickery

To deal with the sonic problems posed by sample transposition, instrument designers created a technique called *multisampling*. With multisampling, a number of separate samples of the same source instrument are recorded playing single notes at a number of different pitches. These samples are then assigned to separate *zones* (ranges of notes) on the keyboard. Multisampling is not a perfect solution, however. Professional sound designers spend lots of time matching samples up and down the keyboard. But a well-produced multisample can sound amazingly realistic across a wide range. EM

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Jim Aikin writes about music technology, plays electric cello, and still has time for rambunctious political opinions. Visit him online at www.musicwords.net.

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The Lowdown on Downloads

By Jeffrey P. Fisher

Selling and marketing your music online.

As CD sales continue to decline, digital downloads are rising sharply. The growth of the iTunes Music Store and similar services points to a paradigm shift in the way that listeners discover, purchase, and play their favorite music. If you're an independent musician, it's now essential to make your music available on the Web. In this column, I'll explore some methods for doing that.

Web Way

Selling physical CDs and downloadable MP3s on the Web makes sense for every musician. Domains, hosting space, page authoring, and payment processing are easily available and managed. The real challenge is driving traffic to your site. Getting your fans and postgig surfers to go there is relatively easy, but the chance of a casual music listener stumbling across your site is rare.

For that reason, being part of a music portal makes sense. These sites are populated by listeners who gather to buy music, and they are crucial places for hooking potential purchasers. If your music is readily available at these sites, you will also have the opportunity to be mentioned in the "listeners also bought" listings, which can spur sales.

MySpace is one such portal. Previously, the site didn't allow artists to sell from it directly, and users had to drive traffic elsewhere for commerce. Recently, however, MySpace announced plans to allow indie artists to sell their music from the site. The artists can set their own rates but will be able to sell only non-copy-protected MP3s. With more than 100 million MySpace members, the popular portal is poised to take on the industry heavy hitters.

FIG. 1: Although it took effort and persistence to get approval from Apple's gatekeepers, Dominique Vouk's debut CD was accepted on the iTunes Music Store.

I Want My iTunes

Gary Bosko (www.pivotentertainment.com) of Pivot Entertainment has learned much about selling music on the Web. When he embarked on promoting Dominique Vouk's (www.dominiquevouk.com) self-titled debut album (see **Fig. 1**), he knew that digital distribution was an area worth focusing on. His first stop was CD Baby, the industry stalwart dedicated to indie music. "I submitted Dominique's music to CD Baby," Bosko says. "Part of that process meant indicating other places I'd like to have consider her music, including iTunes."

Getting Vouk's album onto CD Baby was easy, but iTunes was trickier because the CD first had to be approved by Apple's staff of music gatekeepers. "I stayed in constant contact with the people at CD Baby, showing how we were out there promoting the CD, both off- and online," he says.

Bosko even started calling Apple regularly, though he never reached a live person. "I didn't receive any direct feedback from Apple, but I think they took notice of my persistence and how serious I am about promotion. I'd heard that the iTunes office is understaffed and picky about what they accept. We'd put together a slick album with great songs and an interesting artist in Dominique. They must have liked it, because within six months we were on iTunes."

So far Bosko has been quite excited by his good fortune. "We have had lots of positive reviews, and we see our music purchased right alongside major-label artists. I haven't seen our first earnings report, though."

According to Bosko, the key to getting accepted on iTunes is being professional and serious. He moved forward with CD release parties, concerts, and other promotions regardless of what happened online. "Getting on iTunes, while a really big thing today, is just another promotional piece. We're also distributing ringtones on phone company sites. You can't sit back and relax just because iTunes picks you up. You have to be out there pushing all the time."

Like all artists with a CD, Vouk sells her music at gigs. "We've noticed a clear trend," Bosko observes. "Fans tell us they don't buy physical CDs—just downloads." To solve that problem, Bosko purchases and resells iTunes gift cards. "We run promotions such as 'buy a card and get a free T-shirt.' That way the fan gets a card at the show and

"I Got a $ix-Figure Indie Label Deal Because I Joined TAXI"

Jenna Drey – TAXI Member – www.jennadrey.com

My name is Jenna Drey. That's me sitting next to TAXI president, Michael Laskow.

For as long as I can remember, I've wanted to be a recording artist. I've studied music my whole life. I've read all the books. I've been to the seminars. In short, I've done all the same things you're probably doing.

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Blish (www.blish.com) specializes in selling digital content of all types. It handles all the technical integration (getting your music onto its site) and search-engine marketing, but it takes a 50 percent cut of all sales. Most of the music it has handled has been of the royalty-free production variety, but it's now seeking more indie CD material. Its agreement is nonexclusive, so you can sell your music in other places as well.

Broadjam (www.broadjam.com) provides Web-site hosting and music-distribution services, with the lion's share of money earned going directly to the artists. It has several tiers of membership options that offer a bevy of services to indie musicians.

CD Baby (www.cdbaby.com) sells physical CDs and is the largest Web retailer of independent music. After a $35 setup fee, CD Baby works to sell your music on its own site and offers digital distribution services to places such as iTunes, Yahoo Music, Napster, and more. CD Baby can make your CD available to more than 2,400 traditional U.S. CD retail stores. The site keeps $4 per CD sold and 9 percent of the proceeds of tracks sold through digital distribution. Artists get the rest.

GarageBand (www.garageband.com) has a novel twist on its catalog: it relies on listeners to pick the best emerging music. Its vision is to redefine how music is discovered and promoted.

Groovegate (www.groovegate.com) is open to existing labels and unsigned artists, with an emphasis on club, dance, and DJ content.

The Independent Artists Company (http://iacmusic.com) offers free Web pages to musicians, with additional perks for those who take out a paid membership. It uses a radio station motif in which artists and listeners build playlists as a way to share and promote music.

OmStream (www.omstream.com) focuses on finding music that "elevates consciousness amid the clutter of songs available [from] other download services." It emphasizes world and new-age music.

Pump Audio (www.pumpaudio.com) provides indie music to clients who need tracks for use in their radio, TV, advertising, games, and other off- and online media projects. Independent artists and labels can submit music for consideration and possibly earn income from commercial uses of their compositions.

SoundClick (www.soundclick.com) is a music Web site that features signed and unsigned bands. It offers Web pages, MP3 sales, message boards, and more. Labels can even broadcast videos in broadband quality and sponsor sweepstakes for artist promotions.

can then download the album from iTunes at home. So, we're essentially selling digital distribution at gigs.

"We're looking at putting together a promotional postcard with Dominique's picture on it, which she'll autograph at the gig. On the back of the postcard will be the codes fans need to download her music."

Burn, Baby, Burn

Scott Liebenow was searching for another grassroots platform for distributing children's and Christian material on his label, Little Man Music. He found it in BurnLounge (www.burnlounge.com/lmmusic).

For a yearly fee, BurnLounge lets you create your own online music store where visitors can browse, sample, and buy music. Downloads sold from your BurnLounge store earn BurnRewards that are redeemable for music and merchandise. Sell a BurnLounge site to other customers, and you earn BurnRewards from their sales too. It functions as a multilevel marketing outlet. For an additional $6.95 per month, you can convert your BurnRewards into cash.

"BurnLounge lets me create a business that sells my own and other people's music, and I get paid for doing it," reports Liebenow. "They have a lot of tools and offer support for building a viable retail site."

Liebenow uses MySpace and a similar Christian site (www.coolchristianfriends.com) to connect with buyers. "I use both of these sites to promote my artists, report news, highlight events and gigs, and then send people to my BurnLounge store to buy music." Liebenow advertises his store on his physical CDs and on other promotions.

"I also market through BurnLounge by sending email blasts encouraging people to preview and download from my site. Called Bonfire, you can create music playlists and email them to people, saying something like 'Here are ten great songs you can preview, buy, and download.' And if they decide to buy something, I get a piece of that pie too."

What Liebenow likes most about being on BurnLounge is how he can feature his own music first, and then choose complementary music as well. "I can display my music next to the big stuff. It's a great way to digitally distribute for indie artists."

Bosko uses BurnLounge as a way of involving other people in the distribution of his music. "It's the ultimate street team where other people can help you promote—and they actually make money doing it. On iTunes, you can't change anything on your page. BurnLounge gives you some control of the store, the look, the featured material, and more. That makes it worthwhile to be there."

Click Here

"All these alternative music-distribution outlets are essentially the same," says Bosko. "If you aren't promoting, generating buzz, or sending people to the sites, nobody will buy your music. People either surf the Net to get something specific or for general entertainment. You have to make sure you are their destination in either case." EM

Jeffrey P. Fisher's best-selling book Ruthless Self-Promotion in the Music Industry *(Artistpro, 2005) now has a new second edition. Get more information at www.jeffreypfisher.com.*

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WAVES GTR 2.0 (Mac/Win)

A guitar plug-in suite with a hardware interface.

By Orren Merton

Waves and Paul Reed Smith Guitars are both companies renowned for producing high-quality products. Their collaboration on Waves GTR produced a hardware-software bundle designed for guitar recording. Waves GTR 2.0 is an upgraded version that was released earlier this year. It adds a number of new amp and cabinet models, among other improvements.

The bundle consists of the Waves PRS Guitar Interface (WPGI) and a collection of plug-ins that includes Waves Tuner, a fully featured chromatic tuner; Waves Amp (see **Fig. 1**), which gives you models of 10 amplifiers, 15 speaker cabinets, and 9 microphones; and Waves Stomp, a virtual pedalboard that can be filled with either 2, 4, or 6 (depending on which configuration of Stomp you open) of 23 stompbox-style effects. Each of the plug-ins in GTR 2.0 offers mono, stereo, and mono-in/stereo-out configurations. Waves Amp also features a dual-cabinet configuration, which routes the guitar amp model's output through two cabinet models in parallel.

FIG. 1: The Waves Amp plug-in is the centerpiece of GTR 2.0, offering 10 amp models and a host of speaker cabinet emulations.

