Make Your Small Room Sound Great

Electronic Musician

April 2004

THE ART OF mixing

Three top pros
share their secrets

How to Capture
the Ultimate Lead Vocal

Ten Things
I Wish I'd Known Then

Reviews: Ableton *Live 3.0*
Sony Pictures *Sound Forge 7.0a*
Yamaha *SPX2000*
IK Multimedia *SampleTank 2*
and 5 more

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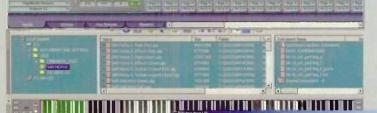

PERFORMANCE

The sleek TRITON Le workstation delivers all the performance wallop of its legendary TRITON brethren in a leaner, meaner package. With new and classic sounds galore, a 16-track sequencer, dual polyphonic arpeggiators, SmartMedia™ storage, and a powerful sampling option, you'll wonder where the "trimming" came in — until you look at the price.

"That keyboard has inspired me to write three new tunes alone in the past month. One of the songs that is going to be on my new record was literally written in five minutes due to inspiration by the TRITON Le." –Eric Johnson

"If you don't have ducats to shell out for a full-blown TRITON, the Le model may look quite attractive to you. And it should. The TRITON Le gives you every bit of synth horsepower available on any other TRITON synth." –Keyboard Magazine

KORG
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TRITON
LE

TRITON
EXTREME

TRITON
STUDIO

INSI

FEATURES

42

Cover photo by Mario Carrieri

62

DEPARTMENTS

COLUMNS

REVIEWS

Technical Nonsupport

We often get reader requests for an exposé of music-tech manufacturers' tech-support practices. Although the idea appeals to my more aggressive side, we have never figured out a fair, accurate approach. After all, we are technical writers, not trained investigative reporters. However, we agree that companies that provide good tech support—and those that don't— should be recognized.

Tech support can be a nightmare for users and manufacturers alike. From the manufacturer's point of view, tech support is expensive. To do the job properly, a company needs highly trained and well-equipped tech-support staff members that are conversant not only with its products, but with other products that might be used with, and could affect the performance of, the company's products.

Although manufacturers are responsible for producing products that work as advertised, we users have to bear the primary responsibility for making our overall systems operate correctly. Yet many users are unwilling to read manuals and do some basic troubleshooting before calling for help. That means tech support, in addition to helping those that have a legitimate need for support, gets blasted with a lot of users who could have found their own solutions. We end users can therefore do our part by doing our homework, troubleshooting what we can, and being prepared.

But the fact is that a surprising amount of tech support in our industry just plain stinks, and that includes support from some major players who should be able to do better. Why is that?

For starters, some companies don't consider tech support a priority because they don't see it as a revenue source. That perception is inaccurate, though, because bad tech support can cost customers without the manufacturer realizing it. As a result, we wait in queues for long periods before we reach a support person. Or we wait for replies to our e-mail inquiries, only to get replies that don't help solve our problem because we need real-time phone support, which might not be offered.

A few companies simply won't admit that they made mistakes in developing a product. That's when we encounter the blame game ("It's the *other* company's plug-in"), or we get treated like idiots because the company can't believe its product is messing up as described.

Most companies could benefit from periodically taking a fresh look at their tech support, including soliciting user feedback and perhaps getting an outside analysis. Supporting products is part of the price of doing business, and customers who repeatedly encounter bad tech support can easily become ex-customers.

So when you need tech support, do your part and be prepared. But if you've done your part yet still have problems with tech support, report it to the company management. If the problem persists, let them know you are ready to give your future business to a rival company that cares about its customers. Then do it.

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The cold, hard facts of room acoustics challenge every studio, large and small. JBL engineers thought it was time for a little magic. Introducing the LSR6300 Studio Monitors, the first reliable solution for tackling the real-world problems inherent in every room. Featuring the exclusive JBL RMC™ Room Mode Correction system, you can accurately measure boundary-induced low frequency modes with the included hand-held acoustic analyzer and then adjust each speaker's 1/10th octave parametric equalizer to correct problems in your room. Designed from the ground up, the LSR6300's uncompromising specs and features give you total control of your music production. See your JBL Dealer today and experience the power of LSR6300 Studio Monitors with RMC - magic you can really put your hands on.

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DMITRY PANICH

GETTING THE GIG

I just wanted to drop a quick note of thanks to the staff of your magazine. I recently became the director of the computer-music lab at the 92nd Street Y, one of New York City's best kept secrets. It may not mean anything to anyone who lives outside of New York, but this is one of the most powerful and progressive community arts-education institutions in the world.

And how did I get the gig? By being a faithful reader of *Electronic Musician*. **EM** has kept me ahead of the pack in terms of knowing the ins and outs of what is available gear wise, and I've always enjoyed the great mix of theoretical and practical articles. Thanks again!

Menon Dwarka
via e-mail

HELP WANTED

I must congratulate you for the good quality of your magazine. As an intermediate-level dance producer, I have questions that, I believe, are interesting for everybody:

How can I measure the soft-synth latency in Cakewalk's Sonar 3 so I can correctly set up the tracks' starting delays? Is there a way to do that automatically, as there is in Steinberg's Nuendo?

What is the best audio-recording/mastering course—available in books

or as a computer course—for learning advanced recording techniques, especially for electronic musicians? Not just a book that explains what is a microphone or a compressor, but an advanced course that explains in detail processor tweaks and settings, advanced equalizing, and fine mixing, pushing students from an intermediate to an advanced level.

David Osivnik
via e-mail

David—Like Cubase SX 2 and Nuendo (rapidly, it seems, becoming one and the same product), Sonar has an autocompensation feature. That means you really don't have to worry about shuffling soft-synth tracks around to stay in sync with your audio.

I asked a few of my colleagues at EM for some suggestions, and everyone seems to like Bob Katz's Mastering Audio, *published by Focal Press (www.focalpress.com). There's also a series from Dave Moulton's Moulton Labs called* Golden Ears *(www.moultonlabs.com/gold.htm) that has received high praise from many folks.—Dennis Miller*

David—In addition to the aforementioned resources, you might want to check out Berkleemusic.com's online production courses.—Steve O

INVESTIGATIVE REPORTING

You do a great job of covering a constantly changing field. Your production tips are always useful, and your reviews and coverage of new technologies are great. I teach an interactive media design course at New York University, and I tell students they should read your magazine on a regular basis.

Here's a suggestion for an article: do a survey or exposé on the technical-

support services and product support offered by the various software companies. Tech support is a joke for most companies. They are hard to reach and often unresponsive, and it's difficult to get an answer from them. We spend good money on software. A review of manufacturers' product support services and track record would be useful.
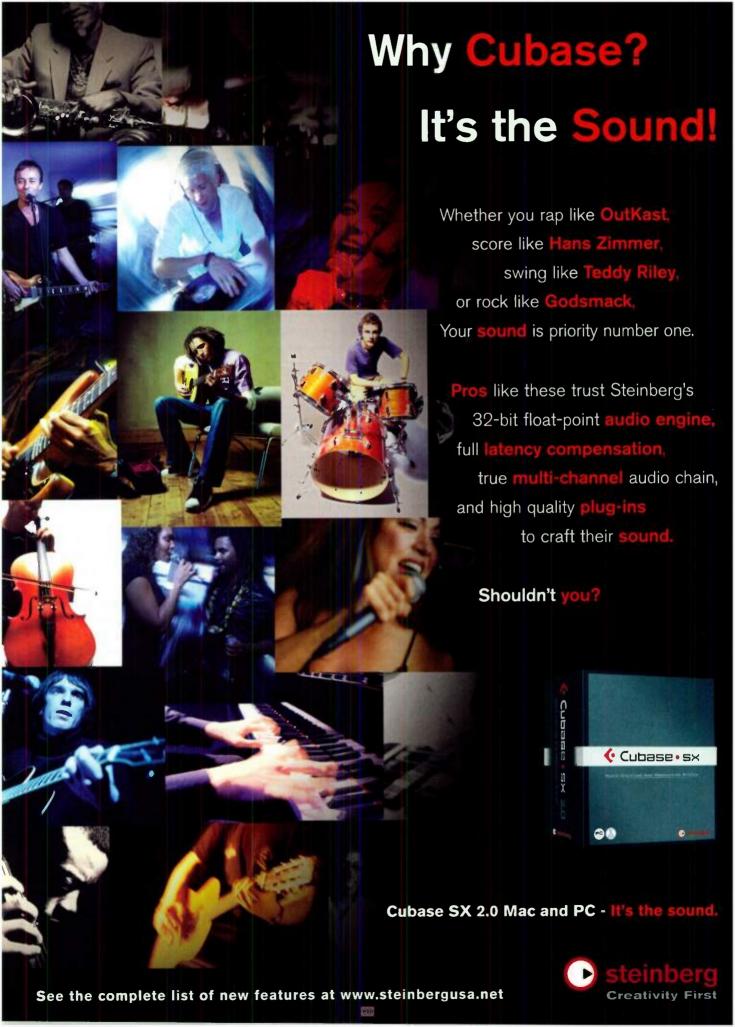
Frank Migliorelli
Interactive Design Director
ESI Design, Inc.

HUSHED TONES

As a guitarist who is always looking for the best ways to capture amp tone in a home studio, I was pleased to see Orren Merton's article in your December 2003 issue ("Recording Musician: Low Volume, Fat Tone"). Mr. Merton outlines some very useful techniques, but I've discovered an incredibly useful tool I thought I'd add to the mix: the Palmer ADIG-LB Filtered Guitar Amp Simulator (www.palmerdirect.com).

After experimenting with cabinet-miking configurations and digital amp-modeling units plugged in directly, I bought the Palmer speaker simulator, a rackmount passive device that simulates the sound of a guitar cabinet extremely well and provides a load for the amp so you don't have to connect speakers to it.

I can now run my Marshall 100W head, Hughes & Kettner zenTera head, or Peavey 5150 head direct to the board and cranked up to any level, and I can monitor with headphones alone. The Palmer shapes the amp's original tone into a great-sounding signal, without speakers! I still have the same

dynamic response I get from miking a 4×12 cabinet.

Nothing is perfect, and purists will always have differing opinions, but I've been playing for 30 years, and I'm very picky about tone. The Palmer's tone is so good in my opinion that I do all of my electric-guitar recording with the Palmer now.

This is not an ad for Palmer—there are other manufacturers of speaker simulators. Palmer is my personal choice.

Marshall Lefferts
Minneapolis, Minnesota

Nice job on the low-volume article ("Recording Musician: Low Volume, Fat Tone," December 2003). I have thought for some time that the low-volume tube-amp space has some serious potential, so I enjoyed your article. Thanks.

Another microamp you might be interested in is Songworks Systems' Little Lanilei ¼-watt, all-tube model. Out of the box, it uses a 12AX7 as a power

tube, which breaks up too quickly for me, so I substituted a 12AU7 and got surprisingly good results—true bedroom levels with decent sound.

The next step up in volume (still with saturation) is a Smicz Amplification Portablues series amp with Smicz's TAD power-tube adapter and a THD Electronics Hot Plate. This combination retains that Class-A articulation and fullness.

Dave Owens
via e-mail

Assistant editor Matt Gallagher adds: Songworks Systems can be contacted on the Web at www.songworks.com; Smicz Amplification at www.smicz-amplification.com; and THD Electronics at www.thdelectronics.com.

ANALOG OPTIONS

I read with great interest your Quick Pick review of the Metasonix TM-3 analog synth module (January 2004). I wanted to comment on interfacing the unit with other rare synths that use the V/Hz (linear) standard. There are other options out there besides adding the Synhouse MIDIJack II.

The PAiA Electronics FatMan (www.paia.com) is a more modern analog synth that outputs V/Hz and is MIDI controlled. I use it very successfully to drive my old PAiA 2700/4700 modular as well as my Korg MS-20 (which requires inverting the trigger). Most MIDI-to-CV converters also can switch between linear and exponential outputs, including the PAiA kit, which has a daughterboard that can be added.

Brian Folkes
via e-mail

STUDIO SECURITY

Kudos on another fine *Electronic Musician*. As usual, the February 2004 issue brims with great ideas and useful articles, and once again, it delivered the goods.

I agreed with Michael Cooper's article on studio safety ("Working Musician: Don't Get Ripped Off," February 2004). I would offer the same advice:

never post your address on a Web site or a flyer unless you're in a commercial space with excellent security and gated access. Posting your address for all the world to see is just asking for trouble someday. Worthwhile clients will find you.

I take Cooper's suggestion of screening phone calls one step further: we don't even accept new clients' calls without first clearing them through our caller ID box. If their ID is blocked or "out of area," tough! Let them leave a message and phone number, and we'll call them back. It may sound harsh, but it easily cuts down on the shady characters.

Remember that anyone can search Google for your phone number nowadays and get your address with a "reverse" name search, unless you specify that the number will not be cross-listed. You want to make sure it's private with the phone company and Web-search engines.

Since we're a small company anyway, I make it clear to my clients that they'll get all our attention while they're here, and everyone else (especially new callers) can get in line and wait their turn. It's tough to do this the first few times, but eventually you get the hang of it, and it's a great way to sort out the time-wasters and troublemakers; they will just move on to easier prey. Assuming your studio gets busy enough to schedule appointments and handle deadlines, you'll eventually break the habit of jumping through hoops for every new caller that rings you up. It's safe to be picky if you can.

Thanks again for a great issue, and keep up the good work!

Joe Hannigan
WestonSound.com

WHAT'S NEW

By Geary Yelton

▼ MOOG MURF

Like other Moogerfoogers, Moog Music's MF-105 Multiple Resonance Filter Array, or MuRF ($435), offers real analog processing and is housed in a rugged enclosure for use on a tabletop or the floor. Eight tuned, resonant bandpass filters have sliders to control each frequency band. A pattern generator called the Animator adjusts filter levels in two banks of 12 sequence patterns to create variations that range from subtle timbral morphing to pulsating grooves. You can control the MuRF's real-time parameters with front-panel knobs or external control-voltages sources, and you can use the tempo-tap function to sync to music. Moog Music; tel. (800) 948-1990 or (828) 251-0090; e-mail info@moogmusic .com; Web www.moogmusic.com.

▼ KORG LEGACY COLLECTION

Korg, maker of the classic MS-20, Polysix, and Wavestation synthesizers, has announced the Legacy Collection (Mac/Win, $625), which provides native software versions of those instruments bundled with 20 effects plug-ins. All three soft synths operate as standalone programs or as VST or Audio Units plug-ins. Also included is the MS-20 USB/MIDI controller, a four-fifths-scale copy of the original MS-20 that features a Velocity-sensitive minikeyboard and patch-cord programmability.

The monophonic MS-20, introduced in 1978, featured two VCOs, two multimode VCFs, two VCAs, a front-panel patch bay, and the ability to process external audio. Three years later, the Polysix stored 32 programs and had a VCO, suboscillator, lowpass VCF, and VCA for each of six voices. The Legacy Collection updates these synths by adding 32-note polyphony, MIDI sync, and plenty of other modern conveniences. The Wavestation, launched in 1990, incorporated rhythmic wave sequencing and vector control to morph between four oscillators. The virtual Wavestation supplies all the factory waveforms, Performances, and Patches of the rackmount Wavestation SR module, and it imports Wavestation SysEx data.

A feature called the Legacy Cell lets you combine the MS-20 and Polysix, two insert effects, and two master effects into a single new instrument. The Legacy Collection runs in Mac OS X on a Mac G4 and in Windows XP on a Pentium 4 SSE2. Korg USA, Inc.; tel. (516) 333-9100; Web www.korg.com.