Plugging In

Clearly, Waves understands that a great guitar sound starts with a great guitar signal. The WPGI (see **Fig. 2**) includes a solid-state preamp that can convert your signal from unbalanced to balanced. It gets the best results with a balanced mixer input or audio-interface input.

The WPGI operates with either two 9V batteries (included) or a 12V DC adapter (optional), and it has a ground-lift switch in case you are experiencing ground loop noise in your signal.

From an I/O standpoint, the WPGI has a 1 MΩ,

GUIDE TO EM METERS

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

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AT4047/SV
Vintage F.E.T. Sound

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AT4060
Modern Tube Technology

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(40) SERIES

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unbalance ¼-inch input that's impedance matched for passive guitar pickups and can output either a balanced XLR or unbalanced ¼-inch signal. Its XLR output can be set to either line (+4 dB) or mic (–28 dB) level.

I was surprised by the lack of a balanced ¼-inch TRS output. Waves recommends sending a balanced line-level signal to your DAW, but most line-level inputs on affordable audio interfaces use TRS inputs. In my case, I used an XLR-to-TRS cable in order to connect the XLR output from the WPGI to a line-level TRS input on my RME Fireface 800.

I wish the WPGI had a direct USB connection to the computer, because that would eliminate the need for an audio interface altogether. Incorporating USB would make the WPGI far more useful with a laptop or live rig.

The WPGI has minimal metering. Its 3-level indicator LEDs show if a signal is present, is at a nominal level, or is overloading. No dB indications are included on the meter. I would have preferred at least six LEDs; dB indications; and a dB scale around the gain knob.

While I have some reservations about the design of the WPGI, I have none about its sound quality. I own a number of rather expensive DI boxes, and the WPGI held its own sonically with all of them.

I tried out the WPGI with a number of different guitars, including a Koll Tornado and a Gretsch Duo-Jet (both with FilterTron pickups); a humbucker-equipped Patrick Eggle Berlin Pro V; and a single-coil Fender Telecaster. In each case the WPGI output a strong, clean signal. It had gain to spare, and introduced no additional noise of its own even when turned up. I was very impressed with how good my guitars sounded through it.

Tune Me Up

Tuners are not the most exciting of plug-ins, but they're a necessity. The excellent Waves Tuner features a pop-up menu with various modes for chromatic, guitar, or alternate tunings; you can adjust the tuning reference plus or minus 6 cents; and you can toggle notes on or off.

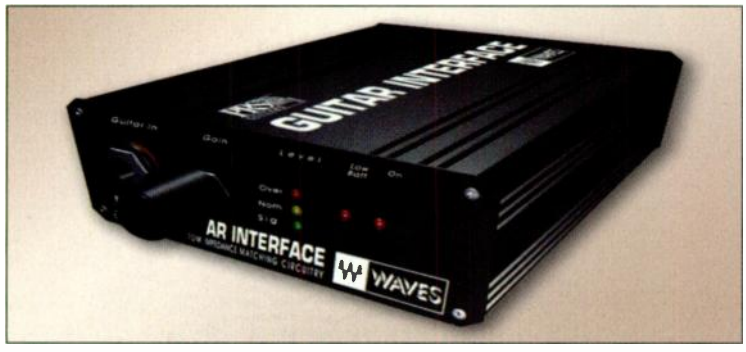

FIG. 2: The Waves PRS Guitar Interface (WPGI) direct box ensures that you can get a clean, good-sounding guitar signal to the plug-ins.

FIG. 3: The Waves Stomp plug-in comes in 2-, 4-, and 6-pedal versions. You can arrange the stompboxes any way within the given pedal slots.

Those features are all solidly implemented, but what makes this tuner stand out is its response. Many software tuners are too jerky to be useful. I've found them to be so sensitive to even inaudible variations in pitch that the tuner jumps from value to value faster than I can tune up. But Waves Tuner captures that magical balance between accuracy and speed; it responds to changes in string pitch quickly enough to be accurate, but not so fast that I feel broadsided by the information.

Getting Amped

Waves Amp has a clean and easy-to-understand user interface, which, unlike some other amp-modeling software, doesn't change appearance to reflect the selected amp or cabinet model. Pull-down menus let you choose the amp, cabinet, and microphone model.

All of the amp models have controls for drive, EQ, and presence. You get a vintage-style VU meter for master volume, below which sits a master volume knob. Also included is a bypass switch and a standard Waves control bar for preset loading, saving, and A/B comparison.

The names of the amplifier models—Clean, Sweet, Edgy, Cream, Drive, OverDrive, Crunch, Hot, Modern, and Direct (a direct box model)—don't specify which actual amps they were modeled from. However, there are enough clues to deduce that they're emulating Fender, Vox, Marshall, and Mesa/Boogie amps, as well as other vintage and modern classics.

The ten amplifier simulations make it possible to generate tones that range from sparkling clean to heavily distorted. In general, the Clean, Sweet, Edgy, and Drive amps are fantastic. The clean and edge-of-breakup sounds they produce are better than in any other simulation I've heard. Adding to the realism, these models are very touch responsive and dynamic. The Edgy amp is my favorite. It gets some amazingly open, rich, and dynamic Vox-like tones (see **Web Clip 1**).

The more heavily distorted amp models sound overly compressed to my ears, and in some cases have an odd, nasally, notch-EQ'd tone that real high-gain amplifiers don't. Nonetheless, with careful tweaking I was able to get some believable tones out of the Crunch and Hot simulations.

Guest Speaker

The software includes 15 simulations of speaker cabinets, along with a No Cabinet option. Waves carried

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over a range of cabinet models from GTR 1.0 and added seven new ones—including 1 × 8, 1 × 12, 2 × 10, and 4 × 12 configurations—modeled from cabinets in the private collection of engineer Peter Dennenberg, who owns Acme Studios, just north of New York City.

GTR 2.0's microphone models include a number based on classic mics such as the Shure SM57, Sennheiser MD 409 and MD 421, and EV RE20 in both on- and off-axis positions. You also get a model of the AEA R84 ribbon mic. It's the first time I've seen a ribbon mic model in amp simulation. Ribbon mics are excellent for miking guitar cabinets (I use a Royer R-121 all the time), so it's great to see that Waves modeled one for GTR 2.0.

The cabinet simulations are very convincing. The cabinet models with single or double speakers and any of the on-axis mic models sound the most realistic. But all the combinations I tried sounded good. The closed-back cabinets and ribbon mic simulations sounded close to my own closed-back cabinets miked with a ribbon mic (see **Web Clip 2**).

One of my favorite tricks is to record a direct output from a tube amp and then use software speaker simulations to add the effect of a miked cabinet to the sound. Using Waves Amp, you can accomplish this with the help of the Bypass switch that's next to the Amp Type selector. When you activate Bypass, it effectively takes the amp out of the circuit but leaves the speaker and microphone simulation on.

Stomp on Me

Unlike most amp-modeling software, which includes pedal effects in the same plug-in as the amp emulations, GTR 2.0 puts its effects in a separate plug-in called Waves Stomp (see **Fig. 3**). You can select the number of pedals (two, four, or six) in

PRODUCT SUMMARY

WAVES GTR 2.0

guitar amp- and effects-modeling system
$600

FEATURES	4
EASE OF USE	4
QUALITY OF SOUNDS	4
VALUE	3

RATING PRODUCTS FROM 1 TO 5

PROS: Excellent-sounding DI interface. Tuner plug-in very responsive. Convincing clean to low-gain amp simulations. Great speaker and microphone models. Stompbox effects based on Waves plug-ins.

CONS: No USB or TRS balanced outputs on DI interface. Minimal metering on WPGI interface. High-gain amp emulations compressed-sounding. Distortion effects sound too digital. Pricey.

MANUFACTURER
Waves
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each instantiation, and you can place any pedal in any available slot.

Because of this separation of the amp and effects models, some popular routings are a bit tricky. For example, to put the effects between the amp and speaker models, you first need to instantiate a Waves Amp plug-in minus the speaker simulation. Then in the next insert slot, open up a Waves Stomp, and finally another Waves Amp with the Direct amp simulation selected, along with a speaker and mic simulation. I would have preferred both the amp and stompbox models to be in the same plug-in interface to avoid such an involved setup, but it's not a major issue. At least this sort of routing is possible with Waves GTR, unlike with a few other simulators.

Waves also bucks tradition by offering effects models that—for the most part—aren't based on vintage stompboxes. Many use algorithms from Waves' own digital effects. Others, like Stomp Phaser, are based on composites of several effects pedals. The OverDrive effect is modeled on a modified Ibanez TS8 Tube Screamer.

Waves' effects are uniformly clean, of high quality, and quite useful, and such is the case with those in Waves Stomp. Even the pitch-shifting effect (Pitcher) is great, and pitch-shifting usually sounds mediocre in amp modelers. The place where I missed the vintage pedal simulations the most was with the distortion effects—OverDrive, Distortion, Fuzz, Buzz, and Metal. They all sounded too digital for my taste.

Doing the Wave

The Waves GTR 2.0 package has a lot to offer. For those who need a great-sounding guitar direct box, the WPGI alone may be worth the asking price. If you are looking for a guitar-amp modeling package with excellent speaker simulations to use with a real guitar amp, Waves Amp provides some of the best. And with a few exceptions (like some of the high-gain amp sounds), the amps, speakers, and stompbox simulations are very good.

I would have preferred more output options on the WPGI, and the shortage of emulations of classic analog stompboxes may dissuade some, as might the price. But if you are looking for a guitar recording package and it's in your price range, Waves GTR 2.0 deserves your serious attention.

EM WEB CLIPS

Orren Merton, author and music-technology editor for Thomson Course Technology PTR (www.courseptr.com), likes to simulate being a guitarist.

FIG. 1: The meters, LCD readout, and essential buttons are conveniently arranged on the PCM-D1's front panel.

SONY PCM-D1

Compact, great-sounding portable recorder.

By Rudy Trubitt with Bruce Koball

The Sony PCM-D1 sounds great, is intuitive to operate, and is the coolest-looking piece of gear I've seen in a long time. Although the price is not for the faint of heart, it's the best device I've found for capturing high-quality audio with minimal setup time. Thanks to its built-in mics, fast power-up, and easy operation, I can have the D1 out of its padded carrying case, clipped onto a tripod with levels set, and be recording audio in less than 60 seconds—and that includes setting up the tripod.

The D1 is ideal for recording concerts, recitals, lessons, and rehearsals. In addition to recording music, we found it to be an excellent tool for recording sound effects, voice, and travelogue soundscapes. In short, this recorder is easy to fall in love with.

Shiny, Shiny

The D1 is mechanically impressive. It has a solid, professional feel—the shell is made of pressed titanium attached to an aluminum frame, with little plastic aside from the button tops, LCD lens, and meter faces. The D1's most distinctive visual feature is the top-mounted pair of shiny microphones, which were custom designed for this product by Sony's mic group (see **Fig. 1**). These electret condenser cardioid mics are permanently fixed to the unit in a not-quite-coincident 90-degree XY configuration. The entire mic assembly tilts up and down, allowing some flexibility in the mic direction relative to your viewing angle of the unit's front-panel controls.

Although you can use external microphones, your options are limited by the 3.5 mm minijack mic input, lack of phantom power, and, more surprisingly, lack of plug-in power (a mic input that provides a small voltage, typically 3 to 4V, to power electret condenser mics). The D1 also uses 3.5 mm minijacks for line input and headphone output. A combination 3.5 mm minijack and optical jack provides line and optical-digital output (see **Figs. 2 and 3**). There is no digital audio input to the device.

The D1's self-noise is impressively low. In our measurements, we found the noise floor of the D1's mic

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preamp (not measuring the contribution of the built-in mics) to be from 10 to 30 dB quieter than those of other field recorders we've tested. When using the built-in mics, the D1's self-noise didn't intrude significantly in our recordings, even with the input cranked up. One unexpected source of self-noise did reveal itself—when a gust of wind pegged the analog level meters, there was an audible metallic tinkling of the needles hitting their stops.

Record Ready

Recording with the D1 is a piece of cake. Pressing the record button illuminates its red LED, causing the pause button to blink yellow. Concentric analog record-level knobs sit opposite the headphone-level knob, placing them conveniently under your right thumb or left middle finger. Their operation is solid and silky-smooth. (We offset the left- and right-channel record levels to compensate for a 1.5 dB difference in the line input sensitivity of our review unit.)

Four switches line a recessed groove around the outer edge of the deck. They include power on, mic attenuation (–20 dB), mic/line select, and a hold control. The switches are easy to operate and have a good positive feel, and their location makes it unlikely that you'll activate them inadvertently. The hold button does not affect the record or headphone levels because those are controlled by analog knobs. A highpass filter for the internal mics and a peak limiter are menu-selectable options. Dedicated switches for these would be more convenient.

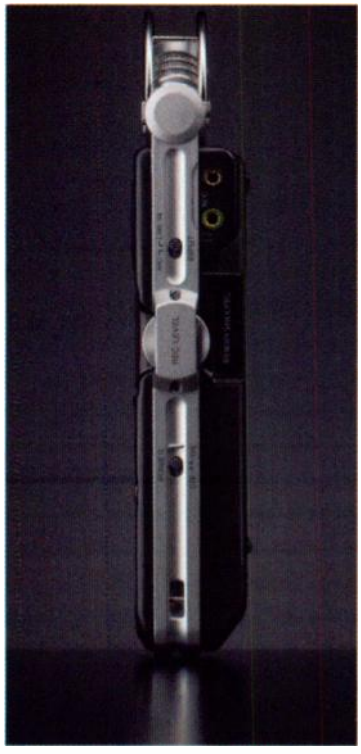

FIG. 2: The right side of the PCM-D1 houses the record-level controls, mic and headphone jacks, and power switch.

The analog level meters are the highlight of the user interface and are visible in virtually any lighting condition. With backlighting engaged, I could read levels in total darkness from a dozen feet away. The meters also include a dual-color LED to indicate signal presence and clipping (–1 dBfs). My only quibble with the metering is that the bottom of the analog-meter scale is –20 dB. When I'm leaving a lot of headroom, which is fairly often, there is relatively little meter movement.

Metering is further enhanced by a large LCD screen, which has left and right bar-graph meters with a wide –60 to 0 dBfs range. Graphical peak-hold indicators and numerical readouts show the available headroom at all times. In short, the D1 provides a comprehensive level-metering solution—better than any recorder I own, portable or otherwise.

The D1 takes a novel approach to peak limiting. Rather than inserting an analog limiting circuit into the signal path before the A/D converter, the D1 splits the analog signal into two paths. The first path is used while the deck is operating below digital clipping. The second path is attenuated 20 dB in the analog domain before being digitized by a second A/D converter. When digital clipping occurs, the second path springs into action, normalizing the –20 dB signal to full scale and substituting its audio for the clipped waveform. Although I appreciate this purist, instant-attack-time approach to peak limiting, in practice we found the D1's limiter operation to be quite audible due to its overly long release, which ranged from just under 200 ms to as much as 400 ms when pushed hard.

File Format

The D1 records 16- or 24-bit audio at sampling rates as high as 96 kHz, writing WAV files to its 4 GB of internal memory. You can add Sony Proprietary Memory Stick Pro modules for additional removable storage. A mini USB connector on the D1 lets you mount it on the desktop of a Mac or PC. We used the unit with both types of computers without installing any special software. As expected, USB 2.0 computers pull audio off the D1 at impressive speeds—roughly 1 GB in 4 minutes on a newer Mac.

The D1 records a new WAV file each time recording is started. You can pause, then continue writing to the same file. Alternatively, you can hit the Divide button to split the currently recording or playing file in two. Among other things, that lets you divide a long recording into the parts of interest and then delete unwanted parts to recover memory.

Each recording is given an 8-digit file name. The first six digits reflect the recording date (YY, MM, DD), and the last two are an ID (0 to 99) that increments for each successive recording made on the same date. For example, the 13th recording made on June 29, 2006, would be named 06062913.WAV.

There are always 10 folders in the D1's internal memory, holding as many as 99 files each. You record into, and play files from, the currently selected folder. You can use an external computer to rename files and move them around. The D1's display shows the first 12 characters of a file name, but you can scroll it to reveal longer, descriptive file names.

The menu system is simple and to the point. A few options might require a trip to the owner's manual. For instance, the LED on/off menu item refers not to the display's backlighting, which has a front-panel button, but to an option that disables the LEDs that illuminate the transport controls. This subtle feature was added to prevent the D1 from calling attention to itself in a darkened theater, especially important

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when it is positioned near the lip of the stage during a performance.

Power

The D1 comes with four Sony NiMH rechargeable AA batteries and a charger. These 2,000 milliamp-per-hour cells can power the D1 for 4 to 5 hours. In a pinch, you can use regular alkaline batteries, which will give an hour or two of operation. An external DC input connector and AC adapter are provided. You can also use an external battery pack. The D1 will accept external voltages between 5.2 and 7.2 VDC, and it displays a warning message if the voltage is outside of that range.

We were quite pleased with the overall battery implementation, which features a removable battery sled to hold the AAs. An extra sled is available as an accessory, although considering the D1's price, I'd prefer that it were included. You don't need batteries when connecting the D1 to your computer because it runs on USB power.

Fun in the Field

The D1 has a bright, detailed sound. Our ears, as well as Sony's published frequency response curve for the internal mics, affirm a broad low-frequency rolloff of roughly –3 dB per octave starting around 700 Hz. The top end is bright, with a bump between 10 and 16 kHz. Sony doesn't spec the mics' ultrasonic response, but we recorded plenty of signal at 40 kHz while dropping tiny steel drill bits onto a concrete floor.

We did a careful A/B comparison of the D1's internal mics with a Røde NT4, another fixed-angle 90-degree XY cardioid. The NT4 has noticeably more low end and slightly better imaging, whereas the D1 adds an attractive sparkle and sheen that the NT4 lacks. In practice we preferred the D1 mics on acoustic guitar and the NT4 mics on a drum kit.

I took the D1 on a trip to southeastern Alaska (see **Fig. 4**). Its relatively small size and quick operation were big advantages. With the unit secured by its handy wrist strap, I captured the splashing of spawning salmon as my four-year-old daughter provided enthusiastic color commentary. Later, in a light rain, with the D1 mounted on a tripod, I was able to capture the call of nesting bald eagles from across a marsh, a distance of about 50 yards. On the downside, due to the mic's diminished low-frequency response, I was not able to capture the low-end oomph from calving glaciers.

Back home my colleague Bruce Koball recorded a jazz quartet consisting of piano, bass, drums, and trumpet. The D1, placed on a tabletop 6 feet in front of the bandstand and pointed a bit away from the drums, captured a pleasing recording of the performance. An EQ pass was needed to restore some lost energy to the acoustic bass.

FIG. 3: USB, digital-out, and line-in connectors, along with the headphone level control, are located on the left side of the PCM-D1.

We also did quite a few sound-effect recordings: passing freight trains, chirping frogs at midnight near a mountain creek, the electric hum of a toaster oven, the gurgle of an espresso machine pump, and so on. The results were uniformly good.

Handling noise can be an issue with the D1, especially at high gain settings. It's also sensitive to wind. The included foam windscreen helps, but it's suitable only for light breeze conditions. Engaging the 200 Hz filter will help in a stiffer breeze, but you sacrifice more low end. I left the windscreen attached most of the time. The mics are plenty bright, so I wasn't worried about whatever tiny effect the foam might have on the top end, and the gray windscreen helps de-bling the attention-getting chrome grille surrounding the mic assembly. In my line of work, the less conspicuous the gear is, the happier I am.

Grant Us These Wishes

We wish the D1 had a prerecord buffer. That would allow you to continuously record into a short buffer and then tack the contents of that buffer onto the beginning of your recording. In effect, you would capture a few seconds before you officially started recording.