▼ TASCAM 2488

Remember the 20th century, when 24-track recorders cost thousands of dollars? Tascam, manufacturer of the original PortaStudio, has announced the most affordable 24-track digital audio workstation yet—the 2488 Digital PortaStudio ($1,499). It records eight simultaneous tracks of 24-bit, 44.1 kHz audio and offers 250 virtual tracks. With four combination phantom-powered XLR and ¼-inch inputs and four ¼-inch mic/line inputs, the 2488 provides 36 channels at mixdown: 24 playback tracks, 8 analog inputs, an internal stereo General MIDI synth, and a stereo effects processor. All playback channels, except the effects returns, are equipped with three auxiliary sends and 3-band EQ.

Twenty 45 mm faders and dozens of illuminated buttons offer plenty of hands-on control. The backlit display lets you view meter levels and graphically edit parameters and audio waveforms. With the built-in CD-RW drive, you can burn stereo mixdowns to disc, back up the internal hard disk, and import or export WAV files. The 2488 can even play Standard MIDI Files with its 64-note polyphonic sound module.

The 2488's USB 2.0 port lets you exchange audio and data files with your Mac or PC. One multi-effects processor provides reverb, delay, chorus, and other insert effects, and another is optimized for electric guitar. The stereo bus has compression and 3-band EQ for processing the mix. Other features include tap tempo, recallable scenes, autolocate, autopunch, and undo capabilities. Tascam; tel. (323) 726-0303; Web www.tascam.com.

Come Hear the World

through Waves...

Waves signal processing software and hardware has become central to professional production for Music Recording, Mixing and Mastering, and in production for Television, Film and Broadcast. Waves is committed to the continued development of the finest tools for audio signal processing available. Waves has a comprehensive selection of Award winning bundles to suit your needs including: The New IR-1, and Musicians bundle and Diamond, Platinum, Gold, Transform, Restoration, 360°, Surround, Renaissance Maxx bundles, and more. Waves has the right tools for the job, and Sweetwater can help you determine which if those tools to use to get the job done.

DOWNLOAD OF THE MONTH▲ ▲ ▲ ▲

▶ AMBIENCE
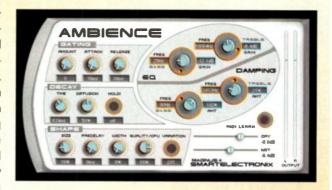
Ambience (Mac/Win, donationware), by SmartElectronix's Magnus Jonsson, is a reverb plug-in with excellent sound quality and an unusual combination of features. It comes in VST and Audio Unit plug-in formats, but the AU version is currently without a GUI. Ambience is donationware, which means you can choose when and how much to pay for it. The demonstration version is fully functional, but it presents a "nag" screen each time it is launched; a donation gets you the non-nagging version.

The Ambience control panel's Decay and Shape sections control basic reverb characteristics such as room size, predelay, reverb time, and diffusion. Clicking on the Variation button changes the characteristics of the room slightly, which is useful for eliminating unwanted echo and ringing. Clicking the Hold button freezes the reverb, which is useful for artificially stretching percussive sounds, for example. When used in con-

junction with automation or MIDI remote control, Hold offers unique sound-effects possibilities. The Shape section contains a Width knob, for adjusting the stereo image from mono to full stereo, and a Quality button, for reducing the CPU load (which can be considerable) at the expense of reverb quality.

Three other control-panel regions—Gating, EQ, and Damping—process the reverb output in different ways. Gating applies a threshold-triggered attack and release envelope, allowing the dynamics of the source to affect the reverb characteristics. It's especially useful for maintaining a percussive part's rhythm in a very wet reverb (see **Web Clip 1**). The Damping section allows you to alter the decay times for low and high frequencies as a percentage of the Time set-

ting. (In a natural reverb, highs and lows tend to decay faster than midrange frequencies.) The EQ section provides high and low shelving of the final output.

Ambience is a great-sounding reverb that's easy to use, yet offers unusual effects-processing features. You can download it from the SmartElectronix Web site (www.smartelectronix.com/~magnus). Trying out the demo is definitely worth your time: Ambience may well become your favorite reverb.

—Len Sasso

▼ ROLAND V-PRO TD-20S-BK

Roland has unveiled its new flagship V-Drum kit, the V-Pro Series TD-20S-BK ($6,495). The company says that the kit has been engineered to feel more like a real musical instrument than previous electronic drum systems. Sporting a black finish, the TD-20S-BK incorporates the new TD-20 sound module, improved mesh V-Pads, a dual-cymbal V-Hi-Hat, and an ergonomically designed drum stand.

The TD-20 offers more than 500 new sounds, greater expressiveness, and a larger dynamic range than its predecessors, according to Roland. You can select from several modeled kick beaters, change virtual cymbal characteristics, and add snare-drum buzz to the kick and toms to emulate

sympathetic resonances. The TD-20 can vary the drum and cymbal sounds in response to your playing and produce more realistic snare-drum rolls and cymbal swells. In addition to 15 dual-trigger inputs, 10 analog outputs, and a S/PDIF output, the TD-20 has MIDI I/O, 8 group faders, 15 compressors and EQs, onboard effects, and a 5-part sequencer.

Included are three PD-125BK (12-inch) and two PD-105BK (10-inch) dual-trigger V-Pads, which offer improved head and rim triggering compared to previous V-Pads. The system also includes the VH-12 V-Hi-Hat—which uses a pair of cymbals for a natural feel and mounts on any hi-hat stand—and two CY-14C crash cymbals, a CY-15R ride cymbal, the new KD-125BK V-Kick pad, and the new MDS-20BK drum stand. Roland Corporation U.S.; tel. (323) 890-3700; Web www.rolandus.com.

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▲ MUSKA & LIPMAN PUBLISHING

Chris Middleton's *Complete Guide to Digital Audio* ($39.99) is probably the most attractive softbound coffee-table book on studio technology ever published. Subtitled *A Comprehensive Guide to Digital Sound and Music-Making*, the lavishly illustrated 192-page book introduces dozens of subjects concerning electronic musicians, from the history and basics of acoustics, sound production, and recording to computers, digital instruments, and studio setup.

Scattered throughout are quotes from Brian Eno, Laurie Anderson, Peter Gabriel, and others. Color photos and screen shots take up at least as much space as the text. I suspect this book will find its way into many recording studio lounges, if only to provide eye candy for waiting clients. Muska & Lipman Publishing (a division of Course Technology); tel. (800) 354-9706; e-mail ct.retail@thomson.com; Web www.courseptr.com.

PEACHPIT PRESS

Among the most recent books in the Apple Pro Training Series is Mary Plummer's *Soundtrack: Create Original Scores for Video, DVD, and the Web* ($39.99). Most of *Soundtrack* provides hands-on instruction using materials from the included DVD-ROM. Plummer introduces Soundtrack's user interface, then progresses to arranging, editing, and mixing songs. She explains how to create temporary scores, sync to video, prepare music for various media, and use the Soundtrack Loop Utility.

When you've completed this 458-page guide, you'll qualify to take a Soundtrack certification exam at an Apple Authorized Training Center. *Soundtrack* is suitable for classroom use or self-instruction, and each of the nine lessons takes from 30 to 90 minutes to complete. If you don't already have Apple's loop-oriented composition software, a downloadable 30-day demo should soon be available on Peachpit's Web site. Peachpit Press; tel. (800) 283-9444 or (510) 524-2178; Web www.peachpit.com.

▲ BACKBEAT BOOKS

Whether you've been using Pro Tools for a while or you're trying to decide which version is right for you, *The Complete Pro Tools Handbook* ($49.95) has plenty to offer. A tutorial guide, José "Chilitos" Valenzuela's massive 518-page book covers the fine points of Pro Tools LE, Pro Tools|24 Mix, and Pro Tools|HD. It describes the various TDM versions and their associated hardware, and it gives a similar overview of the native versions. It moves on to illustrated exercises that use audio clips from the included CD-ROM.
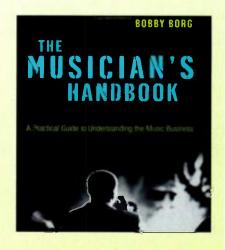
Most of the book focuses on recording, editing, and mastering techniques. It even devotes 67 pages to MIDI. Backbeat Books; tel. (866) 222-5232 or (408) 848-8294; e-mail books@musicplayer.com; Web www.backbeatbooks.com.

▼ BILLBOARD BOOKS

No matter how long you've been playing music, good advice on advancing your career is always welcome. That's just what you'll find in *The Musician's Handbook: A Practical Guide to Understanding the Music Business* ($19.95). Written by Berklee graduate Bobby Borg, a working drummer who has taught music business at UCLA and the Musician's Institute, the book explains how to set and attain goals by approaching your career as a business.

Heavy on text and light on graphics, *The Musician's Handbook* details how to define your target audience, find a market for your talents, and establish career-building relationships. Borg clarifies the roles of business and personal managers, attorneys, talent agents, and record producers. He writes about working solo or in a band, performing live, publishing and promoting your music, and other aspects of surviving and thriving as a professional musician. Billboard Books; Web www.watsonguptill.com.

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▶ WAVES IR-1 AND MUSICIAN'S BUNDLE

Waves is shipping two new processor plug-in products, IR-1 (Mac/Win; native, $800; TDM, $1,200) and the Musician's Bundle (Mac/Win, $200). IR-1 is a convolution reverb processor that furnishes traditional reverb controls for editing sampled environments. IR-1 includes more than a gigabyte of sampled impulse responses, featuring more than 60 concert halls, theaters, stadiums, churches, and other environments. The spaces range from the Luzit Caves in Israel to Nashville's Bluebird Cafe. Each space was sampled from several locations at 96 kHz, with 32-bit floating-point resolution. IR-1 supports DirectX in Windows; Audio Units, MAS, and HTDM on the Mac; and VST, RTAS, and Audio Suite on either platform.

The Musician's Bundle comprises five processing plug-ins for native host-based systems. Renaissance Axx is a compressor, limiter, gate, and expander designed for guitar and bass tracks. Voxx is a similar plug-in optimized for vocal processing. Doubler combines delay and pitch modulation to create track-doubling effects. SuperTap has six multitap delays with numerous modulation options. MetaFlanger provides sophisticated flanging, phasing, and chorusing effects. The Musician's Bundle supports DirectX in Windows; MAS on the Mac; and VST, AudioSuite, and RTAS on both platforms.

Free 14-day demos of IR-1 and the Musician's Bundle are available for download. Waves; tel. (865) 546-6115; e-mail info@waves.com; Web www.waves.com.

▼ NOVATION REMOTE AUDIO XTREME

Synth manufacturer Novation EMS has introduced two-, four-, and five-octave versions of its USB/MIDI keyboard controller and 24-bit audio interface: the ReMote Audio 25 ($1,099), ReMote Audio 49 ($1,249), and ReMote Audio 61 ($1,399).

Featuring a semiweighted action with Aftertouch response, a programmable *x-y* touch pad, and a pitch-bend and modulation joystick, the controller also provides 22 knobs, 9 sliders, and 24 buttons—all user-assignable. The Xtreme software package, available at no extra charge, adds the V-Station virtual analog synth plug-in, Steinberg Cubase SE, and a special edition of Ableton Live. The four- and five-octave models also come with Steinberg's sampled acoustic piano plug-in, The Grand SE.

Sixty-four user-programmable software templates (more than 50 are included) let you customize the front panel to control a variety of virtual instruments and sequencers. You can quickly scroll through the templates, and dedicated transport controls make sequencing faster. On the ReMote Audio's back panel are two Neutrik combo connectors for audio input, two balanced ¼-inch outputs, a S/PDIF output, a ¼-inch stereo headphone jack, expression and sustain pedal jacks, three MIDI ports, a USB port, and a wall-wart connection. The ReMote Audio can also receive power through USB or from six C batteries. Novation Music/Von Audio (distributor); tel. (727) 786-8777; e-mail sales@vonaudio.net; Web www.vonaudio.net or www.novationmusic.com

▶ EVENTIDE OCTAVOX AND REVERB

Bringing more classic effects to the Pro Tools environment, Eventide has released Octavox (Mac, $595) and Reverb (Mac, $695), a pair of TDM plug-ins for Digidesign Pro Tools|HD and Pro Tools|HD Accel. Both plug-ins are based on Eventide's flagship processor, the Orville.

The Octavox Harmonizer plug-in is an eight-voice diatonic pitch shifter that can produce rhythmic musical sequences by applying tempo-based delay to each voice. You control pitch and delay using the Notation Grid, which presents notes on a traditional music staff. A randomizer simulates the natural variations of eight individual sound sources.

Reverb borrows several high-quality algorithms from the Orville, including halls, chambers, plates, ambience, and lo-fi effects. For each reverb type, Reverb provides user parameters for contour, stereo delay, and compression, as well as 3-band stereo parametric EQ both pre- and postreverb.

Both plug-ins can store snapshots of parameter settings as presets, and both support Mackie's HUI and Digidesign's ProControl and Control24. They also support sampling rates as high as 96 kHz on Pro Tools|HD Accel systems and use the Pace iLok for copy protection. You'll need a Pro Tools–compatible Mac, Pro Tools 5.1.3 or greater, and Pro Tools|HD or Accel hardware. Eventide; tel. (201) 641-1200; e-mail audio@eventide.com; Web www.eventide.com. ✪

By Scott Wilkinson
and Joanna Cazden

Cybersinger

A new musical lifeform lives in the Internet.

Throughout history, many of the most important discoveries have been accidental. For example, in 1778, while experimenting with certain chemicals, Joseph Priestly found an unexpected gas, which he called "phlogisticated air." This gas was later identified as nitrous oxide, or "laughing gas" (which gets you pretty phlogisticated, that's for sure!).

Perhaps the biggest discovery of all time was made just last year by Dr. Spanky N.R. Ganglia of Callosum Corp. U.S. As you may recall from a previous column (see "Tech Page: Music on the Brain" in the April 2000 **EM**), Ganglia has developed a remarkable device called the Mindophone, which detects electrical brain activity associated with musical thought and translates it directly into audio signals. This allows users to record any music they "hear" in their head, but without bothering with all those annoying lessons and practice sessions.

More recently, Ganglia's laboratory announced the completion of a synaptic syntax called M-Dopa (musical dopamine), which acts like an online neurotransmitter and allows Mindophone signals to be streamed over the Internet. To test this function, he loaded the M-Dopa algorithms into the company's central server and connected the Mindophone to one of the server's ports. Ganglia's goal was to send musical signals from a subject's brain through the device to another computer on the network, which would then play the signals from its sound card.

However, when Ganglia fired the system up, he discovered something astonishing. Before his subject even put on the Mindophone headpiece, music started emanating from the target computer. It was strange music, too: a melange of pop, rock, jazz, rap, classical, and all 142,673 subgenres of electronica, including trance, dance, prance, hip-hop, skip-hop,

The human brain's neocortex has 20 billion neurons, which is only 0.002 percent of the quadrillion transistors that are actively connected to the Internet at any given moment. What kind of music could you make if your brain had 50,000 times the neurons it does now?

ambient, hellbient, nambi-pambient, and so on.

On a hunch, Ganglia typed "Hello" on the computer's keyboard. "Hello yourself," came the quick reply on the screen. "Who are you?" asked Ganglia. "I am me." He typed, "Where are you?" and read the response, "Everywhere."
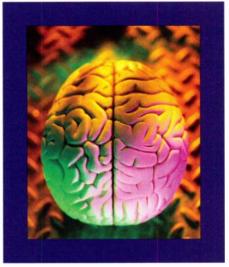
Ganglia was stunned. He realized that he was conversing with a new lifeform: the Internet itself had become sentient, and his M-Dopa software was allowing humans to hear its musical musings for the first time. He also realized that music files pervade the Web, providing a rich resource for the evolving Net-mind's self-expression.