We also wish for plug-in or phantom power and more-robust input connectors so that we could use a larger variety of microphones. A digital input would have been another way to expand the D1's front-end options, although dragging around a separate mic-pre A/D is at odds with the operating ethos of the D1. It would also be

PRODUCT SUMMARY

SONY PCM-D1

portable digital recorder
$1,999.95

FEATURES	4
EASE OF USE	5
AUDIO QUALITY	4
VALUE	4

RATING PRODUCTS FROM 1 TO 5

PROS: Bright, clean, and detailed sound. Intuitive, immediate operation. Superb level metering.

CONS: Some handling and wind-noise issues. No digital input, plug-in, or phantom power. No prerecord buffer.

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"The Automap function is what really sets the ReMOTE 25SL apart from other controllers... With Reason, Automap works brilliantly well... Reason is a knob-heavy program that works well enough with a mouse and keyboard, but it flies with a controller such as this."

"Even putting Automap aside, though, the ReMOTE SL is a great controller: it's well made and a pleasure to work with, and those extra-large displays improve the experience of interacting with software a great deal... returning to a smaller display will introduce feelings of claustrophobia."

Derek Johnson - SOS - Jan 06
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ReMOTE SL's AUTOMAP is set to become the saviour of MIDI-based music production, automatically mapping your session's instruments, mixers and effects at the touch of a button or click of a mouse.

Automap supports* -

REASON Cubase Logic Pro 7

*For details on how automap functions with each sequencer, visit www.novationmusic.com/automap

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Superb semi-weighted, high-quality keyboard, velocity- and aftertouch-sensitive

"If the same control set were packaged with a larger keyboard, I think the draw would be **irresistible**"

Derek Johnson - Sound on Sound

Sweetwater knows all about the world of virtual instruments, and they know that their customers demand the best. That's why the ReMOTE SL series has been such a hit at Sweetwater - no other controller offers the combination of tactile control and comprehensive feedback that the ReMOTE SL delivers. To find out how the ReMOTE SL will change the way you work with virtual instruments, call Sweetwater today.

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nice to be able to drop markers in the WAV file during recording and playback to highlight sounds of interest for later reference.

Why We Like It

The D1's self-contained nature is a strength as well as an occasional weakness. Your options are limited when it comes to suppressing wind and handling noise; you can't exactly stuff the D1 into a field-recording-style wind-shield blimp assembly and expect to operate its controls. And in a concert setting, there will surely be times when you'll have to choose between optimal mic placement and access to the recorder's controls.

FIG. 4: The handheld PCM-D1 field recorder en route to Ford's Terror in southeastern Alaska.

But taken on its own terms, this deck excels. The D1 is a great example of what happens when a manufacturer sets out with a clear goal and follows through. We understand that the design team included a number of performing musicians; a priority was clearly given to high-quality audio and simple, convenient operation. These two goals are harmoniously blended. If low cost had been among the design criteria, I doubt the end result would have been as satisfying. The price of this unit puts it out of reach for some, but those who take the plunge will be glad they did.

Rudy Trubitt and Bruce Koball record music and sound in the San Francisco Bay Area. Reach them at rudy@trubitt.com and bkoball@motionwest.com.

FIG. 1: The Telefunken USA M16 Mk II has a vintage look and is ruggedly built.

TELEFUNKEN USA R-F-T M16 Mk II and AK 47

Vintage quality at a modern price.

By Myles Boisen

Telefunken USA's first success was a historically accurate replica of one of today's priciest vintage microphones, the Telefunken Ela M 251. Although the company has since cloned other vintage mics—ranging in price from $4,000 to $16,000—it has also created the R-F-T line for those of us who cannot afford a vintage prize or a spare-no-expense replica. The first two mics in the line are the M16 Mk II and the AK 47.

Past, Present, Future

The M16 Mk II and AK 47 are large-diaphragm tube mics with vintage visual appeal (see **Figs. 1 and 2**). The M16 Mk II is sonically styled after the AKG C 12 and Neumann U 67, and the AK 47 is designed with the Neumann U 47 in mind. Both Telefunken USA mics share the same thin, cylindrical shape—similar to a C 12's—with a powder-coat finish and chromed accents on both ends. The M16 Mk II is light gray, while the AK 47 is a darker charcoal gray. Antiqued Telefunken logo badges adorn the front-address side of each mic, and an engraved model and serial number appear on the back of the bodies.

The hardwood storage case that comes with each mic is one of the handsomest I have seen. With its sturdy latches and round edges, this box could easily be mistaken for an original case brought out of storage after 50 years. Each mic comes with a shockmount, a 25-foot multipin mic cable, which connects the mic and the power supply, and a limited five-year warranty.

Nine pickup patterns—from omnidirectional to figure-8—are available from a switch on the power supply. This arrangement allows the pattern selector to be located in a control room and adjusted remotely by the engineer. There are no attenuation or filter switches on the mics or the power supply.

Full Metal Jacket

Telefunken USA's quality-oriented approach to manufacturing is evident inside and outside these mics' hefty all-metal bodies. In the tradition of classic European transducers, the parts are heavy-duty and precisely machined, and fit tightly with no visible screws. Inside the M16 Mk II, a new old stock (NOS) GE 6072a JAN tube—the same model used in Telefunken's Ela M mics—is held securely in place, shockmounted with rubber collars. The AK 47 is outfitted with an NOS Telefunken EF732 subminiature tube. Upon initial inspection, it's hard to see how Telefunken USA has cut corners to make a mic that lists for less than half of its other models.

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couldn't match the other microphones' low noise specs.

At the same preamp gain settings, the M16 Mk II and AK 47 in cardioid mode exhibited a noise floor 5 to 6 dB higher than the other test mics. In omni mode, the M16 Mk II had a noise floor about 3 dB higher than in cardioid. I'm sure these specs are well below the noise of some vintage and lower-cost tube mics, and I wouldn't expect their self-noise (20 dB A weighted) to be a problem in most circumstances.

On Voice

Singer-songwriter Kim Boekbinder from the band Vermillion Lies was a willing subject for two afternoons of Telefunken testing. Boekbinder sang and played guitar in the live room at Guerrilla Recording, about three feet in front of the test pairs. Grace 101 preamps were used for this test, along with Blue Kiwi and Monster cables, and the results were recorded into Pro Tools.

In cardioid and figure-8 modes, there were minor differences within each mic pair in terms of timbre and self-noise, as is typical with tube mics. Frequency traces for each of the M16 Mk IIs showed minor differences of 1 to 2 dB between 900 Hz and 4 kHz. Nonetheless, the two microphones were matched closely enough throughout the audible range to reliably demonstrate the M16 Mk II's character. The AK 47 models did not come with individual frequency traces.

In omnidirectional mode, the M16 Mk IIs were more closely matched. While I do occasionally use omni pickup for close-miking acoustic guitar, at this distance the sound was too roomy and diffuse for practical recording purposes. Comparatively, the AK 47 worked better for me in omni, although it was still less than ideal. The cardioid and figure-8 patterns of both models gave clarity and focus to my performer, and balanced the guitar and voice nicely. The robust output level of these mics made it possible to record fingerpicked acoustic guitar and quiet vocals without undue noise.

When it was compared with the M 147, the AK 47 sounded smoother and less aggressive in the upper mids, with a little less low-bass clarity and fullness. On vocals and guitar, the AK 47 also sounded more hi-fi due to an extended high-end response, and it benefited from the intimacy of a tighter pickup pattern.

The M16 Mk II was comparable to the M 147 in terms of pickup pattern and output gain. The M 147 conveyed less brightness and yielded a more focused and prominent bass on the guitar, with the mix favoring the guitar over the vocal. But on solo acoustic guitar, recorded at a distance of 18 inches, I preferred the brighter tonality of the M16 Mk II over the M 147.

The AK 47 was remarkably similar to the Lawson L47 MP tube mic, another Neumann U 47 emulation and a personal favorite of mine for the last ten years. When comparing them, I was really surprised by the equally robust, mellow tube quality of both mics. In addition, the AK 47 scored points with me by delivering more incisive highs than the Lawson on quieter material, while never getting sibilant or harsh.

Predictably, a comparison of the AK 47 to the M16 Mk II showed the former to be superior in low-end and midrange pickup, as well as smoother and less sibilant. In cardioid mode, the AK 47 also had a tighter pattern than the M16 Mk II, sounding closer, bigger, and less roomy.

In the Field: M16 Mk II

During the review, I was fortunate to be able to audition the M16 Mk II on a variety of vocalists in a range of musical styles. Jazz vocalist Pyeng Threadgill sounded good through the M16 Mk II, and the upper ranges of her voice were definitely highlighted. Threadgill told me that she felt the mic may have emphasized her airy highs and breath sounds too much. Being very familiar with her voice from a previous album project, I didn't agree with that assessment. But I did feel that the Telefunken USA lacked smoothness in the crucial 3 to 5 kHz range, compared with other tube mics I have used on her.

For a project by Bay Area band the Unravellers, I used the M16 Mk II on two background singers. The M16 Mk II was predictably bright and airy on the female singer, allowing her to cut through a dense mix with ease. On the male singer, the mic still had plenty of edge, but sounded hollow or thin when the vocalist dipped into his lower range.

On a jazz tenor saxophonist with a bright, contemporary sound, the M16 Mk II delivered an even, pleasing tone with plenty of high-end gloss. As expected, there was not an abundance of low-end tone, and occasionally the sound was grittier than what I heard in the room.

In the Field: AK 47

The AK 47 also got a thorough workout on an assortment of singers. In general this model was much closer to my ideal of how a tube mic should sound on vocals, with lots of rich tonal character and just the right amount of high-end cutting power. The mic's timbre was perfectly balanced on a pair of female singers, capturing rich resonance without getting overly bright. On a pair of male singers the AK 47 also got high marks, although at times it was just a little too crisp in the 4 to 5 kHz range. But sibilance was never a problem.

This model seems particularly sensitive to placement when close-miking vocals. Often I found that moving the

PRODUCT SUMMARY

TELEFUNKEN USA R-F-T AK 47

tube microphone
$1,599

AUDIO QUALITY	4
VALUE	3

RATING PRODUCTS FROM 1 TO 5

PROS: Affordable. Vintage quality in looks and build. Nine remotely selectable pickup patterns. Wooden case. Five-year warranty.

CONS: Shockmount doesn't latch.

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singer 1 inch closer or farther made a significant difference on high-end clarity or proximity-induced lows. With singer Scott Rosenberg of the band P.A.F., I set the pattern selector one notch to the left (subcardioid), yielding a more balanced sound with less proximity effect. On Rosenberg's whisper-to-a-scream dynamic range, the AK 47 also delivered smooth tube compression, and even a bit of classic overdrive distortion.

In addition to vocal applications, a good test of a tube mic is how well it works with wind instruments. In this regard I was very impressed with the AK 47's ability to deliver both tone and clarity. I used it on B-flat and bass clarinet, flute, oboe, English horn, and tenor and alto sax. It performed admirably on every track and often reminded me of the smooth tonality of my Lawson tube mic.

Engineer Bart Thurber, who shares the Guerrilla Recording facility with me, was enthusiastic about the sound of the AK 47 on his rock projects. He cited its ideal coloration on electric guitar and its ability to handle high SPLs when placed up against a guitar cabinet.

He also used a spaced pair of AK 47s (in subcardioid mode) in a variation of a 3-mic drum recording technique. Listening to these tracks, we were both favorably impressed by the big, spacious sound the mics provided. The cymbals were crisp yet not overbearing, and the toms, which were not close-miked, had an excellent blend of tone and presence in the mix.

Two Distinct Personalities

The M16 Mk II tube condenser mic has a strong personality, not to mention impressive vintage looks and build quality. Around the studio it delivered a useful presence enhancement on a variety of tracks and never sounded muddy or undefined. However, the mic possessed little of the harmonic richness or smooth, silky highs that top-dollar tube mics are known for.

Of course, strong timbral coloration on a vocal or instrumental source guarantees success in some settings and a mic change in others. Recordists favoring a bright pop timbre—particularly on female vocals and lead instruments—may take a shine to the M16 Mk II due to its glossy highs and gradual bass rolloff. The tonal variation provided by the multiple patterns adds value and versatility to this affordable tube mic.

For general use I prefer a large-diaphragm condenser with more midrange and low-end warmth, and that's just what the AK 47 delivers. I was consistently impressed by this mic's rich tone, graced by an extended high end that kept it from being too warm or undefined.

Despite minor problems with the shockmount and self-noise, in the ever-expanding realm of tube mics under $2,000, the AK 47 is a worthy contender.

Myles Boisen picks a peck of pickup patterns at his Guerrilla Recording studio in Oakland, California.

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FIG. 1: In Hybrid mode, Arturia Prophet-V combines the features of a Prophet-5 (bottom) and a Prophet-VS (top). The joystick (upper left) can be assigned to two MIDI Control Change messages, so you can sweep through the oscillator mix in real time.

ARTURIA Prophet-V 1.0 (Mac/Win)

It's more than just a model of the Prophet-5.

By Jim Aikin

The Sequential Circuits Prophet-5, which debuted in 1978, was the world's first fully programmable polyphonic synthesizer. Although the Prophet-5's total sales figures never approached those of the popular synths of the 1980s, it has a mythic status due to both its warm analog sound and its unique position in history. The elegant wooden end panels add a coolness factor as well. Unfortunately, real Prophet-5s are hard to find and even harder to keep in good repair.

Given Arturia's quest to model the most popular vintage analog synths, it is not surprising that the company has created Prophet-V 1.0. However, Arturia went one step beyond merely mimicking a Prophet-5 by including a clone of the Prophet-VS, an instrument that used vector envelopes to crossfade between four oscillators playing single-cycle digital waveforms. To top it all off, Prophet-V offers a Hybrid mode, which lets you create patches that combine the voicing features of both instruments (see Fig. 1). All three modes share certain features, including 32-note polyphony, external MIDI control of most parameters, and two effects processors (a chorus and a stereo delay).

Prophet-V can operate as a standalone instrument or as a VST, AU, DXi, or RTAS plug-in. I used it in standalone mode and as a VSTi on my main Windows computer, a 3 GHz Pentium 4. A Syncrosoft USB dongle is used for copy protection. I compared the appropriate sound and features of Prophet-V with those of a real Prophet-5 owned by Rick Eberly, a former Sequential Circuits employee who currently provides repair services through Wine Country Productions in San Jose, California.

Take-5

The Prophet-5 emulation in Prophet-V offers what you would expect: two oscillators (A and B), a resonant 24-dB-per-octave lowpass filter, two 4-stage (ADSR) envelopes, an LFO, a mixer, a VCA, a noise module, and a polyphonic modulation (poly mod) section, which helped define the sound of the original instrument (see the sidebar "The Real Thing"). Rather than simply

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replicating the Prophet-5, Prophet-V adds a number of welcome enhancements, such as antialiasing oscillators, an effects processor, and a monophonic Legato mode.

Arturia has been careful to keep the look and feel of the original Prophet-5 while doing its best to reproduce the synth's signature sound. For example, Prophet-V's filters sound rich and organic when being modulated by Oscillator B. I spent some time duplicating a modulated filter patch created on the Prophet-5, and the Arturia version was able to nail the sound.

At first glance, Prophet-V is almost too faithful to the original. There's no Velocity response, for instance, unless you switch to Hybrid mode and use the Velocity modulation routings on the VS side. The original set of 40 factory patches is included, and they are accessed in 5 banks using a set of 8 buttons, as on the original. (The remainder of the 400 patches included in Prophet-V are accessed from a pull-down menu rather than through the panel buttons.)

Next to the Unison button is a knob that controls the amount of detuning, which is useful because a digital instrument doesn't exhibit the random pitch fluctuations from voice to voice that an analog instrument does. This knob is active even when Unison mode isn't. Turning it up causes the exact intonation of a given interval to change due to voice reallocation, just as it did on the Prophet-5.

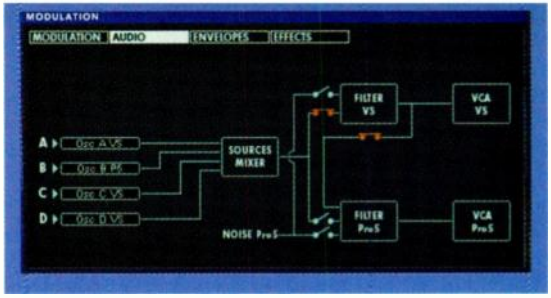

FIG. 2: In Hybrid mode, Prophet-V's four oscillators (left) can mix Prophet-5 analog models with Prophet-VS digital wavetables. Active signal connections are shown in orange; here, the two filters are in series.

Prophet-V has an "analog tone" modulation source that causes rapid, subtle fluctuations in waveform pitch and phase. The depth of this modulation ought to be user programmable, but it isn't. In the low register, the tone of Prophet-V sounds significantly less stable than that of the Prophet-5 I compared it with. I didn't care for the sound quality of this tone; it made me nervous. Possibly the instrument Arturia used as the basis for its model wasn't in good condition.

Prophet-V's LFO has a Sync button, which locks its rate to an external MIDI clock. In sync mode the LFO has only 16 possible rates: from 8 beats through 1 beat, and from ½ through ⅛ beat. Certain useful values seen almost universally in syncable LFOs, such as dotted

THE REAL THING

Sequential Circuits (the name was later changed to Sequential) built a line of Prophets, including the original Prophet-5, the Prophet-10, the Prophet-T8, the Prophet-600, the Prophet-VS, the Prophet-2000, and the Pro-One monophonic synth. None of the later models achieved the popularity or legendary status of the Prophet-5.

The Prophet-5 was capable of 5-note polyphony and had a bank of 40 memory locations for storing presets. (Offline storage was handled using an analog-tape cassette interface.) The actual synthesizer was analog throughout, but digital circuits were used both to scan the keyboard and to store the settings of the knobs and buttons. There was only one audio output.

The Prophet-5 voice had two oscillators and a noise source, a resonant lowpass filter, a low-frequency oscillator, and two ADSR envelope generators. The release segments of the latter could be switched on or off with a Release button, a feature borrowed from the Minimoog. A Unison button created a big monophonic sound by assigning all five voices to a single key. Because the oscillators were analog, there were always small discrepancies among them in pitch, so Unison mode provided a fat, chorused sound.

Oscillator A had two waveform choices, a square wave and a sawtooth. Both could be active at once. A Pulse Width knob controlled the square wave's shape, and a Sync button could be used to sync the frequency of Oscillator A to Oscillator B. Both A and B had frequency knobs for tuning in half steps.

Oscillator B had three waveform buttons (sawtooth, square, and triangle), selectable in any combination. A fine-tuning knob allowed Oscillator B to be detuned from A. Next to Oscillator B's Pulse Width knob were two more buttons. One switched Oscillator B into low-frequency mode, allowing it to be used as a second LFO. The other enabled or disabled keyboard tracking. When tracking was switched off, Oscillator B produced a tone at a constant frequency.

The LFO and noise source could be used for modulation in the Mod Wheel section, which had five destination buttons (the frequencies and pulse widths of the two oscillators, and the filter cutoff). A Source Mix knob in this section allowed the LFO signal to be crossfaded with the noise source. The depth of modulation was set directly from the mod wheel. Aside from the mod wheel, there was no way to add vibrato.

The Poly Mod section had two Source Amount knobs, one for the filter envelope and one for Oscillator B. Three destination buttons were provided, for the frequency and pulse width of Oscillator A and the filter cutoff. The Poly Mod section was responsible for some of the Prophet's most evocative sounds. It could be used to control Oscillator A's pitch or pulse width from the filter envelope, and also to drive the filter with an audio-rate signal from Oscillator B.

The Prophet's other features included a glide-rate knob, a button that produced a reliable A-440 for tuning purposes, and a handy tune button that caused the instrument to tune itself. Tuning required several seconds, during which the Prophet produced no sound. Typically, tuning was required several times while the instrument warmed up and about once an hour afterward.

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quarter notes and dotted eighths, are not available—a glaring oversight.