Meanwhile, Ganglia had to explain his stupendous discovery to Occipital Investments, the venture capital group funding his research. He pointed out that the Internet, like the human brain, has a vast number of nodes and links. Fifty million servers are connected to half a billion personal computers worldwide, and every one of them has 30 to 60 million transistors. Even if only a tenth of those computers are online at any given time, the Internet hums through a quadrillion transistors, 50,000 times more than the brain's 20 billion neocortical neurons from which human consciousness arises. If the Net-mind had surpassed that level of complexity, no wonder it felt like singing!

For now, Ganglia's discovery remains its own reward. But who knows what the new entity—which has chosen to call itself HAL—might do? What new meaning will "world music" take on as a worldwide intelligence starts creating it? Will the universe of audio files (and audiophiles) evolve toward unimaginable harmony or toward catastrophic cacophony? Ganglia wondered, helmeted, in awe. The M-Doped Mindophone transmitted his questions to HAL, which responded reassuringly with its rendition of "A Bicycle Built for Two." ☻

By Matt Gallagher

Live from New York

Terry Silverlight records a burning fusion album with all-star guests.

A s a first-call drummer in New York City, Terry Silverlight works with many top-level musicians. He also works in his personal studio as a composer, a producer, and an arranger. *Wild* (Silverlight Records, 2004) is Silverlight's second solo album, which he mixed at home and coproduced with Will Lee. "It revisits my jazz-fusion roots," Silverlight says. For *Wild,* Silverlight assembled distinguished musical guests including Lee, Paul Shaffer, Edgar Winter, Hiram Bullock, Chuck Loeb, Lew Soloff, David Mann, Charles Blenzig, Mike Ricchiuti, and John Clark.

"My goal was to capture the spontaneity and energy of a live performance," Silverlight says. Lee advised Silverlight to record at New York's Avatar Studios. "We booked two half-days," Silverlight says. "Avatar's Studio B is a smaller room. [Engineer] Jason Corsaro set up the room so we could all see one another. My drums were in the middle of the room while the grand piano, Rhodes, B-3 organ, sax, and trumpet were [each] partitioned off. Will played electric bass and went direct. [Guitarists] Chuck and Hiram went direct in stereo, and their amps were placed in a partitioned corner. I was amazed at the minimal leakage.

"Everything went through a Neve console directly into Pro Tools," Silverlight says. "Each drum and cymbal was miked individually, and Jason tuned the drums to match the timbre of the room. When I mixed, every time I'd add EQ, compression, or gates, it would detract from the original recording. So, aside from a little reverb on the snare, you hear the drums the way they were recorded." Pro Tools engineer Lawrence Manchester converted the sessions into MOTU Digital Performer files so that Silverlight could continue working on the album at home.

"There was very little overdubbing," Silverlight reveals. Clark and Winter recorded parts in their own studios and provided Silverlight with AIFF files. "Dave and Lew overdubbed themselves a few times on 'Brown 'n Serve' at Avatar," Silverlight adds. "I recorded myself on keyboards in my studio on 'Windsurfing,' 'Wild,' and 'WTC.'

"I mixed the entire album using DP3, a dual-processor Mac G4, and the Tascam DM-24 digital board," Silverlight says. "I used the DM-24 as a slave to DP. I used some reverb and EQ on the DM-24, but all other effects were plug-ins. I don't own high-end outboard gear. Plus, plug-ins are automated, and you can save and recall settings instantly."

Working with numerous large audio files proved to be a major hurdle. "My computer's processor was taxed," Silverlight says. "As a work-around, I often bounced individual tracks with plug-ins, fades, and volume curves embedded. I then went back to the original tracks, disabled the plug-ins, and removed their voice allocation, which freed up memory and processing power. In DP, you can save the mixing board's layout with all plug-ins enabled, and then disable the plug-ins and save that as a separate setup."

Silverlight tested his mixes in varied listening environments. "I listened in headphones, on my Yamaha NS10Ms, on home-stereo speakers, in my car, and on other people's systems, including Will's," he says. "They sounded different in every situation, so I had to come up with EQ settings that seemed to translate equally on every system.

"The music features an interplay between all of the musicians that could only be captured live, not as overdubs," Silverlight concludes. "I tracked in a great studio to achieve quality sounds from each instrument, instead of trying to replicate that [sound] in my own studio or resorting to overdubbing. I think I got exactly what I had intended musically." ◈

For more information, contact Silverlight Records; Web www .terrysilverlight.com.

DAVID SOKOL

***Wild*/Terry Silverlight**
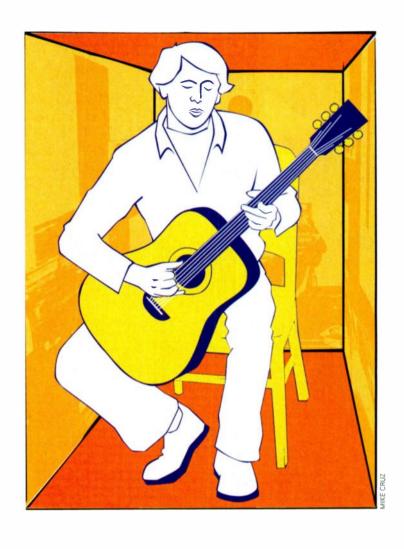
A New Approach to Personal Studio Acoustics

How small-room acoustics affect the music you record and mix.

MIKE CRUZ

More recording and mixing is being done in very small rooms than ever before. Small-room recordists include not only amateurs and weekend engineers in their home studios but also serious professionals doing critical music projects for major labels. Much of the conventional wisdom about room acoustics, measurement, and treatment fails to take into account the unique behavior of small rooms. This article explains some important but often misunderstood or overlooked acoustic principles relating to low frequencies in small rooms.

Many people believe that room acoustics is a complicated subject that can be understood only by those who have a Ph.D. in math or physics. However, the behavior of sound waves in small rooms is actually pretty simple, at least for the purpose of addressing problems that are common to recording studios and control rooms.

All acoustic anomalies are caused by reflections off the walls, floor, and ceiling. I make a distinction, however, between problems caused by reflections at low frequencies (below about 300 Hz) and those caused by midrange and high-frequency reflections. Above 300 Hz, reflections are perceived mainly as echoes, ambience, and reverb. Below 300 Hz, skewed frequency response is a much bigger problem. In all cases, waves bounce around the room much like a cue ball on a pool table; unlike a cue ball's, though, their behavior changes in three dimensions.

By Ethan Winer

Four!

A New Approach to Personal Studio Acoustics

PEAKS AND NULLS

Contrary to popular belief, all small rooms have *peaks* (boosts in amplitude) and *nulls* (dips in amplitude) at all low frequencies, not just the mode frequencies determined by the room's dimensions (see the sidebar "Acoustical Physics"). That situation has a profound effect on how you should approach low-frequency acoustic problems. For any given low frequency, every room has places where a peak exists and others where a deep null exists. Likewise, for any given location in the room, there will be frequencies with a peak in the response and others with a null. Specifically, a null occurs at a distance from most boundaries (walls, floor, and ceiling) equal to one-fourth the wavelength of the frequency. (For more information, see Wes Lachot's article, "Bass Waves in the Control Room," at www.ethanwiner.com/waves_wl.html.) Other nulls occur at odd multiples of that distance: three-fourths of the wavelength, five-fourths, and so on. Similarly, peaks occur at two-fourths of the wavelength, four-fourths, six-fourths, and so on.

Some boundaries have stronger peaks and nulls than others, due to multiple reflected waves coming from different directions and combining in the air. In fact, all acoustical problems in all rooms are caused by reflections.

ACOUSTICAL PHYSICS

Nodes, modes, and standing waves are three key properties that all rooms have, and they are closely related to each other. A *node* is a place in a room where a null or dip in the frequency response occurs. A node is caused when two waves meet in the air and combine out of phase (see **Fig. A**).

In a typical case, waves emitted from a loudspeaker reach a wall and are reflected back into the room. At some distance from the wall the original wave will have a positive pressure, while the reflected wave is negative—or vice versa. If the reflected wave is exactly equal in level and is exactly 180 degrees out of phase with the original, the waves would cancel completely at that particular location. At other levels and phase relationships, the waves will cancel to a lesser degree. (When they're in phase, they increase in level instead of canceling.) Total cancellation never occurs in practice, because no wall is 100 percent reflective at any frequency.

A *mode* is a natural resonance that occurs in a room, and the frequency of the resonance depends on the room's dimensions. A typical rectangular room has three fundamental mode frequencies: one for the length, another for the width, and another for the height. Sound travels at a speed of approximately 1,130 feet per second, so the resonant frequency between two opposite walls can be determined by the following formula, in which "Feet" is the distance from one wall to the other:

$$\text{Frequency} = \frac{1{,}130}{\text{Feet} \times 2}$$

Twice the distance is used because a wave travels from one side of the room and back to complete one cycle. Each dimension actually has a series of modes, because higher frequencies can also occupy the same distance. That is, wall spacing that exactly fits one cycle of 70 Hz also accommodates two cycles of 140 Hz, three cycles of 210 Hz, and so forth.

The most common type of mode is the *axial* mode, which occurs between two opposing surfaces such as the floor and ceiling. There are also *tangential* and *oblique* modes, which are weaker and thus have less impact on the room's response. Tangential modes complete one or more cycles after bouncing off four room surfaces, literally like a cue ball going around a pool table in a diamond shape. Oblique modes are weaker still and bounce off all six surfaces to complete one or more cycles.

A *standing wave* is a wave
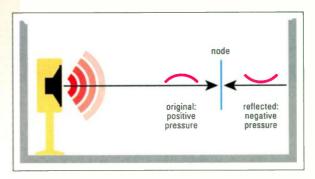
FIG. A: A node occurs when direct and reflected waves that are out of phase with each other collide in the air.

that's not moving—it literally stands still. Standing waves occur at node locations in the room, and they result when two equal yet opposite waves arrive from different directions and collide. A few inches away, just outside the node, the waves are traveling toward each other. There's no motion, though, at the one precise location where the wave fronts meet. (This is much like the isometric exercise of pushing your hands together.) Some people wrongly consider modes and standing waves to be the same thing, because standing waves can occur at modal frequencies. But they are not at all the same, because one is a wave and the other, a mode, is merely a propensity to vibrate. Moreover, opposing waves can create nulls at nearly any frequency in any room, not just those frequencies that correspond to the room's dimensions.

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A New Approach to Personal Studio Acoustics

Because the peaks and nulls in small rooms occur at regularly spaced frequency intervals, the net result can be considered a type of comb filter. That is exactly how flanger and phaser effects work, except in this case the filtering happens acoustically in the air as the waves collide, reinforcing or canceling each other. The general term for this phenomenon is *acoustic interference.*

The effect is much more pronounced in small rooms than larger ones because the walls are closer together and the reflections are therefore stronger. The strength of the reflections also depends on the density of the walls, with rigid walls reflecting more frequencies and lower frequencies. Indeed, the worst environment for a home studio is a basement, because cement walls are more rigid than standard Sheetrock walls. One of the great ironies of acoustics is that thick, rigid walls that improve isolation between rooms cause more acoustic problems within the rooms

than they solve. With standard walls made of one layer of Sheetrock, the lowest frequencies pass through to some extent and are partly absorbed when the wall vibrates in sympathy. By reflecting more frequencies and lower frequencies, walls made of cement or multiple layers of Sheetrock increase the damage caused by acoustic interference.

As I explained, peaks and nulls occur at predictable quarter-wavelength distances from every room boundary. The nulls are often strongest at the rear wall because the loudspeaker's *wave front* (the initial change in air pressure caused by a sound) that travels the length of the room is strongest in that direction. But other reflections occur at other boundaries, and they combine in and out of phase to bolster or reduce the quarter-wavelength nulls. To create a deep null, the opposing wave fronts must be nearly identical in level. It takes very little contribution from an errant reflection arriving from somewhere else
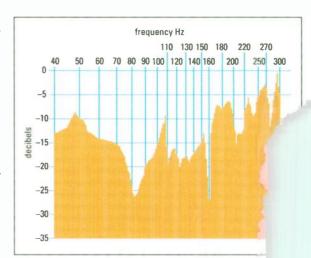
FIG. 1: This frequency response was plotted at 1 Hz inter[val] control room measuring 16 × 10 feet and 7.5 feet high. A[s bad] as it looks, this is typical for small, untreated rooms.

to disturb the precise balance to create a deep null.

If you want to find out just h[ow seri]ous the acoustic problems are [in your] own control room, you'll nee[d a sine] wave generator and a tape m[easure.] But you can't just place a mic[rophone or your] ear at a distance from a wall [and ex]pect to find a deep null. You al[so have] to move up and down, and [left and] right, to find where the null is [least in]fluenced by a competing reflec[tion from] the floor, ceiling, or another wall.

KARMA MECHANICS

I want to share with you some conventional wisdom that is valid, in spite of the fact that few people understand why. Many recordists who don't have proper acoustic treatment have learned to play mixes in their cars in order to get a better sense of the bass levels. Of course, most car stereos are a poor second to a good pair of monitor speakers. Yet that method works surprisingly well, discounting the nuisance of having to keep burning CD-Rs to play in your car. Many people think that a car is a good place to assess mixes because we spend so much time listening there, but they also listen through their studio monitors.

As you know by now, acoustic reflections cause a series of peaks and dips throughout the entire low-frequency range. Those problems are much worse than an overall lack of bass or an overall increase in bass, for which you could more easily compensate. For whatever ills most car stereos have, they do not usually suffer as much from acoustic comb filtering, because much of the low-frequency energy passes right through the car's lightweight walls to the outside. By passing through the walls instead of being reflected, the low-end response is more uniform than in many rooms. You can easily prove this to yourself: play some bass-heavy music fairly

loudly in your car, then roll up all the windows and get out. All you will hear outside the car is the bass that escapes through the walls and windows instead of being reflected.

Note that some car stereos, and many boom boxes too, have a permanent loudness-type compensating boost at the upper-bass range. That boost is intended to fool inexperienced listeners into thinking the system has more bass than it really does. If your car or boom box has such a boost, you'll have trouble hearing bass accurately. But many car stereos, including the stock stereo in my aging '93 Camry, have a surprisingly flat and extended low end.

In most rooms, the peaks caused by acoustic interference are usually less than 6 dB, but the nulls are typically 30 dB deep or more (see **Fig. 1**). In fact, most people first become aware of bass problems in their rooms when they notice a lack of bass at the mix position compared with other locations. A slight peak is not nearly as noticeable or damaging as a deep null. Further, many nulls have a very narrow bandwidth.