Every control on the Prophet-V panel can be linked to an external MIDI Control Change message. When I attached a MIDI slider to the filter cutoff knob using the Learn function, I was pleased to hear how smoothly the filter responded—my slider sounded like a component on a real analog synth. Another nice touch is that the position of the onscreen mod wheel is stored with the preset. This allows sounds to have built-in vibrato, which was not possible on the Prophet-5.

A Vector in the Mix

The Prophet-VS introduced the concept of two-dimensional vector envelopes, which after the demise of Sequential Circuits were adopted in the Yamaha SY22 and Korg Wavestation. The VS had four oscillators that played single-cycle digital waveforms, and the vector envelope was used for crossfading among the oscillators.

While the voice architecture of Arturia's Prophet-VS is much the same as that of the original, this side of the software is considerably less authentic than the Prophet-5 side. The front panel is quite different. The VS's arpeggiator (described by Julian Colbeck in *Keyfax Omnibus Edition* as "the bee's knees") is missing, and the filter response is multimode rather than strictly lowpass. There is no facility for loading your own waveforms into slots 0 through 31 of the wavetable; only the 96 factory waves are available.

I couldn't locate a Prophet-VS for comparison, but I vividly remember that its digital wavetables aliased quite markedly in the upper register. Fortunately, Arturia has eliminated this rather horrible sound.

The Prophet-VS has two LFOs, three envelopes (two are 5-stage envelopes, and one is a 4-point envelope for the VS Mixer), and an unusual panning mode that allows individual voice channels to be given static panning positions for a big stereo spread. The Arturia VS can produce some sweet "vintage digital" sounds, especially when envelope looping is applied.

Common to both the Prophet-5 replica and the VS replica is a choice of High, Low, Last Note, Reset, and Circular voice assignment modes. The first three produce monophonic results. In Reset mode the lowest-numbered available voice is used, while in Circular mode the synth cycles through its available voices. If your patch has a long envelope release, Reset mode will steal notes that may still be sounding if the key has been lifted, which may not be what you want.

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Unfortunately, Circular mode produces nasty crackling distortion, so it won't be usable until Arturia fixes the bug.

The modulation matrix allows 8 sources to be patched to 15 destinations, but with some limitations. For instance, the keyboard can control oscillator mix, filter cutoff, or panning, but not LFO rates and amounts. Each source has only one amount parameter, which is applied to any destinations that are switched on for that source. This system is a bit primitive but usable. The maximum depth available for any modulation source is 1.00. The units are arbitrary, and may be different for different sources, but in general I felt that more modulation depth should have been allowed. The maximum amount of Velocity to amplitude, for instance, just isn't dynamic enough to suit me.

Hybrid Mode

In Hybrid mode, the features of the Prophet-5 and the VS are integrated. Four oscillators in all are available; the first two can be from either the Prophet-5 or the VS. Oscillator mixing is handled by the VS joystick and vector envelope. The two filters can operate in parallel or in series. What's unusual about the series routing (see **Fig. 2**) is that both VCAs are still active. This lets you use the two amplitude envelopes to crossfade between the sound of the VS filter by itself and the series output.

By using Hybrid mode but ignoring the VS oscillators, you can create Prophet-type patches with extra LFOs, envelopes, and so on. I like having this kind of flexibility.

PRODUCT SUMMARY

ARTURIA Prophet-V 1.0

software synthesizer
$249

FEATURES	4
EASE OF USE	4
QUALITY OF SOUNDS	4
VALUE	4

RATING PRODUCTS FROM 1 TO 5

PROS: Authentic sound. Hybrid mode adds many features. Smooth response to MIDI Control Change messages.

CONS: Prophet-VS design is not fully authentic.

MANUFACTURER
Arturia
www.arturia.com

What Shall It Prophet?

Overall, Arturia Prophet-V replicates the sound and feel of the Prophet-5 very well, while adding a number of features that modern synth users will want. With the inclusion of the VS model and Hybrid mode, Prophet-V offers plenty of bang for your buck.

Jim Aikin is a regular contributor to EM and other music-technology magazines, a cellist, and an author. You can visit him online at www.musicwords.net.

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UNIVERSAL AUDIO

Solo/610

By Eli Crews

In the November 2006 issue (available online at www.emusician.com), I reviewed Universal Audio's Solo/110. This time around I got a crack at its tube cousin, the Solo/610 ($799). Unlike its solid-state counterpart, this portable microphone preamp and DI box is based on a vintage design by the father of the company's present owners, the legendary Bill Putnam. The 610 console was one of the first recording consoles ever made and was renowned for its supreme sonic quality. For the past few years, Universal Audio has been selling oodles of the various reissue packages of the 610, and the Solo/610 comes in as the most affordable device yet in this exquisite heritage.

Universal Audio's Solo/610 allows you to get a piece of the 610 legacy at a much lower price point.

All in the Family

The similarities between the Solo/610 and the Solo/110 are many; for starters, the housing is equally portable. The 610 also has a DI input with a Thru jack, a ground-lift switch, and a switch for determining whether the XLR output jack is at mic or line level, making it easy to use as a direct interface. On the front face, the two oversize knobs for gain and level retain the ridged, retro look of the original 610 console. Small flip switches are used to activate phantom power, polarity reversal, and a 100 Hz highpass filter. There are also switches for mic or DI input and low- or high-input impedance (500Ω or 2 kΩ for mic, 47 kΩ or 2.2 MΩ for DI). Functionally, the only features lacking when compared with the rackmounted 610 variations are high- and low-shelving EQ (having been replaced by a low-cut filter), a line input, and a stepped (as opposed to variable) gain control.

Guitar

My initial comparison tests were made using the DI input. I often track guitars direct, both for reamping later as well as for blending some of the direct sound with the amp sound to achieve more high end than you can really get out of a guitar amp alone. Plugging my Fender Mustang into the Solo/610 immediately made me smile. Not that it was a totally usable tone on its own, as direct guitar rarely is, but it was pretty close. Checking it against the 610 half of my Universal Audio 6176, the Solo/610 had a very similar sound. I favored the 6176, which had a little smoother low midrange, but only slightly. Compared with the Solo/110, I preferred the depth of the Solo/610 a bit, and there was a slight sense of compression that made the guitar sit better when strummed or picked hard.

Bass

I was equally pleased running electric bass into the DI input. In this application the Solo/610 was almost identical to the 6176. Compared with the Solo/110, however, the Solo/610 had significantly better low end. There was also a larger difference between the low- and high-impedance settings, the low setting giving a fat, round sound and the high setting imparting more of a modern, bright tone. Both were quite usable, depending on preference. One interesting note is that I could never get the Solo/610's tricolored signal-level LED to go into the orange by plugging any of my basses or guitars directly in. Granted, I have older instruments with passive pickups, and a simple boost box let me access the nice harmonic distortion available when the gain of the Solo/610 is pushed. The Solo/110, on the other hand, saturates quite a bit easier, which makes sense given it specs out 20 dB hotter for instrument input gain.

Workhorse

As for the mic preamp, I have run quite a few instrument and mic combinations through the Solo/610 over the past few months, and it has handled them all beautifully. From French horn to rock vocals, from U 87s to SM57s, it has yet to disappoint. The open, airy, clear-yet-warm sonic signature of the 610 preamp is definitely apparent here. If I had to name a favorite use for this box, I would say it's as a preamp for electric guitar. Whether I put up a Royer R-121, Sennheiser 421, or any other commonly used guitar mic, I was always very happy with the way the Solo/610 made the guitar sound: full, thick, and present.

I Like It

All in all, I was very impressed with the performance of the Solo/610. I expected it to be a watered-down version of the 610 preamps I have come to love, but apart from the lack of EQ, I felt that it really held its own next to its more expensive counterparts. Not to say it's the cheapest preamp/DI you can get, but for the quality in this box, you'd be hard-pressed to find a better value.

Value (1 through 5): 4
Universal Audio
www.uaudio.com

Z.VEX EFFECTS

Ringtone

By Myles Boisen

In the ever-expanding universe of way-out boutique pedals, Zachary Vex is a tireless innovator. One of his latest offerings is the Ringtone ($499), a ring-modulation pedal that offers something new in the realm of guitar stompboxes. Like all Z.Vex pedals, the Ringtone has high-quality components, an attractive hand-painted case, and extended battery life.

Season's Greetings

Wishing You All the Best for a Joyous Holiday Season

Gary Garritan

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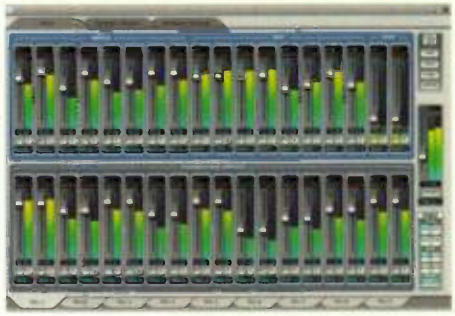

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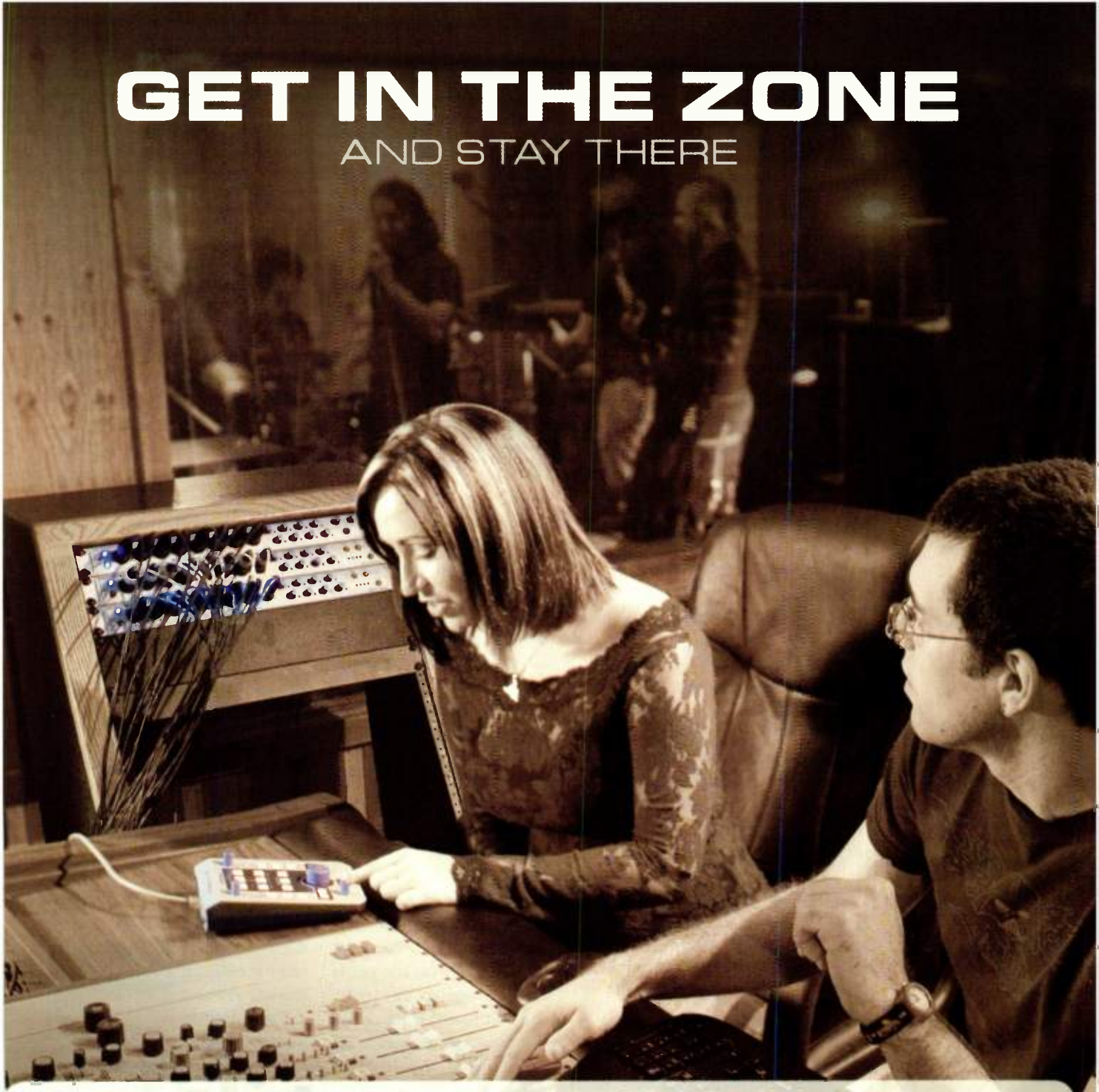

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Sequential Circuits

At the heart of the Ringtone is a sequencer with eight steps of independently tunable ring modulation. Each step of the sequence is indicated across the top of the stompbox by a red LED and is accompanied by a small black knob about the size of a pencil eraser. These eight knobs control the carrier pitch of a ring-modulator circuit.

For those not familiar with ring modulation, the effect accepts two signals and gives you only the sum and difference tones at the output, with none of the original signal present. For example, if the input signals are 800 Hz and 200 Hz, you would hear a 1,000 Hz tone (sum) and 600 Hz tone (difference) combined at the output.

With the Ringtone, an internal carrier oscillator—with a frequency range of 2 Hz to 1 kHz—is used to modulate the input signal, to create a harmonically complex sound. The ring-modulated output can range from a slow tremolo (when the carrier frequency's tuning control is turned fully clockwise to get a subsonic pitch) to a thick, and often dissonant, array of harmonics (when the tuning control is turned fully counterclockwise). At around the 11 o'clock position, where the carrier pitch becomes audible, the tremolo effect becomes fast enough to create modulation sidebands.

Step-by-Step

The Ringtone also includes a true bypass switch, the Random/Step switch, and the Run/Step mode switch. The sequencer speed is controlled by a knob marked SPD to the left of the eight tuning controls. The speed ranges from approximately three to ten steps per second. At faster speeds, and with higher carrier-pitch settings, the Ringtone can generate a sequence that sounds just like a cell phone ringtone.

When you are in Run mode—the R/S control is set to R—the left footswitch toggles between linear and "random" sequencing. In manual Step mode—with the R/S control in the S position—the left footswitch is used to cycle through the eight ring-modulation settings. Step mode greatly expands the versatility of the Ringtone, because it gives you eight ring-modulation "presets." The online tutorial shows how you can tune each step to complement a particular chord in a song, but this setup suggests many other possibilities.

The Ringtone includes an internal trim pot that adjusts the balance between the input and processed signals. When the trim pot is fully counterclockwise, you hear only the unprocessed input signal. When the trim pot is set at the opposite extreme—fully clockwise, as set by the manufacturer—you hear only the processed, ring-modulated signal and none of the original signal. Whether bypassed or with the trim pot set to pass the input unprocessed, the signal quality of my custom Strat-Tele hybrid, which has single-coil and humbucking pickups, was clean and uncolored.

Ring-a-ding-ding

When used with an electric guitar, the Ringtone worked best with single notes, double stops, and triads. Dense chords created a thicker sound but also yielded a multitude of harmonics that tended to obscure the character of the chords. (In this situation, you may want to use the internal trim pot to reduce the ring-modulation effect by adding unprocessed guitar into the mix.) Adding a distortion box to the signal chain, either before or after the Ringtone, intensifies the sum and difference tones generated by the ring-modulator circuit. The pedal also delivers a credible tremolo effect at its lower carrier-frequency settings.

The audible carrier pitch, though necessary for the functioning of the circuit, can be a distraction. On the other hand, it is fun to set the Ringtone to play a self-generating pitch sequence on its own, although the output level is low and a bit noisy when the effect is used in this manner.

I had a blast running mixes through the Ringtone, matching tempos on groove pieces, warping sounds, and providing subtle new textures to drones and other ambient material. Besides being a unique guitar effect, the Ringtone has great potential for creative processing with bass, keyboards, and other electronic sources, including individual tracks in the studio.

Value (1 through 5): 4
Z.Vex Effects
www.zvex.com

The Z.Vex Ringtone lets you sequence through eight steps of ring modulation.

GRAPH TECH

Ghost Modular Pickup System

By Orren Merton

EM ★★★★★ hot pick

When you play a note or adjust a knob on a MIDI keyboard, it transmits MIDI data. A guitar pickup, however, produces analog electrical current that must be converted to MIDI data before it can control MIDI devices. To accomplish that conversion, a *hexaphonic* (literally meaning "6-voice") pickup sends 6 discrete electrical signals (one for each string) through a 13-pin connection to an external converter unit. The converter continuously analyzes the signals' variations in frequency and translates them into MIDI data. Graph Tech has entered the MIDI guitar arena by introducing the Ghost Modular Pickup System, which features the Hexpander MIDI-ready preamp.

If your guitar needs a MIDI retrofit, the Graph Tech Ghost Modular Pickup System features the Ghost piezo saddle pickup and the Hexpander MIDI-ready preamp module.

Ghost in the Machine

The system's most essential components are a piezo saddle pickup and a preamp module. Graph Tech manufactures a variety of saddle pickups (from $119 to $275) for Strat- and Tele-style, Tune-o-matic, and other guitar bridges, as well as Precision- and Jazz-style, Hipshot, and other bass bridges. Separate piezos in the saddle send an electrical signal from each individual string to the preamp.

You can choose from two preamp modules, the Hexpander and the Acousti-Phonic. Each is a circuit board installed in your guitar. The Hexpander preamp ($250) lets you use the Ghost system as a MIDI controller. The Hexpander routes the hexaphonic signal through a 13-pin output to a third-party MIDI converter (not included). The Hexpander comprises the preamp module, requisite wiring modules, and a 13-pin jack, and must be installed inside your guitar. The system's modularity lets you choose from many optional features such as a volume pot ($26), a program-select switch ($33), a 7-pin wiring harness to route your guitar's magnetic pickup's signal through the 13-pin output ($13), and a switch to toggle between the magnetic pickups, Hexpander system, or both ($26).

The Acousti-Phonic preamp ($100) promises to make your electric guitar sound like an acoustic, but I didn't request one to evaluate. Both preamps can be installed in your guitar at the same time.

Ghost Tracking

The main complaint of musicians who use guitars as MIDI controllers is their typically poor tracking. Hexaphonic systems are notorious for tracking inaccurately, too slowly, or both. Phantom notes (triggered notes you didn't play) and double triggers (two notes when you played only one) are usually caused when a hexaphonic pickup misinterprets finger noise.

The Ghost system tackles those problems with a proprietary harmonic-dampening technology that squelches finger noise by filtering out specific frequencies for each string. Harmonic dampening reduces false triggers without sacrificing accurate tracking and results in a very clean signal. The Hexpander also incorporates a low-latency preamp design that offers far faster tracking than magnetic hexaphonic systems.

Ghost Rider

Anyone who repairs electric guitars can install the Ghost system in your existing guitar. Graph Tech's Web site has a list of qualified technicians who can perform the retrofit. I've had years of experience playing guitars with magnetic and piezo hexaphonic retrofits, and two months ago I had the Ghost Hexpander system installed in my custom-built Koll Guitars Tornado.

I used the Hexpander connected to a Roland GI-20 MIDI interface to control several software synthesizers (see **Web Clip 1**). The Hexpander tracked as fast as or faster than any other system I've tried. When I played rapid-fire runs, every note came through. But accuracy is more important than speed, and that's where the system really shines. I usually need to erase lots of errors when I record sequences with a MIDI guitar controller, but with the Ghost system, I've definitely noticed fewer false triggers. My only quibble is that I wish the system had an optional LED to let you know when it's working.

I also had a chance to try the system with an Axon AX100mkII interface. The Hexpander's circuit board has a tiny Traktion switch that optimizes the tracking curve for either Roland- or Axon-type converters. With a flip of the switch, I got the same excellent results using the Axon.