WHAT YOU KNOW IS WRONG

Standard real-time room analysis using pink noise to measure the frequency response in 1⁄3-octave bands completely misses the peaks and nulls that exist in all rooms. When pink noise is analyzed in bands, the levels of all frequencies within each band are averaged together. Even measuring at 1⁄12-octaves is far too coarse to see the true room response. I have observed peaks and adjacent nulls at less than 1⁄12-octave spacing in many small rooms. Unless you measure the exact frequencies of the peaks and nulls that occur where you place the microphone, however, a room might appear perfectly flat when it actually has many large, hidden aberrations.
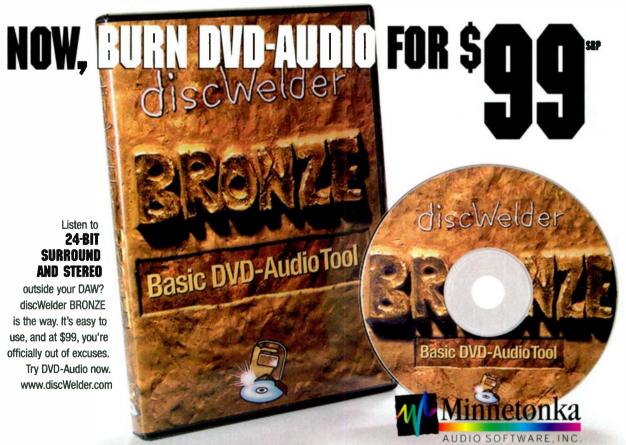
One excellent way to measure the true low-frequency response of a room without having to use special equipment or analysis software is to play low-frequency sine waves and measure the result at various locations (see **Fig. 2**). Such a test is useful because all music ultimately consists of sine waves that sustain for some length of time.

FIG. 2: Acoustician Doug Ferrara prepares to measure the response of a typical project studio's control room.

When a bass player holds a long note in a slow ballad, that note contains primarily two sine waves: the fundamental
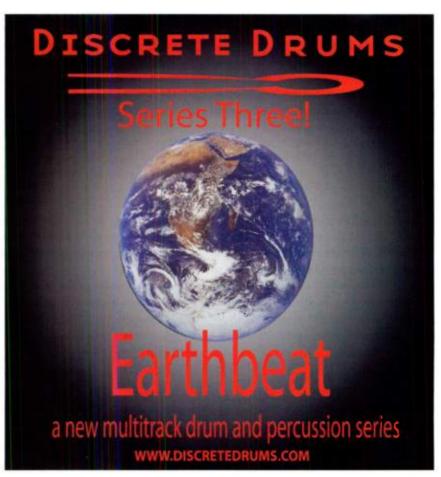
pitch and its second harmonic. Likewise, kick drums produce mostly sine waves, though the waves often fade quickly and are accompanied by the click sound of the beater against the drumhead. Cymbals, maracas, and every other musical instrument create sound composed entirely of sine waves, even if those waves sustain only briefly.

Even if you measure the room using a bandwidth of 1 Hz (whether with pink noise, impulses, or sine waves) you must consider another issue: the physical size of acoustic nulls is often extremely tiny, so the notion of a sweet spot for mixing is meaningless unless you're willing to clamp your head in a vise. In one test, I located the physical center of a deep null at 100 Hz in my own control room. I then moved the microphone four inches to the side—equivalent to turning your head a little—and the level rose by 15 dB. When the mic was 18 inches from the null center, the level was 20 dB higher. If you measure a room's response using sine waves 1 Hz apart, you'll still need to measure every frequency at dozens of locations within a cubic foot or two of the mix position to get the true picture (see Fig. 3).

One saving factor is that our ears are spaced several inches apart. When one ear is in the center of a very deep null for a given frequency, the other ear is likely to be out of that null, though it might be in the center of another. The severity, narrow bandwidth (high Q), and small physical size of acoustic nulls are the main reasons equalization can never correct low-frequency problems in small rooms. Whatever you do to flatten the response in one location will surely make things much worse somewhere nearby.

The old-school method of acoustically treating low frequencies is to calculate the room's modes based on its dimensions and then design custom bass traps that target those specific frequencies. That approach is inadequate because it addresses only the modal frequencies, ignoring the peaks and nulls caused by acoustic interference that occur at all other low frequencies. Further, even if you consider only the

first five axial modes for each dimension, that would still require building bass traps for as many as 15 different frequencies. A much better solution would be to use broadband absorption, because that flattens the response throughout the entire low-frequency range.

In some severe cases—say, with a room that's 8 × 8 × 8 feet—it could be useful to complement broadband absorption with traps that target the enormous resonance that exists when all three dimensions are the same. In this case, the resonance is at 70 Hz, but similar resonances exist at all multiples of 70 Hz, so broadband trapping is still needed because the related frequencies must also be treated. (Besides, any room that's only 8 × 8 × 8 feet probably doesn't have space for a sufficient number of traps that target such a low frequency.)

MIX AND MATCH

Another important point, which again defies conventional wisdom, is dispelling the myth that you can learn to make great mixes in an untreated room. The biggest problem most people have when mixing is getting the right bass levels. Often a mix that sounds correct in your control room will sound boomy when played elsewhere. Most small rooms have a deep null at the mix position in the vicinity of 80 Hz. (The exact frequency depends largely on how far you sit from the rear wall.) As a result, you tend to mix with too much bass to compensate for what you're hearing. When a mix can be made to sound good both inside and outside your control room, it's said to be *portable*.

Studio designers often advise playing a commercial CD of the same type of music as you're mixing, with the goal of matching the bass levels to obtain a portable mix. The problem with that
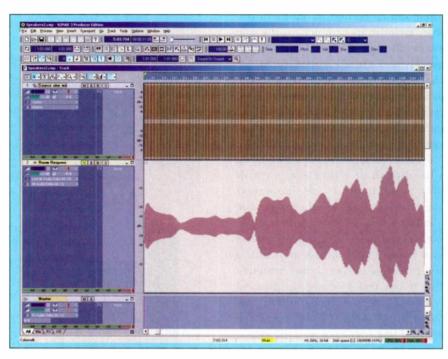
FIG. 3: Any DAW program can measure a room's response with no special test gear. This Sonar project plays 260 sine waves in 1 Hz increments from 40 to 300 Hz. Each tone on the first track sustains for one second and is played through the speakers. A microphone at the mix position records to the second track, which then reveals the room's response at each frequency.

approach is that matching bass levels with a commercial CD works only if both songs are in exactly the same key. Let's say your song is in the key of E, and your room has a response similar to the one shown in Fig. 1. Whether the bass is playing a low E or the octave above, either the fundamental frequency or the all-important second harmonic will align with the deep null at 82 Hz, making the bass seem very thin even though it really isn't. But if the reference song is in the key of A, either the low A or the octave above will align with the enormous peak in the response. With those two particular keys, at least, trying to make a well-balanced mix by matching bass levels is doomed to fail.

I visit a lot of audio newsgroups and Web forums, and participants often ask if they should buy a subwoofer to improve their ability to mix. Although a subwoofer can help compensate for inadequate loudspeakers, it will not solve the problem of an irregular response caused by acoustic interference. Often a subwoofer just compounds and hides the problem. In truth, even if your
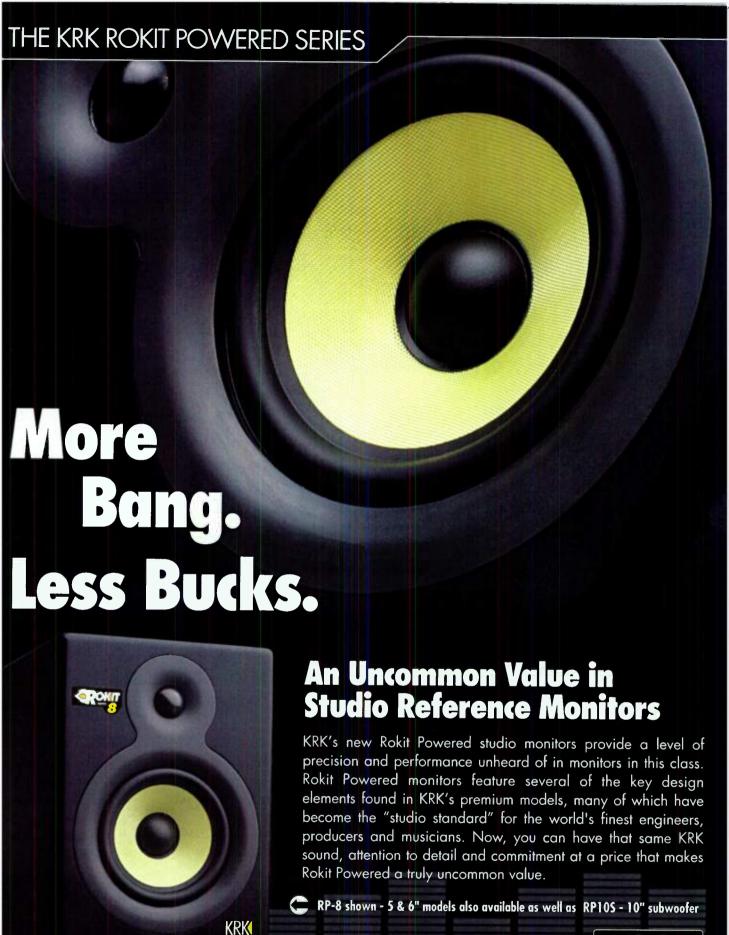
monitor speakers cost as much as your house, the response you hear will still vary by 30 dB or more in a typical small, untreated room.

THINK DIFFERENT

In this article, I explained some of the common acoustic problems facing owners of personal studios. I also showed why conventional thinking doesn't always apply to the small rooms often used for recording. If you'd like to learn more, I created a short video clip that illustrates some of the concepts discussed here (www.realtraps.com/videos.htm).

Understanding small-room acoustics is not especially complicated. All you need to know is that virtually all problems are caused by waves reflecting off of a room's boundaries. Armed with that knowledge, you are on your way to solving your studio's acoustic shortcomings.

Ethan Winer *is based in Connecticut, where he and partner Doug Ferrara design bass traps and other acoustic treatment for Real-Traps. Visit them at www.realtraps.com.*

Mixing Strategies

of the Pros

ixing a song is like driving on a very busy and confusing freeway. There are lots of choices and decisions to make, and an incorrect one will send you off in the wrong direction. You're surrounded by bad drivers (unfocused producers and egocentric musicians), who can definitely delay your arrival. Bad monitors are like potholes, and distractions abound, all threatening to destroy your concentration and set back your timetable. Misleading signs are everywhere, and a moment's hesitation can lead to a nasty accident. Getting to your destination will require a good road map; fast, accurate decisions; and a will to win.
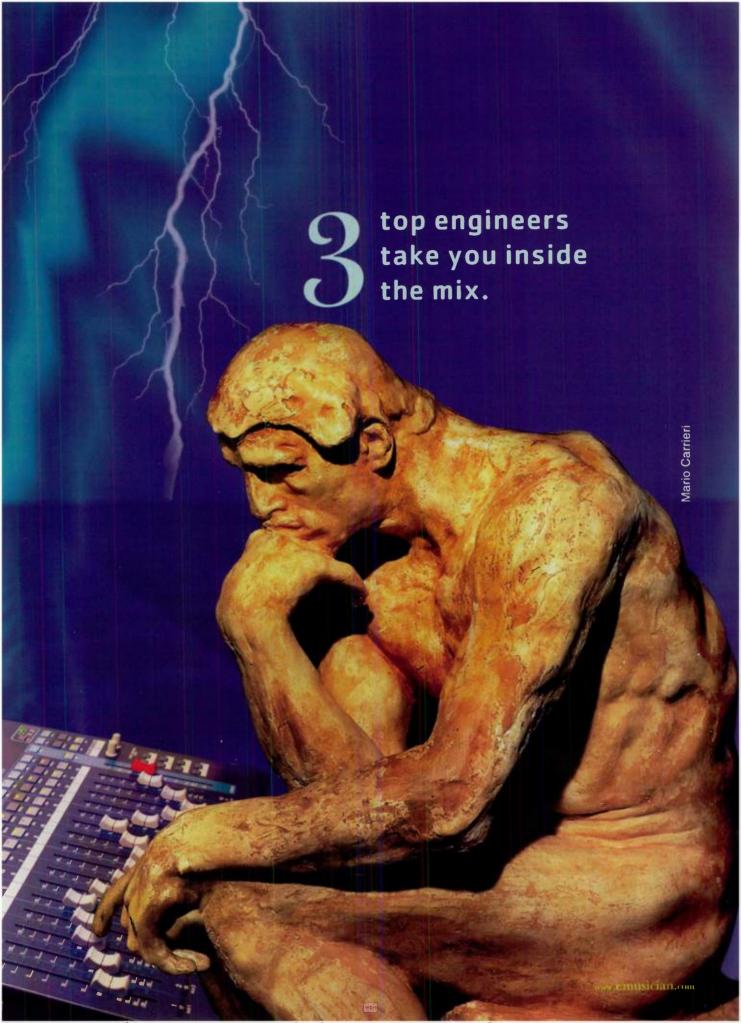
So how do the seasoned vets of the mixing world survive this chaos day in and day out? I corralled three of the best and asked them. Of course they have great equipment, but take it from me: if Roger Nichols, Tim Palmer, or Dave Pensado were working in your home studio, they would still get a great mix. Using their ideas will get you where you're going much faster, and the ride will be a whole lot smoother.

Tim Palmer is a modern-day scion of the British sound who works on, as he says, "everything from Goth and metal to pop." His extensive credits include U2, Tears for Fears, Pearl Jam, Ozzy Osbourne, Faith Hill, and David Bowie, among many others. When we spoke, he'd just finished mixing a live album for singer-songwriter Jason Mraz and a project for the subversively glam Kill Hannah, and was packing for a trip to Finland to work with HiM.

The perennially busy Dave "Hard Drive" Pensado is continually in demand by R&B and pop royalty. If you watch music videos or listen to the radio, you've likely heard some of his chart-topping mixes such as Pink's "Get the Party Started," Christina Aguilera's "Beautiful" and "Dirrty," and the diva rendition of "Lady Marmalade" from the *Moulin Rouge* soundtrack, which featured Christina Aguilera, Pink, Mya, and Lil' Kim.

By Maureen Droney

3 top engineers take you inside the mix.

Mario Carrieri

MIXING STRATEGIES OF THE PROS

Pragmatic and outspoken, Roger Nichols is an iconoclastic Grammy winner who's legendary for his work with the pristine madness of Steely Dan (13 albums) as well as John Denver (27 albums), Rickie Lee Jones, Take 6, Béla Fleck and the Flecktones, and Roseanne Cash, among many others. On the day I visited him in the studio, he was completing mixes for a Rogers and Hammerstein tribute album that featured performances by such artists as Phoebe Snow and Toots Thielman.

These multiplatinum mixers have very different styles, but they share a lot of the same philosophies. I found out how they approach their mixes, and the mental processes that enable them to deal with all the inherent obstacles.

HAVE A PLAN, STAN

All three mixers were emphatic in stating that diving blindly into a mix, trusting that the muse will lead you to inspiration, is a recipe for disaster. Before you begin, they agreed, you must take the time to figure out what it is that makes the song work best.

"Maybe it's the vocal, maybe it's the lyrics, maybe it's the groove or a hook," says Palmer. "If it's the lyrics, you have to think from a vocal perspective: is the music working to help the story to come across? In another song, the key thing might be that amazing and memorable guitar lick. Or, sometimes a song has a great groove. Then you've got to make the drums and bass really solid so they drive the whole thing along. It's your job to figure out what that special something is, and then carve your mix to bring it out. You must play to the strengths of the recorded material."

"What's important is that you have a concrete vision of where you want to go," Pensado agrees. "Then you'll find a way to get there. One of the main differences between me and a lot of the producers I work with is that I have the skill to do the mix in a few hours. They

might be able to do a similar mix, but it would take them at least a week. I think the lesson there is to mix for the forest and not the trees. When you start mixing for the trees, you paint yourself into a lot of corners that it's difficult to get out of. You want to keep yourself on line with the vision you started out with.

"Here's something to think about. When the old radio shows of the '40s wanted to create the sound of horse's hooves, they used a couple of coconut shells. A real horse walking through the recording studio wouldn't have sounded as much like a horse to the audience as the sounds that they created. That tells us that the original sound you're given is not as important as the image you want to convey in the mix.

"For me, and I think for a lot of mixers, it's very visual: you tend to want to see the horse, or, in the case of a great rock mix, the color of the guitar someone is playing. It actually becomes like a painting. Some of the early Nirvana mixes, for example, were very impressionistic. There are impressionistic mixes, abstract mixes, realistic mixes, paint by numbers mixes, where you just fill in the colors, and, of course, [laughs] Jackson Pollack mixes."

ROUGH STUFF

Rough mixes are what you throw down at the end of a session to give yourself a

working road map. However, they often take on a life of their own. Everyone has a horror story about the artist (ask Bruce Springsteen's engineer about the trauma of *Nebraska*) or A&R person who got attached to a seriously flawed rough mix and could never be convinced that there might be a better way to go. In those cases, trying a different direction is just a waste of time. But most people want to be open to creative possibilities and to the hope that the mix will take their song to a whole new level. So, isn't it best to start with a completely clean slate? Well, no, not according to these pros.

"It's very helpful to listen to the rough mixes before beginning," says Palmer. "You can definitely get off on the wrong foot if you go straight in and start pushing up faders without knowing what your goal is. If you have engineering ears, when you push the bass drum up you can tend to think, 'That bass drum is dull, I need to do this and that to it.' But maybe, to make the song work well, the drums need to be dull and not too defined. Here, if you didn't work out what was important before you started, you can make all the wrong decisions. You get to the end and realize that you've wasted your time. The song was much better with the low-fi drums."

"In a rough mix I'm listening for the

elements," says Pensado. "The most obvious question would be, Are there a lot of effects like reverb and delay on the instruments, or was the mix more up-front, dry, and in-your-face? Where is the vocal relative to the track? Because if the A&R guy has been listening to a loud-ass vocal for six months and you put it 'right,' you could have some problems. The other thing I listen for is—I hate to use the word, but—vibe. When you sit down at a console you're sitting down at a blank canvas, and the first stroke you put on that canvas doesn't come easy. The first thing you sit down to do is the hardest on any creative project. The rough mix gives you a starting spot."

Nichols also finds roughs useful, but for different reasons. "I think the whole process of making a record simply revolves around listening and paying attention," he says. "That may sound obvious, but in my opinion, a lot of people aren't doing it. For example, I get stuff to mix that other people have recorded where, especially if it's a Pro Tools session, they automatically put a limiter on every channel. Just because they can, or because they think they should. It doesn't matter what it is. It could be an empty channel, but it's got a limiter on it!

"Hopefully they recorded in Pro Tools and they can send me the session file so I can take the compressors off and start over. If I have the rough mixes, I'll listen to what they did first to get the concept of what they were trying to do overall. Did they like the vocals or the percussion loud? Then I'll start over, without their compression, to get to that platform of the loud vocal or percussion, or whatever it was that defined the whole feel of the song."

WHAT'S UP FIRST?

Whether you're laying out a 100-input console or mixing from a computer-based workstation, you've got to begin

with some piece of the song's instrumentation. Some engineers will throw up all the faders right off the bat for a quick reference balance. Traditionally, many other people begin with drums. Pensado tries to let the song determine his first moves.

"For some songs you might start with the rhythm track, others with vocals, some with just the kick drum," he explains. "A good friend of mine who's a mixer always starts with vocals. Sometimes I do too, especially if it's a group like Take 6 or Boyz II Men that's known for their vocals. I'll bring the music in around them. Something to keep in mind is that, unless you have an unlimited amount of time—which you never do—with a heavily laden vocal track you don't want to be working on your vocals at midnight. If I only have one day to do the mix, and the vocals are important, I'll get to the vocals early on, if not first. Then, when it's 11 o'clock at night, and I'm working more on instinct than creativity, I have confidence that my vocals are sounding right."

Nichols takes a holistic approach. "I've seen people start a mix by pushing up the kick drum before they've listened to the song," he says. "They've never heard the song, but they're trying to get a good sound on the kick. Then they work on the snare, then the bass. When they finally turn up all the instruments, it doesn't fit. Then they have to mess with this, mess with that, and they end up wasting a lot of time.

"The first thing I do is turn up all the faders and get a feel for what the song is and what the mood is supposed to be. Then I'll get a rough sort of balance between the instruments—this is how loud the strings should be, and the piano and the vocal. Then, if I hear that the vocal kind of sounds muddy against something else, I'll start equalizing a little bit, and fixing things so they meld together. Then I'll go back and listen to the drums by themselves, and the piano by itself, and refine that. I'll go around a couple of times in a big circle and then, usually, it's close to done, except for vocal rides."

EVERYTHING IN ITS PLACE

Mix engineers must routinely deal with multitudes of tracks, lots of good parts, a few bad ones, and often a band full of

Tim Palmer has mixed for a range of artists from David Bowie to U2 to Ozzy Osbourne to Faith Hill.

MAUREEN DRONEY

musicians who all want their individual parts turned louder. But beyond just finding the optimal levels, the engineer must figure out the best stereo placement, fitting each piece into the overall puzzle. The task is complicated by the fact that vocals, keyboards, and guitars often use a lot of the same frequencies to assert their presence. So when you're facing that daunting wall of midrange, skillful placement along with judicious EQ and filtering will really pay off.

"Everything can't be in the middle," says Nichols, a survivor of major track wars. "Unless you want to make a mono record, which is okay. But that's another challenge. If there are two things going on during the song—say, both a guitar and piano play throughout—then my first inclination is that maybe the piano should be a little bit on one side. Even though it's in stereo. So I'd cheat it over to one side, and I'd cheat the acoustic guitar over to the other side, so you can sort of hear what they're doing all the way through. They're the bed you're going to use.

"Then I'll probably put the bass in the middle. But not 100 percent of the time! Sometimes whatever the bass is doing gets muddy because of what the piano is doing. Then, maybe the piano should be a little to the right, and maybe the bass should be off center just a little bit. If that's a mistake, you put it back in the middle. You're always trying little refinements of placement to ensure that everything adds to the song instead of subtracting from it."

"It's not just about creating a nice balance and making everything sound good," says Palmer. "Maybe some things shouldn't sound so good; maybe something should be way out of balance. When you're first learning, because you have pride in your work, you tend to think of every sound as an individual item that should sound great. But as you move along through the years, you realize that sometimes, for one thing to sound good, something else may have to suffer."

So too much use of the solo button can steer you wrong? "That's right," Palmer says. "You have to look at the overall picture. It's not always about perfection. Sometimes you can take a part out of context, and it's not so good, but within context it's amazing.

"You listen to a sound and think 'That bass playing is rough, I don't know about that.' Then you put it with the drummer and it sounds great. Here's your plan helping again. Because if you go in and start fixing that bass up, putting it in time with the bass drum and lining it all up, you may have just ruined the whole groove."

IN AND OUT OF TUNE

That leads us to a theory put forth by Palmer. On the great old records we all seem to love, the bands were tuning mostly by ear, especially during the basic tracks. Everyone may have started out referenced to a tuner, but as the overdubs progressed, things shifted back and forth a bit. Tuning was a bit more about taste than exact science.

"Think of the perfect tuning in a song as a thin, straight line," he says. "In those days, after the basic track was laid down and a few overdubs were added, some of the tuning may actually had drifted, but not to the detriment of the sound. In reality, some of the parts may be a little sharp (above the line of perfect tuning) and some players may be a little flat (below the line of the perfect pitch). The tuning line is now a lot wider and thicker, and so is the sound!

"When the vocalist went to sing, he obviously aimed for the center of the pitch, but had a lot of space either way. He could go a little sharp with emotion, or flat a touch, and it still sounded good; it could actually add to the performance. Now, with each overdub everybody retunes. You get an overall pitch 'line' that's very thin, and that doesn't give much room for expression. When the singer performs, it's not as much fun! If he goes a little sharp, it

TIPS FOR BETTER MIXING

• Before you start moving faders and twisting knobs, have a clear vision of what you want the mix to sound like when it's finished, and stick to it.

• There's no hard-and-fast rule as to whether to build your mix one element at a time or start with all tracks up. Be flexible, and let the needs of the song determine your approach. Avoid doing detailed tweaking on individual elements until you have a good idea of what the rest of the tracks sound like.

• Don't dehumanize your tracks with overly aggressive pitch and time correction. Remember that subtle irregularities in rhythm and pitch can often make the overall song sound better.

• When equalizing or adding an effect to a particular element, always take into account how that change will affect the mix as a whole.

• Too much processing can negatively affect the overall sonic quality, so be judicious. When equalizing, subtract rather than add when possible.

• Preserve your dynamics. Don't go overboard with compression. When working with programmed tracks, add in enough dynamics to make the song exciting.

• Monitor on a variety of speakers at various levels, and try to check your mix in different acoustic environments.

• Don't stay on one mix for too long. Take breaks, and even switch to a different song for awhile if possible. When you come back to a song after a prolonged break, listen through it and create a list of the changes you want to make.

• Don't feel constrained by convention. Experiment when you can, and if it sounds good, do it!

doesn't sound good anymore. So what do you do? You tune him or her as well. They join the line and now the whole thing sounds small!

Pensado thinks that although pitch correction can be helpful, it's often overused and can detract from the feel of a song. "A great example is blues," he points out. "When you go up a minor third, sometimes it sounds better when you don't quite get there. Violin players, when they're going up the scale, they play a different interval between B and C then when they're coming down the scale. So the human ear readily accepts imperfections in pitch, and it's a very, very personal thing how you interpret it." He counsels a selective approach to pitch correction, rather than just strapping it across an entire track: "Find those notes that just could never be resung [or replayed] again, correct those, and move on."

Rhythmic feel can also be damaged by too much correction, says Palmer. "If you chop the whole thing up into enough pieces so it's all bang on a grid, the ear doesn't hear any natural flamming of instruments any more. In many cases a bit of that flamming makes the music sound bigger and more appealing. With a great band—Led Zeppelin or whoever—you have the drummer
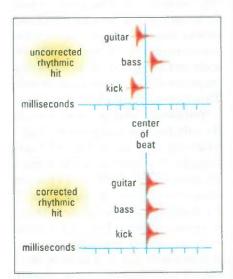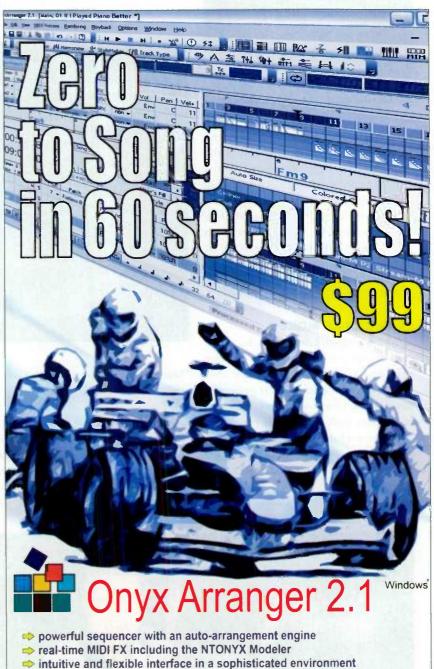

FIG. 1: Tim Palmer observes that too much rhythmic correction can eliminate the natural flamming produced when an ensemble hits a simultaneous beat. Slicing it up and putting it "on the grid" can actually make it sound thinner.

who might play really laid back, the bass player who's slightly ahead of the beat, and the guitar player who's slightly behind. When they hit that big chord together, you've got 'one, two, three,' all hitting your ear at slightly different times. It sounds huge, with just enough natural offset [see **Fig. 1**]. Think of an orchestra; one of the main reasons it sounds good is because they are a bit out of time and out of tune!"

LESS CAN BE MORE

Most recordists know that the more processing you use, the more phase shift and artifacts you introduce into your signal path, and this is generally detrimental to a good sound. Yet it's so tempting, and so easy, to reach for those powerful knobs and that brand new plug-in. Nichols deals with this conundrum by taking a subtractive approach.

"Say I'm working on a vocal," he says.

Q&A: USING CDS FOR REFERENCE

Do you reference your mixes against finished CDs during the mixing process?

Palmer: If I'm not familiar with a room, yes, I'll listen to some other things I've done."

Nichols: If I am in a studio I have not mixed in before, I will play back something I did somewhere else that came out well so I can hear what the speakers are doing in the room. Once I have it scoped out I use what I heard to modify the mixes so they match when I leave. I do not play the CD any more during the mixing.

Part of it is the Meyer HD-1 speakers I use. I either rent or bring my own. I have used the HD-1s on everything I've done since 1989. Most studios where I have used them now own a pair so I don't have to cart mine.

Pensado: I like to reference to CDs, and I like for my clients to hear it. It gives people in the room who aren't accustomed to my monitors a frame of reference. Actually, I've been told that I listen to way more music during my mix process than most engineers. I always have a music television station on and a couple of CDs in the player. I'll just randomly hit it. If what's in there sounds better than what I'm doing, I keep working. If it doesn't, I'll print!

"Mainly I want it to sound the way the singer really sounds in the room. The way I do that—instead of trying to add something to make it better—is to figure out what's bad and get rid of it. If you start out saying, 'This needs to be a little brighter, and that needs to have more bottom end,' pretty soon you've got EQ adding 57 little things, when maybe all it really needed was for you to find the bad part and remove it. All of a sudden it's better because you took out the sound of the room where the vocal was done that wasn't a very good room. Or you took out some woofy part of the piano, where you had to put the lid on a short stick with a blanket over it. If you look for the bad thing and remove it, you're about 60 percent there. *Then* you can say, 'Oh, a little brighter here . . .'"

Palmer agrees. "It's very easy to over EQ," he says, "it can be better to look for frequencies that are unnecessary and remove them. If I'm looking for a frequency to remove, I'll boost the EQ radically and sweep it until the offending frequency sticks out like a sore thumb. Then I know exactly where to cut."
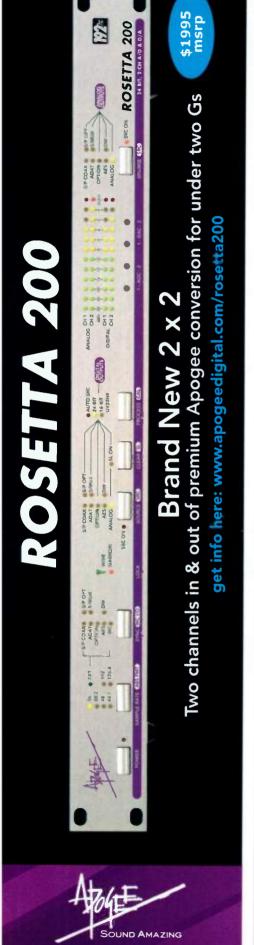
"You should never put something on the mix because you perceive that's what you're supposed to do," says Pensado. "For instance, If I'm doing a hip-hop mix I probably won't put any compression across the stereo mix. If I'm doing a 'Lady Marmalade,' I've got it crunched. Why? I knew it was going to be a pop song getting a lot of airplay on a wide variety of stations, and I wanted my compressor to be controlling things more than the radio's.

"But say on 'Beautiful,' by Christina Aguilera, there's nothing on the stereo bus—no EQ or compression. We wanted an old-fashioned sound. There are some

Veteran engineer Roger Nichols's client list runs the gamut from Steely Dan to Take 6 to Béla Fleck and the Flecktones.

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MIXING STRATEGIES OF THE PROS

songs I mix, say on Pink's 'Get the Party Started,' where I wanted you to hear the mix. I wanted it to be part of what was hooky about the song. But with 'Beautiful,' if anything at all made you think of anything besides Christina's voice, it was a failure. So I 'anti-mixed' it. I really worked on subtracting my personality from the mix."