If you're considering retrofitting your guitar as a MIDI controller, Graph Tech's Ghost Modular Pickup System should be at the top of your list. Its modular nature allows you to tailor the components you choose to exactly what you want. Most important, it tracks with minimal latency and impressive accuracy. A guitar with a Ghost system installed is truly a capable MIDI controller. EM WEB CLIPS

Value (1 through 5): 5
Graph Tech
www.graphtech.com

SOUNDTECH

LightSnake Guitar/Keyboard USB Cable

By Orren Merton

Recording a direct electric guitar signal using your computer isn't always as direct as you might expect. Because guitar pickups produce a high-impedance output, you'll usually need a preamp to boost their signal to line level. You'll also need an audio interface to deliver that signal into your computer. Guitarists, especially those who prefer mobile systems, want simple alternatives that sound good.

Some manufacturers have tried to simplify direct recording by producing guitars with USB ports. USB guitars have onboard analog-to-digital (A/D) converters and send digital audio streams directly to your computer. For the vast majority of guitarists who don't play USB guitars, SoundTech makes the LightSnake Guitar/Keyboard USB Cable ($69.99). The LightSnake lets you plug your guitar (or any other instrument with a mono ¼-inch output) directly into your USB-equipped computer, with no additional hardware needed.

On one end of the LightSnake is a ¼-inch TS plug, and on the other end is a USB plug; both are molded into the cable. Inside the USB plug's housing is a 16-bit, 48 kHz A/D converter that's also 44.1 kHz compatible. To ensure a strong signal from your guitar pickups, the high-impedance (1 MΩ) analog circuitry boosts the signal by 20 dB, bringing it up to line level before conversion. The plugs at either end are fitted with green LEDs that flash when the cable is connected, indicating

that the cable is operating properly.

Because the LightSnake is a plug-and-play, class-compliant USB audio device, it doesn't require any drivers when used with Mac OS X or with Windows 98SE or later. The cable ships with ASIO drivers for older versions of Windows that don't offer class-compliant support. It also ships with a CD-ROM containing demo versions of Sony software for Windows, including Acid, Sound Forge, Vegas, and others; no Macintosh demos are included.

Light Speed

No matter what audio interface you use, latency can be a problem when you monitor your guitar through software. To circumvent that issue, SoundTech allows you to directly monitor your signal before it is digitized. The first-generation LightSnake cable (the kind I received at first) ships with a separate splitter that has a ¼-inch plug and two ¼-inch jacks. You insert the splitter's plug into your guitar, the LightSnake's ¼-inch plug into one of the splitter's jacks, and a connection to your guitar amplifier into its other jack. The second-generation LightSnake cable doesn't include a separate splitter; instead, the splitter is integrated into the cable's ¼-inch connector.

I applaud SoundTech for offering a direct monitoring option to get around the latency problem, but monitoring directly from the dry guitar isn't practical if the sound you want to hear is being processed by software. Luckily, I found that by adjusting the buffer in my audio software to 64 samples, I could achieve a latency low enough to avoid distraction. Higher settings resulted in too much latency, and lower settings compromised the signal with crackles and pops.

Some software doesn't allow you to specify separate input and output devices, which might mean that you couldn't use the LightSnake as your computer's audio input without losing its audio output. However, Mac OS X users can create an Aggregate Device in Audio MIDI Setup, assigning the LightSnake as the input and another device (including built-in audio) as the output. Windows XP's Control Panel lets you independently choose an input and output for class-compliant USB devices.

The SoundTech LightSnake lets you connect your guitar, bass, or keyboard directly to a USB-enabled computer, without the need for a separate audio interface.

Light and Sound

My initial experience with a first-generation LightSnake cable was abysmal: I heard more noise than signal. I figured something must have been wrong, especially because SoundTech says that a Host Side Data Loss (HSDL) noise-reduction system was built into the cable. After some investigation, the company discovered that a flaw in manufacturing had affected some of its first-generation cables. SoundTech sent me a second-generation LightSnake, which was far better. The signal was strong and clean, and the HSDL noise reduction was definitely effective. Both of my electric guitars sounded quieter through the LightSnake than they would have through a tube amplifier.

The LightSnake cable won't appeal to guitarists who need 24-bit audio, high sampling-rate precision, or specialized low-latency drivers, or to anyone who prefers boutique microphone preamps and high-end A/D converters. But it does offer a simple, effective, and affordable 1-cable solution for getting guitar or other instrument signals directly into your computer. For those reasons, the LightSnake is worth looking into. EM

Value (1 through 5): 3
SoundTech
www.soundtech.com

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Waves native processing

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Rich Lexicon reverbs with plug-in convenience

Lexicon is noted for reverbs, and now the new MX400 and MX400XL dual stereo/surround reverb processors bring that classic Lexicon sound to your MOTU desktop studio, without the taxing CPU overhead associated with high-end software-only reverb plug-ins. The new single-rackspace, 4-in, 4-out MX400 combines an intuitive front-panel design with Lexicon's "Hardware Plug-in" technology, a unique USB connection and AU plug-in interface that lets you control and save your MX400 reverb settings directly from within your Digital Performer projects, just like your other plug-ins, while offloading the intensive reverb processing the MX400 hardware. Featuring a wide array of rich, complex reverb algorithms, delays, effects and dbx dynamics, the MX400 series also offers 4-channel surround algorithms that dovetail perfectly with Digital Performer's state-of-the-art surround mixing. A 4-in, 4-out design, the MX400 series offer pro connectivity through XLR balanced I/O (MX400XL), or TRS I/O (MX400). Both products also offer . In any application, the MX400 series products provide versatility and sound that is unmistakably Lexicon.

Control room monitoring

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

Power conditioning

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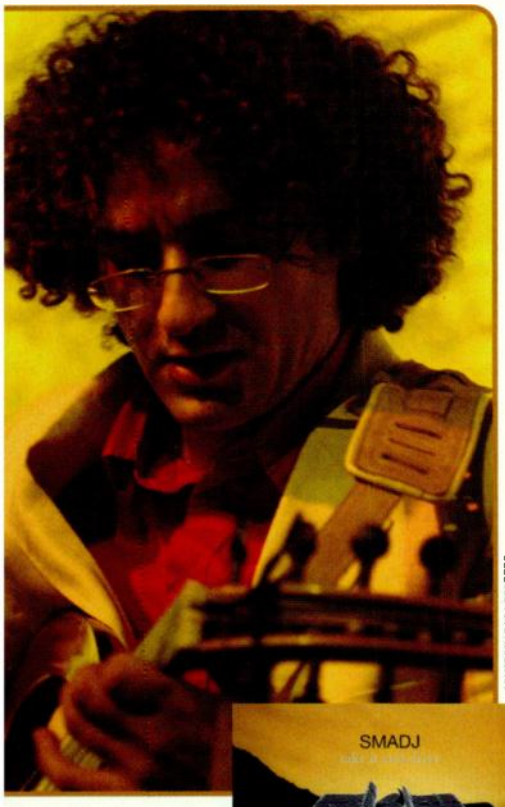
Track It and Drive

By Kenneth Partridge

Smadj records much of his latest CD on the road.

Should Jean-Pierre Smadja—known as Smadj—ever find himself marooned on a desert island, he wouldn't be in any rush to drop a message in a bottle or make a radio out of a coconut. "I'd need my oud and electricity for the laptop, and I'd be fine," says the Tunisian-born musician. "I could make music, and perhaps years after that, you'd have the results."

Indeed, it was with little more than an Apple PowerBook G3 and an oud (an ancient Arabian precursor to the guitar) that Smadj wrote and recorded his latest CD, *Take It and Drive* (Rasa, 2006), over a period of three years. An amalgam of hypnotic beats, Oriental melodies, and Middle Eastern mystique, the CD marries East with West and old with new.



Take It and Drive/Smadj

COURTESY RASA RECORDS

"Each time I encounter a new culture or new music or a new way to play, I try to work with it," says Smadj, who was raised in Paris and now lives in Istanbul. "I discover a culture, and I try to learn it; I try to mix my sound with it."

More than cultural cross-pollination, *Drive* signifies a meeting of heart and hard drive, as Smadj's digital compositions maintain a distinctly human quality. "It was a period where I tried to do electronic, but live," he says.

To accomplish that, he built most songs from the bottom up, using programs such as Ableton Live 3 and Propellerhead Reason 2.5 and ReBirth 2.0 to record beats on the fly and then play them for collaborating musicians, who contributed vocals, guitar, and other instruments. With Steinberg Cubase 5 as his primary recording software, Smadj reached into a vast library of samples he'd compiled over 15 years working as a sound engineer. From there, he'd either load clips onto samplers (such as the Akai MPC 2000 XL and the Ensoniq EPS 16+) or tweak them with software.

RIFFS

Smadj

Home base: Istanbul, Turkey

Key software: Steinberg Cubase 5, Ableton Live 3, Propellerhead Reason 2.5

Primary instrument: Oud

Web site: www.rasamusic.com

When overlaying vocals or oud, Smadj did as necessity dictated. Because much of the album was made while he was on the road, he often recorded in hotel rooms, where having an elaborate microphone setup wasn't feasible. Instead, he relied on mics such as the Aiwa CM-S33 (a stereo condenser), which he plugged directly into his PowerBook.

"The preamp is not so bad on this laptop," he says. "When I was on tour and had an idea, I always took this microphone to record the oud or other instruments."

Time being scarce, Smadj employed a technique he wouldn't ordinarily use, incorporating first-take recordings of each new part into a song's 2-track master as he went. In that way, the masters kept growing, but the track counts remained low—a process he also used when adding contributions from other musicians.

"After each recording—it could be electronic or acoustic—I edited the thing, so I didn't do the classic process of multitrack recording," he says. "Most of the time, I had [a 2-track master] in each step."

That doesn't mean the individual tracks on *Drive* are undoctored. From the robot-stutter vocals of "He Said" (created with Live) to the low bass rumble on "Betty" (a sample looped through a subfrequency generator), Smadj's digital fingerprints are everywhere.

Sometimes the challenge was deciding when to stop tweaking and let a song be. For that, he'd seek the opinions of friends and family, which suggests that he might not be so effective in a Robinson Crusoe–type scenario after all. "You can never stop—you need help," he says with a laugh. "You need someone else telling you, 'Stop it now; it's time.'" EM

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