DON'T DISTURB THE GROOVE

"Groove is a very personal experience," says Palmer. "It's really the amount of 'out of timeness' that you personally enjoy and accept. Sometimes I'm working with an artist who will constantly want to replay a part because they feel it is out of time, while I'll be telling them it sounds great. Their personal reference point is just different.

"It's important to keep as much of that feel as you can. If I have to move a few parts around during a mix I will move them by ear. Remember, it's important to listen to music rather than look at it. A lot of people now just look at the screen and make decisions based on that. But if you listen, it might sound good even if it looks wrong. Using the screen can, of course, be helpful at times, especially if you're wondering why something's not working. It's convenient to have a look around and see what's going on. But if you're finding problems before you've heard them, that's when you get into trouble."

"To me, the groove is the mood of everything taken as a whole," says Nichols. "The internal rhythms, the delay on the guitar, the little space between the vocal and the reverb, the way the kick and bass connect, the sustain of the acoustic piano, the bounce of the congas. It's all the little nuances that make everything work together.

"There doesn't have to be a lot of stuff on the record, it just has to feel right. I have gone into the studio intent on doing overdubs on a song, and when I put up the mix and listened, it didn't need anything else. We'd try anyway, but at the end of the day, we didn't use anything we added. The song was done, but we hadn't realized it."

THE EFFECT OF EFFECTS

"I like to find one reverb and use it as a main reverb for the whole song, sending everything to it a little bit," says Nichols. "Then for the vocal, I'll use three or four different reverbs, splitting out the vocal to different tracks [or channels]. When the vocal gets louder, the quality of the reverb changes because it's sending to a different unit. When the vocal is soft, maybe the reverb has a long decay time so it's nice and moody and has all this big ambience. That long, three-second reverb doesn't work when the person's screaming into it; it will be two choruses before the thing finally decays.

"Taking the vocal apart and splitting it out to different tracks allows you to give the loud parts of the vocal a completely different reverb send and completely different compressor or EQ settings than the soft parts [see Fig. 2].
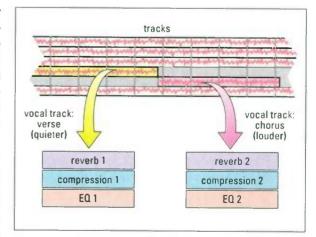

FIG. 2: One way Roger Nichols adds variety to a vocal track is to split the various sections of the song onto separate tracks and then use different reverbs and even different EQ and compression on each.

Of course, those things are easy to do using Pro Tools. You can clone tracks so you've got two vocals or two snare drums—one for the cross stick, another when he's playing the snare—and make them different. Doing those things will really make the mix easier."

Pensado says, "I look at reverb in two ways. First, it's a front to rear panner. If you want something to go to the back of the mix, put reverb on it. This is the most fundamental psychoacoustic trick we have. Of course, sometimes I use reverb just because it sounds good. Who cares if it sounds like it's in a sewer pipe, as long as it's a great effect?

"I don't like to pan my effects returns hard left and right. I'd rather get identical units or plug-ins, then pan one hard left (both returns) and the other hard right (both returns). Select the same, or similar, programs on each unit and tweak them so they're slightly different.

"I like to use 30 to 40 delays, from a 128th note to a half note, panned all over the place. I'll intentionally make some of the delays out of time, creating what I call asymmetrical delays. Just move them until they sound good. Plug-ins are perfect for this, because some have 6 to 12 delays built in."

Some engineers like to precisely time their delays to multiples or fractions of a song's tempo. Palmer says, "I do that if I'm going for a specific effect, but often I think delays are better when they're out of time and create a bit of rubbing. Especially with music being so precise these days, it can be good to loosen things up a bit and relax them. Maybe a delay that's bang on will just disappear into the snare drum sound, and you'll find yourself turning it up louder and louder to hear it! If it's a little bit out of time, you'll get a bit of rub and you might hear it more clearly.

"Of course," he notes, "all this is specific to each mix and artist. When I work with U2 they don't have these tun-ing or timing issues. They play as a band and don't need to be loosened up. They don't cut things up in Pro Tools and they don't overtune. With U2, maybe I want the delays sitting just right in the pocket because there's already enough movement within their instruments."

THE CASE FOR DYNAMICS

"The current trend for overcompression doesn't do it for me," says Palmer. "It makes things sound very, very loud, but also very undynamic and small. Lately, even the mastering engineers seem to be bored with the cheap thrill of overcompression! If you're mastering specifically for the radio, you obviously have to make sure your songs aren't going to dip compared to your competition. But a commercial CD for home use should have some depth. Lately every CD is prepared as though it's on the radio already. I want my CDs to sound more dynamic and have more variation in sound and level."

"Usually by the second verse, if you've got a drum machine at pretty constant volume or parts that aren't dynamic, you're not noticing them anymore," says Pensado. "But if you just kind of yank a drum up here, or something up there, your ear finds it and you'll remember it for another 32 bars or so. Something I've learned from the visual analogy is that it's okay to make things loud for one or two bars, then tuck them back where they should be. It's okay to take the kick drum and at the beginning of every eight bars turn it up 8 dB. Make it stupid loud. The engineers might say, 'Ooh, did you hear that?', but 99.9 percent of the people who buy the record are going to go, 'That's cool!'"

"It takes me about an hour, but sometimes I'll put every kick and snare at the level I feel it would have been if it was played live [see Fig. 3]. And almost invariably, you feel like the drums should come down a bit at the second verse. And then there's the kind of bridge that should subtract energy from a song, so that when you hear that last chorus come in on the radio, you're going to drive straight to Tower to buy the record."

THE FINAL FRONTIER

Should you find one set of monitors you like and stick with it, or listen on every different speaker you can find?
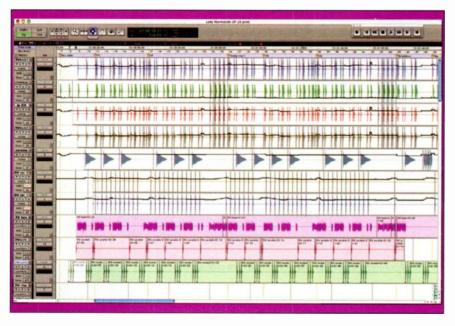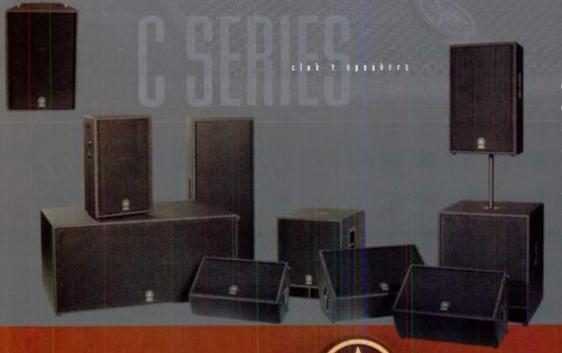

FIG. 3: This Pro Tools screen shot from Dave Pensado's mix of "Lady Marmalade" (performed by Christina Aguilera, Pink, Mya, and Lil' Kim on the *Moulin Rouge* soundtrack) demonstrates how you can add dynamics into programmed music. Notice how the volume levels (as depicted by the black lines) have been altered for the song's various sections.

Sanity is elusive in the domain of psychoacoustics, especially when you've been at the mix for a while.

"I switch between several different speakers for different purposes," says Palmer. "If you're trying to get a tight sound with the bass and the bass drum knitting together, you might put it on the big speakers where you'll get the best bass response, then crank it up a bit to make sure it's solid. When you're balancing vocal levels and you're thinking about what the emotions of the music are and if they speak well, you're probably going to be listening more quietly. At that point you don't need to be blown away by fidelity, you're just listening for a balance you think is correct. I'll usually do that on a Yamaha NS10 or an Auratone—whatever I have that's fairly quiet. For fidelity I'd go more for my Genelecs or the big speakers. And of course my car is a good point of reference for me because I listen to a lot of music there."

"If everybody in the world had the same speakers and power amps I could do a mix in five minutes," laughs Pensado. "To be really good at understanding what happens to your mix at different volumes and on different speakers, you need to listen in a variety of environments. I use four sets. I love the old Yamaha NS10s, I've got a set of Augsbergers that are dual 15 TADs with a TAD driver, I've got a set of old Auratones that are on their last legs, and I use the little mono 3-inch speaker that comes with the ½-inch Studer machine. Even after all that, I'll try to listen in the car, or go to the lounge and listen on a boom box. I also have a little college-radio-station transmitter I've put together so I can send FM out to the radio in my car from the control room—under FCC limits, I want to point out. I also really like it when we have the budget for clients to take the mix to their home studio to check it out."

Okay, lots of different speakers. But what exactly are you listening for? "A lot of the music I do requires the kick drum to be as important as the guitars are in a Led Zeppelin song," says Pensado. "If you're listening on the big speakers, you can be thinking 'Man, it's Grammy time!' then you go down to the Auratones and there's nothing there. That tells you the frequencies below 100 cycles are right where they need to be, but the frequencies above there are pretty much nonexistent. This gets into the area of psychoacoustics. You're not ever going to get the low end from the kick drum to come out of the set of Auratones. But what you can get are the frequencies that make you think you're hearing the lower frequencies. That could be anything from 200 to 3K.

"A lot of times you'll add top end to a kick just so your ear can find it. You're not making it louder or fatter, but the perception is that the kick drum got louder. Actually, it just gave your ear the opportunity to find it. Think of it this way: if you have a power amp on your system, it takes several hundred watts to move the low end, but only 20 or 30 watts to move the high end. Your ear hears high frequencies much more efficiently than

low frequencies; what you're doing is playing on the ear's efficiency to find the low end.

"Here's an example. Put on the mix and listen to the level of the vocals relative to the music. Pick a song that has a lot of full-range frequency, not one that's thin on the bottom, and when you go to the smaller speakers it's going to sound like the vocals are too loud. When you go to the big speakers they'll sound like they're not loud enough. What's right is probably to have your vocals a little loud on the smaller system and a little—just a little—more difficult to hear on the bigger system.

"As an engineer," continues Pensado, "if I make the decision that most of the people who will buy this song are going to be blasting this as loud as they can in a car, I won't pan things quite as wide. And I'll use less reverb, because they're going to be getting the reverb from their environment. It's like when you go to a big dance club where the music is reverberating off the walls; if you have too much reverb on your mix you're in serious trouble. That's why we do dance mixes pretty dry, except for maybe an effect on something you want to sound big and wide and nebulous."

Q&A: LEAVING ROOM FOR MASTERING

How much room should you leave for the mastering engineer to add the finishing touches?

Palmer: I do like to leave some room for mastering. I don't like to kill the mix by adding too much compression too soon. I'd rather keep an album sounding as dynamic as possible. If more compression is needed for the radio, the mastering guy can squish it a little further.

Pensado: I'm fortunate that I work with the same few mastering engineers I trust, and I prefer not to put too much compression on my stereo bus if I know they are going to work on it. These guys are great; they can do it as good or better, and I've saved myself going through those electronics, which they're going to go through anyway.

Nichols: Because I actually have a mastering business and have been mastering since the mid-'70s at ABC/Dunhill, I know when to quit! The monitors and the monitoring environment are the important things. If you are not exactly sure about how things are going to translate, then give yourself a little more room for mastering. If the monitors and environment are perfect, you can cut it a lot closer.

You can also leave yourself flexibility with alternate mixes. In Nashville, they usually print about 20 versions of each mix. Vocal +1 [dB], vocal –1, backgrounds +1, vocal 0, backgrounds –1—all the permutations you can imagine. When the songs get to mastering, if the vocal is too low in the chorus, they just edit in the version with the vocal up louder.

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Other recorders available from Korg: D1600mkII, D1200mkII, PXR4, CR-4.

ARTIST RELATIONS

Mix engineers face more than just technical challenges; they also have to learn to deal successfully with their clients. "Sometimes a band wants something you think is crazy, just nuts," says Palmer. "Then you try it, and it's pretty good! You can't let your ego get in the way. On the other hand, I recently met with a major artist who wanted to do things in a way I didn't think was right for the project. I thought about compromising and doing things his way. But ultimately I realized I would be very unhappy. I told him I wasn't the right person and left. If you don't enjoy your job, you're not going to do it very well."

"One of the neat things about success," says Pensado, "is that the people who are telling you what to do have probably been successful. That helps with trust, and that bond you definitely need to have between an engineer and a client. But I believe conflict is always a necessary ingredient in creativity. *[Laughs.]* Show me a totally happy environment and I'll show you some crap coming out of it! It's not coincidence that Lennon and McCartney or Jagger and Richards weren't speaking half the time. Or that Van Gogh cut off his ear!

"I like it when a client disagrees with me and can back it up," continues Pensado. It makes you do things and think about things differently than you have in the past. Most people you work with have great suggestions; sometimes they don't know how to articulate them. I'm thinking Fahrenheit and they're describing in centigrade. If I can get the formula that can translate that, we're doing some great stuff.

"I think the ideal balance of working with a client is to give them what they want, but 15 or 20 percent more. With some clients, maybe only 10 percent. Then they're going to feel, and

PENSADO PICKS PLUG-INS OVER HARDWARE

Dave Pensado has had plenty of experience working with all types of processors, including many classic analog units. But surprisingly, if you gave him a choice between using the original hardware or the software equivalents, he would choose the latter in most cases. Why? "What these plug-ins do is give you the opportunity to take a classic sound and tailor it for today," he says. "The concept is to walk a very fine line between giving listeners something that reminds them of something classic, but that also sounds new. So if you're into retro everything, just use the originals, you'll be fine. But if you want that plus more, the plug-ins give it to you. It's just a more modern sound."

One of the plug-ins that Pensado likes is the McDSP Compressor Bank CB2, a TDM plug-in that he uses to emulate the sound of a classic Fairchild compressor. Among the artists whose vocal tracks he's used this plug-in on are Brian McKnight and Beyoncé Knowles. "What I like about it over the analog Fairchild is that I have more flexibility over the controls. I can control attack and release, I can control the amount of compression [see **Fig. A**]. With the real Fairchild, I can't get in the cracks enough to do what I want with Brian's and Beyoncé's vocals. Some of the older effects are almost a one-size-fits-all type of approach. But the CB2 plug-in allows me to tailor it for each song and each vocal."

Another example comes from Eventide. Pensado prefers the plug-in version of that company's Harmonizer H910 (one of the components in Eventide's Clockworks Legacy Bundle for TDM) to the original hardware unit. "The Harmonizer plug-in is better because it's stable," Pensado says. "A lot of the old first-generation or second-generation digital equipment had stability problems. This Harmonizer plug-in gives you that classic guitar sound, that classic instrument sound, without the stability problems. It's just an all-around improved, but still sonically identical, version of the original."

He says that when he's done blind listening tests to compare the plug-ins he uses and the hardware

FIG. A: For the vocals on Brian McKnight's recent CD *U Turn* (Motown, 2003), Dave Pensado used this Fairchild compressor-emulation setting on the McDSP CB2 compressor. He used a similar setting for Beyoncé Knowles's vocals on the Destiny's Child album *Survivor* (Columbia, 2001).

processors they emulate, the plug-ins usually win. "When you don't know which is which, you'll probably pick the plug-in 70 percent of the time," he says, "if you're being honest with yourself." (For more of Pensado's observations about plug-ins, and to see more of the settings he uses, refer to "Dave Pensado's Plug-In Secrets" at www.emusician.com. You can also contact Pensado directly by e-mail at fdpen@ix.netcom.com.)

—*Mike Levine*

rightly so, that when they're coming to you, they're getting new stuff. Not just what they want, but also a sound that's special and really works. It's a fine balance. You've got to understand your client psychologically to know just how far ahead of the curve you want to take them. Rarely is it more than 20 per cent. Give them more than that, and they'll leave studio thinking you're kind of cool and hip, but a week later you'll probably recall it and get all that stuff out of there. Eventually they're going to take you back in the direction of the rough mix because that's their comfort zone."

KNOW WHEN TO STOP

"I generally work from my instincts and have a feel for when the mix is right," says Palmer. "Some mixers will work for 12 hours, then pull the faders down and start again. I've never been like that, I would rather work fast and then

have another go later if I am not happy. Also, I find it helps to keep the hours sensible, and to attempt to keep some distance. Taking breaks instead of spending hours on end in the control room helps to keep perspective. Listening through a door or in your car is useful. I sometimes take a break and put the TV on, check out some MTV. *[Laughs.]* There's always the chance you might hear something you can use. But seriously, keeping objectivity is half the battle.

"I take a mix up to a point where I think it sounds pretty good, but I don't push it too far," Palmer continues. "Then I go home—hopefully not too late at night! In the morning, the first thing I do is listen to the mix and make my notes with a fresh brain and fresh ears. Morning is probably when I'm the most critical. I'll get a list—whether it be 2 or 20 things to do—and go straight to the studio and do those changes right off. Then I'm pretty much ready for the band to arrive.

"I don't want to be part of this factory-line sensibility that some modern-day mixers employ: 'Get it done; next!' Occasionally, though, it can turn

out really well to have to work fast. When you don't get too much time to overthink, you're working purely off your instincts, which can be good. On the Tin Machine records—that people either love or despise—I would start to mix and David Bowie would often say, 'I love the rough mix, don't spend too long and take the edge away.' I'd tell him there were a few things I could get better, so he'd give me one hour to mix. That was it ! You really had to work from your gut. That's all well and good, but when you're unhappy, and the budget doesn't allow for a second look it's very frustrating!"

"I like systems with total reset, so that when you open it up it's exactly where you left off," says Nichols. "Especially when you're working on a whole album. When you don't want to hear the first tune anymore you can save it and go to something fresh. You can get it 50 percent done, then go through the whole ten tunes and do the same, then go back. That saves you from getting tired. You don't want to spend four hours riding the vocal, then you ride something else and now the vocals are all too low.

"You bring the vocal up more, and some of the rides you did no longer mean anything. That's chasing your tail. A break and going to another song can really help. When are you done? When there's nothing left to do! If you can listen to the whole record, and not just focus on one thing, if nothing bothers you, and you can hear everything, then it's okay. That's it."

On the other hand, Pensado says, "I have a saying that you're never through with a mix, you just run out of time. I'm not sure that you're ever really done. I don't get to do a lot of recall mixes, but unlike a lot of engineers, I actually like doing them. Even when you've nailed a mix, when you hear it again you'll have other ideas. That's just a sign that you're growing. A mix is never really finished, but it is definitely possible to overmix something. There's a point in the process where another two hours can very easily make it worse. You have to look for those signs. You ask yourself, 'Was I

Q&A: DEVELOPING YOUR EARS

How did you develop the ability to know what a good mix should sound like?

Palmer: You're not likely to have it right out of the box. It's something that you develop over time, something that you definitely get better at. When I first started out as an assistant, I'd see people spending five or six hours equalizing bass drums and think, "How can they tell anything after all that?" But the longer I've done it, the easier it's become. You have a lot more clarity because you know where you've made all your previous mistakes.

Nichols: I don't call myself a musician, but I'm a guitar owner, and a keyboard owner, and I can muddle through. That ability has helped me to know what I liked, and why. Knowing what to do came partly from listening to the balance of instruments and the timbre of their sounds. The other part of it is a feel thing, as basic as when you hear a

song and something in you goes "Wow, that's cool!" When I'm mixing, I'm always trying to create—besides a sound—something emotional. The way the bass sustains under a piano lick, or the way some other parts happen together . . . that's what gives the song its mood. To me, that's at least as important as getting an individual guitar to sound good.

Pensado: If you're a good mixer, you mostly have the canvas in your head when you start. It's just a question of visualizing the sound as you work. That sound comes from the records you heard as a child, the experiences in your life, all sorts of things that give you references. Most of us who do this rely heavily on those references. That's why it's so important to be open to new music, new things, and to force yourself to change your references. It's incredibly easy to have them stay solidified, and that's not good.

more excited about this an hour or two ago?' The minute you start feeling it's getting worse instead of better, you were done an hour ago and you should probably backtrack!"

PARTING SHOTS

Finding your own voice is one of the most important and rewarding things you can do, in life and in mixing. But in our increasingly homogenized world, it can be a risky business that requires hard work and a high level of self-awareness. On the other hand, great art, and lasting records, don't come without pushing the envelope.

"I think these days a lot of people are affected by marketing when it comes to how they're working," says Nichols. "It's about what equipment they're using, and what somebody else who's successful did with that equipment. They're not really paying attention to the specifics of what they themselves are recording or mixing.
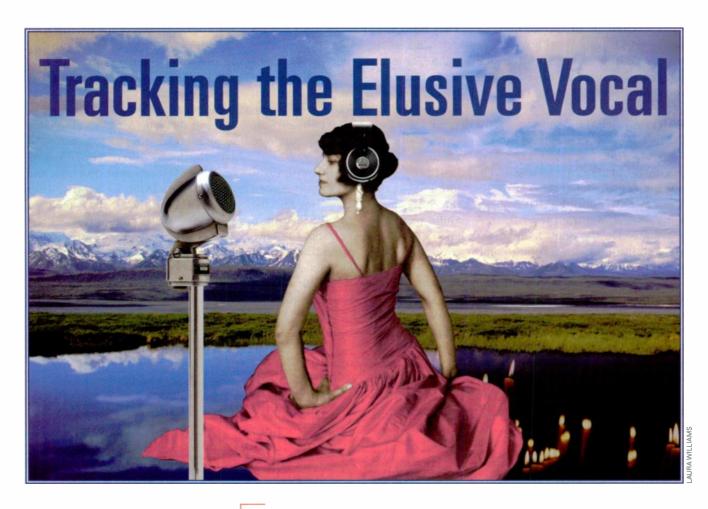
They think they have to use a certain drum machine, or a certain microphone, or a certain compressor that's set in a certain way because that's what they've heard they're supposed to do. To me, what's important is to experiment, some little bit, every single time so that you can come up with your own rules."

"If people in our industry thought more about the correlation between the visual elements of mixing and the audio part, we could take mixing to another level," says Pensado. "We have to step up the music to where a kid wants to buy our record as opposed to a piece of software or a PlayStation game or any of the other temptations that are out there for 20 or 40 bucks. Obviously we all would like to have better music, but those of us who do what I do can contribute a little bit by making the music we have better. That doesn't require replaying the parts, it requires understanding what

the original vision of the producer or the writer or the artist was and trying to piggyback your vision onto that rather than having a contradictory vision, or none."

Palmer states simply: "The song is, and will always be, king. It's funny that as we increase sampling rates and bit rates in the recording side of music, the public is moving the other way and downgrading from CD to MP3. They are showing us they really care about the songs, artists, and performances. That is not an excuse for poor production and mixing, but a reminder about what makes someone want to own a piece of music. Sometimes I feel we miss the point. Don't forget that the best cure for a bad mix is a great song!"

Maureen Droney, whose engineering credits include projects for Carlos Santana, George Benson, John Hiatt, Whitney Houston, and Aretha Franklin, among many others, is the Los Angeles editor for Mix.

Tracking the Elusive Vocal

Successful techniques for recording lead singers.

By Sean Carberry

I f I had a nickel for every time someone has asked me what's the best vocal mic to buy for *x* amount of money, I'd have one huge pile of nickels. I don't mean to imply that it's not an important question, because it is. But the only simple answer is "Whichever one sounds best," a reply most people don't find helpful.

The problem is that the only way to respond effectively is to ask more questions: What kind of voice are you recording? What style of music? What mic preamps and other gear do you have? What's the recording medium? What's the purpose of the recording (demo, fun, commerce)? What other microphones do you have?

These questions all lead up to a much larger one, the one that most people seeking opinions about vocal mics are really asking without realizing it: how do I record a great vocal track?

The choice of a microphone is just one part of the equation. It's an important part, naturally, but it's not the major determinant for recording a good vocal track. You might even say it's a red herring. Some of the greatest vocal recordings ever made were captured with microphones that many studio snobs would never consider using for the task.

My favorite vocals stand out in my mind because of the performances more than the actual sound. (Although great sound can elevate the quality of the track, it's ultimately just a vehicle.) All that a high-quality microphone guarantees you is a better shot at getting a great-sounding reproduction of what may or may not be a good performance. The old "garbage in, garbage out" cliché definitely applies.

So how do you record a great vocal track? It helps to have a great singer

Tracking the Elusive Vocal

singing a great song through great recording gear. Of those four elements, the easiest one to compromise on is the recording gear. I will discuss gear in this article, but I also want to talk about ways to use it to produce a better vocal performance. My focus is exclusively on the recording process. Mixing is a story for another time, especially considering that you can't do much about the performance—other than some pitch correction and editing—once you're at the mix phase.

ALL IN THE PHONES

You might be surprised to learn that the quality of the headphone mix can often determine whether a singer gives a good performance in the studio. Even top-notch, highly experienced vocalists need a good monitoring reference when tracking. Some singers are more fussy than others, but the more comfortable they are while singing, the better.

If a vocalist has to think about what he or she is hearing, that will surely have a negative effect on their performance. If the singer is doing anything other than singing, you've lost. Without having the proper monitoring, the elements of pitch, emotion, and energy will suffer, and self-consciousness and loss of focus will likely result.

So the first step is to determine what the singer needs in the mix. If you have 48 tracks of instruments, densely arranged with all kinds of potentially distracting sounds, it can be difficult to create a good cue mix. Sometimes the arrangement for the recording is more complicated than what the singer is used to. This is frequently the case with singer-songwriters who are venturing into the studio for the first time. They're often baffled by singing along with drums, bass, guitars, strings, layers of synths, and percussion. Remember, just because it's on the multitrack doesn't mean the singer needs to hear it.

When it comes to the headphone mix, less is more. You should strive to build a mix that features only the essentials, keeping the "sweetening" elements out unless the singer asks for them or they're necessary to elevate the

singer's emotional intensity. It's also vital to keep the mix clean enough to give the vocalist a solid pitch reference. A muddy headphone mix can blur the singer's sense of pitch. If a singer is having intonation trouble, I listen for elements that might be causing the pitch confusion. By pulling out an instrument or two, you can often clean up the mix and make life better for everyone.

Another important factor is how loud the singer's own voice will be relative to the rest of the mix. Whether due to vanity, hearing impairment, or just force of habit, many singers like their voices to be incredibly loud. Others prefer them lower. Regardless, the first thing to do is ask! Don't guess or do it the way the previous singer liked it. Ask what they want and give it to them. The same goes for reverb. Some singers want lots, others none, and others might want delay. Again, ask and make it so—it's always time well spent.

Some inexperienced singers have no idea what works for them in the headphones. Occasionally, I have to play games like varying the vocal level in their cans over the course of a take to see if the performance gets better one way or another. You need to do whatever it takes to get the singer singing and not thinking.

Another detail is the headphones themselves. Some singers carry their own or know from past experience what make and model they like. If possible, find out beforehand and get a set. At minimum, make sure you have a couple of different brands and models available at the session.

SETTING THE MOOD

Look for other ways to make the singer comfortable. A good place to start is with the vibe of the studio. Many recording spaces feel very sterile or clinical, and some singers are sensitive to that. Lighting, temperature, air quality, and decor can all affect a singer's state of mind. Although some can deal with any environment, others need candles, tapestries, aromatherapy, and pictures of their childhood home in the booth. You don't have to hire a feng shui consultant, but within reason, try to
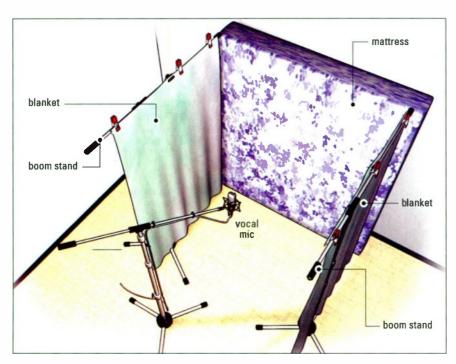

FIG. 1: You can construct a temporary vocal booth using a mattress, blankets, and boom stands.

In search of the perfect sound

accommodate a singer's requests. Believe me, it will pay off with fewer takes, fewer punches, and a more inspired performance.

Be wary of making the vocalist feel as though he or she is in a fishbowl. Although some singers don't care who is in the control room or peering through the glass, others get uptight.

I did a project a few years ago with a singer who was an old friend with little studio experience. She was nervous to begin with, and the presence of band members in the control room watching through the glass made her tense and caused a tiff. I cleared the room and spent a few minutes chatting with her to calm her down and take the pressure off. I lowered the lights and said I just wanted her to feel comfortable and sing her song on her own terms.

In the end we got a great vocal on tape, and it had little to do with the AKG C 12 in front of her; she was able to get in the mood and immerse herself in the song and the performance. When the rest of the band heard the playback, they were surprised by how emotive her performance was, and they realized how much their presence had intimidated her during the earlier takes.

I LIKE MIC

Mic selection and placement are obviously very important considerations. If you're fortunate enough to have access to an expensive large-diaphragm tube condenser mic—such as the venerable Neumann U 47, the silky AKG C 12, or the divine Telefunken Elam 251—you certainly have an advantage. But even a classic mic doesn't guarantee a great vocal track.

Anyone who has recorded more than one vocalist knows that no one mic is perfect for all vocal applications. One singer will sound great through a certain mic, while another sounds like a

cat caught in a blender. Why? Because all voices are different. There are certain truisms in the studio; particular mics tend to work well on female voices, while others tend to work well on deep male voices, and so forth. But given the time and resources, you should try a variety of combinations. You'll be surprised with the results.

Here are some questions you need to ask before choosing a mic. What overall sound do you want in the track—warm? clear? present? intimate? edgy? thin? Which qualities in the singer's voice do you want to accentuate, and which do you want to minimize? What is the singer's dynamic range? Does the singer have good mic technique, avoiding excessive plosives and either staying in one place so that the sound doesn't vary or moving back and forth to adjust for volume changes?

I want to dispel the myth that large-diaphragm condensers always make the best vocal mics. Sometimes they do, but often they don't. Small-diaphragm condensers and moving-coil and ribbon-dynamic mics can sound incredible on the right singer. In fact, many of the revered vocal performances of the 20th century were captured by ribbon mics. Keep an open mind, and trust your ears.

Many singers who have spent a lot of time in the studio know what mic (or mics) sounds best for them. It's a good idea to ask singers which mics they have used successfully in the past—it can save you a lot of trouble. Some singers will even ask ahead of time for a particular mic. Appreciate it when it happens, because you're likely dealing with a pro who has a good understanding of the recording process.

Otherwise, if you have a lot of experience and know the sound of your microphones, you can often guess which one will complement a particular singer for a particular application. For instance, for a singer with a nasal voice, I probably won't use a Neumann U 87 (a standard choice for vocals), because its midrange peak will tend to exaggerate the worst qualities of that type of voice. Instead I might use a ribbon mic or even an EV RE20 (which is a large-diaphragm dynamic).

Furthermore, I might not use the same mic from track to track. An intimate ballad with a detailed sound and the singer performing close to the mic calls for a different mic than the one I would use on a rock song with belted-out vocals. Some mics don't respond well to high sound-pressure levels, and
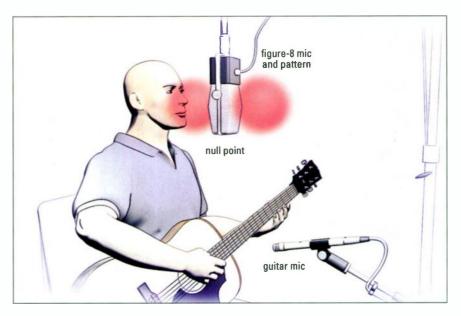
FIG. 2: For recording a singer playing acoustic guitar, a figure-8 mic with its null point aimed at the guitar is often effective. This arrangement picks up mostly vocals and minimizes leakage from the guitar. Optionally, you can use another figure-8 mic with its null point aimed at the singer to pick up the guitar and reject the vocals.

many singers' voices change timbre when they change their delivery. You need to consider all the details to make the best mic choice.

GET IN LINE

If I have the time in a session, I usually set up three or four likely mic choices for a singer to audition. I explain to the vocalist that we are looking for the mic that goes best with his or her voice. I set the mics up side by side, set levels as accurately as possible, and get a working headphone mix going. I then have the singer take a warm-up pass or two at a track, singing a verse through each mic. Then we listen back and hear which one sounds best. Sometimes none work well, and I leave one up and try two or three more.

Some singers are happy to spend hours auditioning mics and getting a vocal sound. In most cases, however, you'll need to make the choice quickly to avoid wasting time and money and ir-ritating or tiring the singer. Still, if you make it clear that you're trying to find the most appropriate mic instead of just using the usual setup, you're likely to get appreciation and cooperation from the vocalist.

A key element to this process is making sure that you, as the engineer, have a good knowledge of the sound of your microphones and of microphones in general. This comes with practice and experience, and I recommend spending your spare time listening to different mics and getting a feel for how they sound. Bear in mind that mics of the same brand and model don't all sound the same. I once spent an afternoon with a fellow engineer testing all 12 of his Shure SM57s to figure out which one sounded best. We compared them and made notes about their sound qualities so he could choose the
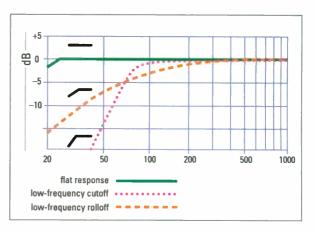
FIG. 3: The low-frequency attenuation circuitry built into many studio mics gives you options for reducing the proximity effect and any low-frequency room noise present in the recording environment. This chart shows the effect of the two low-frequency attenuation choices available on the Shure KSM44.

Tracking the Elusive Vocal

most suitable among them for any given situation.

REFLECTING ON YOUR SPACE

Correct mic placement is crucial. You might have a great mic, but you can dramatically alter the sound quality—for good or ill—with the placement, relative to both the singer and the surrounding environment.

A neutral-sounding recording space devoid of reflections that could color the sound is ideal for cutting vocals. For that reason, and to block out unwanted sounds, the vocal booth is a key tool for recording engineers. Even if you don't have a studio with an actual vocal booth, it's easy to construct a makeshift one that will produce good results. Creative deployment of gobos, clothes lines, blankets, mattresses, or any number of reflection-absorbing materials can transform any studio, bedroom, or living room into a fairly inert-sounding space.

In the past I have made booths by leaning a mattress against a wall to create the back and draping blankets over two horizontal booms on stands to create the sides. A third blanket-draped boom stand can close off the front of the booth if necessary (see **Fig. 1**).

You can also make gobos quickly and cheaply by building a frame out of 2×4s, stapling cloth to one side, filling it with insulation, and stapling cloth to the other side. Nail a couple of pieces of the 2×4s to the bottom to make feet, and you're in business. The time you take to improvise a booth will pay off.

SHAPES OF THINGS

More often than not, you end up using vocal mics with cardioid patterns. This

FIG. 4: To reduce plosives, you can use a metal pop filter (left), a foam windscreen (or "sock," center), or a mesh pop filter (right). Pop filters are better in most situations because they don't attenuate highs as much as foam screens do.

is because you're generally trying to minimize room sound (or reflections from a music stand holding lyrics), and a cardioid pattern is tighter and will pick up fewer reflections. And because the majority of microphones are cardioid, the choice is often made for you in advance.

If you are recording in a reflective space or in a room with other instruments, you might need a tighter pattern for better isolation. A common approach to recording a singer playing acoustic guitar is to use a vocal mic with a figure-8 pattern; aim the null point at the guitar to get the least amount of bleed from the instrument (see **Fig. 2**). Some mics have multiple or variable polar patterns, so experiment. Just keep in mind that as you tighten the pattern (moving toward figure-8) you generally lose some high-frequency response; the situation dictates the desirability of that loss. (Sometimes it's useful to set a multipattern vocal mic to figure-8 to intentionally darken the sound a little rather than using EQ.)

PROXIMATE CAUSE

A side effect of directional patterns is *proximity effect*: that boost in the low frequencies that happens when you position a directional mic close to the sound source. Sometimes the proximity

COMP IT UP

A comp track is a single "best of" track that's put together from multiple takes of the same song. To assemble it, listen to the individual vocal takes and determine where the best performances of the various lines and sections are. Do this with a lyric sheet and pencil in hand, writing down the segments you prefer: verse one, take three; verse two, lines one through three, take two; verse two, lines four through six, take four; and so forth. Once you have your road map, you're ready to go.

On a tape-based system, bus all the tracks to a new track. Make sure the levels all match, roll tape, and "play" the comp using the channel mutes. Mute and unmute the various tracks as you go through the song to get the best performance onto the comped track. The great thing about this process is that it's completely nondestructive. If you screw up, the original takes will still be intact, and you can just try again. If you make a mistake halfway through, you don't have to start from the top; you can punch in on the comp track and pick it up from the line before your error. When you're done, listen to your newly created comp track and make sure you're happy with the performance and the levels.

If you're working on a hard-disk-based recorder, assembling a comp is even easier. I usually pick the track that is the best overall and simply cut from other tracks and replace sections in the master track. (I would recommend cloning the master track first so you have a backup in case you're unhappy with the comp.) You also have the luxury of doing subtle time shifting if you want to make rhythmic adjustments.

effect can be a singer's best friend, filling out a thin or shaky voice. Other times it just creates mud. Because you'll want to take advantage of this effect in some situations and not others, it's important to know how to control it.

Many mics have built-in rolloff filters designed to reduce handling noise and the proximity effect. Some give you several choices. For instance, the Shure KSM44/SL has two types of low-frequency filtering: low-frequency rolloff (see **Fig. 3**), which is a subtle attenuation designed specifically to counter the proximity effect, and low-frequency cutoff, which is a more extreme attenuation designed to eliminate rumble from the surrounding environment. (That's an important consideration for recording at home, where passing traffic or running kids can create low-frequency rumble.)

A rolloff (whether on the mic, preamp, or mixer) is one way to reduce the proximity effect; using an omni-directional pattern is another. There are many affordable multipattern condensers. By setting the mic to omni, you'll eliminate the proximity effect and make it possible for the vocalist to work the mic extremely closely. The trade-off is that the omni pattern picks up more room sound. You'll have to judge whether it's worth it.

Omni mics are also useful for singers who can't stay still. With a unidirectional mic, the proximity effect and the narrower sweet spot necessitate a stationary singer. Much movement will cause significant variation in the sound. An omnidirectional mic largely eliminates that issue, letting a singer bob and weave with much less effect on the mic's tone. Again, you might lose some control over the sound because of additional reflections, but at least it will be more consistent over the track.
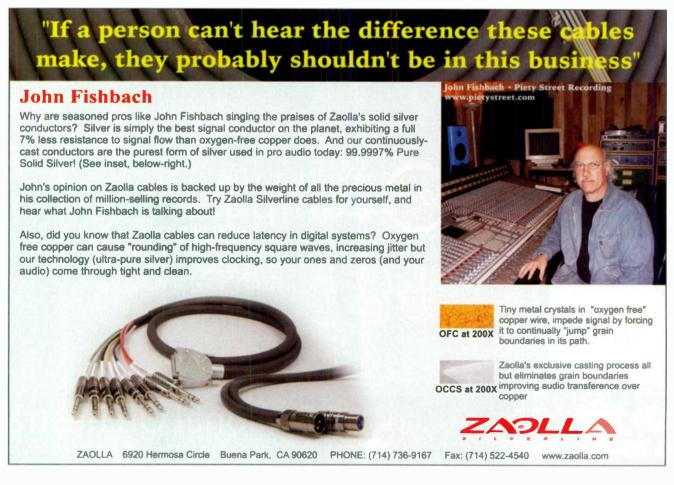
If you have a variable-pattern mic, try using a pattern that's a click or two past cardioid in the direction of omni (a wide-cardioid pattern). I generally get more "air" in the sound that way, and it reduces proximity effect a notch or two.
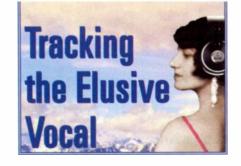
CLEAN AIR

Air and presence are important qualities in a vocal, but they're not at all similar. Air refers to the frequency range above 10 kHz that creates a sense of openness; it gives the sound more depth and dimension and (to my ear) adds intimacy. Presence is somewhere in the range of 3 to 8 kHz, depending on the voice. It lets the voice cut through a mix and helps articulation. It's also a quality that is overengineered into many personal-studio microphones these days.

Some people believe that all vocals have to be bright and edgy, but that's very unnatural sounding in many cases. I'm particularly sensitive to overbright sounds, and I still gravitate toward warmth and air rather than heavy presence. Be careful when buying any mic

Tracking the Elusive Vocal

with a heavily colored sound; such a mic can be wonderful on the right vocalist, but terrible on others. You need to let the singer and the music dictate the vocal sound, not the mic.

POP GOES THE SINGER

Another question is whether to use a pop filter. Popping of *p* and *b* sounds *(plosives)* is a fact of life when recording vocals because you're placing a microphone inches in front of someone moving quite a bit of air. I generally like to have singers four to eight inches from a condenser mic and sometimes an inch or two from a dynamic.

Moving the singer further away from the mic dramatically reduces the force

of plosives, but I find it usually flattens the vocal sound a bit and accentuates the room more, and I don't like that trade-off. Nor do I like raising the mic a couple of inches above the singer and aiming the capsule at the singer's mouth. That gets the capsule out of the

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If the singer is doing anything other than singing, you've lost.
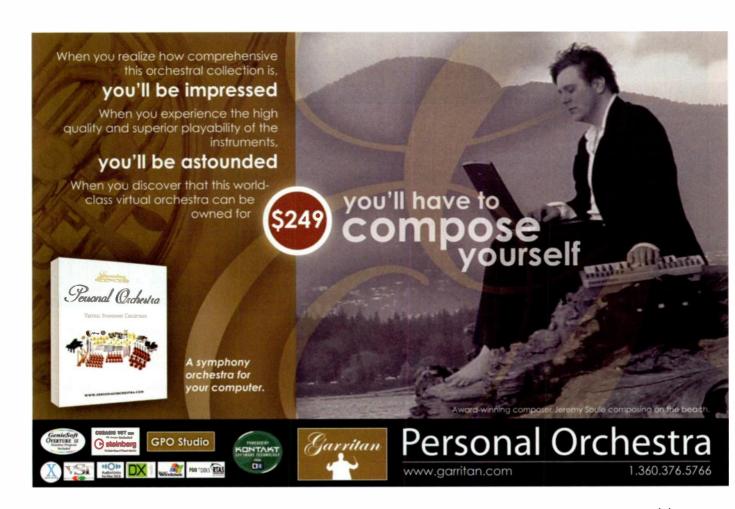
line of fire, but it can also add off-axis coloration, which usually does not improve the sound quality.

You can avoid moving back the mic or vocalist by using a *sock*, that thick gray foam windscreen that's included with many mics. Although socks generally stop most plosives, they often filter out

a significant amount of high end as well. A sock can be useful if you want to cut highs, but otherwise it will only diminish the airiness of a vocal.

A better option is an external mesh pop filter, the kind made of stocking-like material stretched across a plastic ring. But even that will attenuate the highs to some degree. I think the best choice is a metal-screen pop filter. It does a solid job of stopping plosives, has the smallest impact on the sound quality, and it looks cool, which always counts for something in the studio (see **Fig. 4**).

Another benefit of using an external filter of any sort is that it creates a distance marker. When you find the ideal distance between the mic and the singer, the pop filter can serve as the "goal line." Many singers like to eat the mic; some forget where you positioned them. The filter can keep them and their lunch off your prized possessions, and it keeps the singer in the zone.
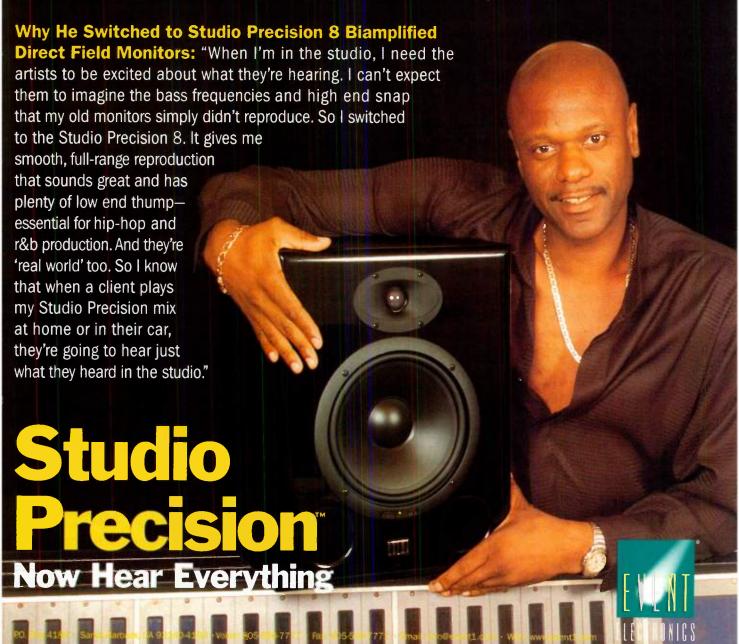
DOWN THE LINE

Once you've found the mic you and the vocalist are happiest with, you can continue putting together the signal chain. After the mic comes the mic preamp (if you're really hardcore, the next item is actually the cable, and it can make a difference). Mics can behave very differently through different preamps. Many people compromise the sound by using cheap mic pres that don't sound good or that don't have strong enough phantom power to get the mic up to optimum operating voltage. It's kind of like buying a Ferrari to drive in Alaska. You can't get the most out of the car on a snow-covered road, and you can't get the most out of a microphone if the preamp isn't up to the task.

Time and resources permitting, I typically audition a few different mic preamps to see which best complements the mic I've chosen. (If I'm already familiar with the preamp, I can simply plug into it, knowing that it does "that certain thing" that I like.) Experienced engineers know mic preamps are an important variable in the recording chain; some are neutral, others extremely colored. Seasoning to taste is the best approach, as always.

Although it might be tempting to immediately start equalizing a singer's voice to get the sound the way you want it, you should use equalization as a last resort, after you've exhausted whatever mic-and-preamp combinations and mic placement options are available. I say this because it's best to use the smallest possible amount of gear in your signal chain. Doing so minimizes noise and distortion, and it also reduces the number of things that can go wrong. (It makes it easier to duplicate the sound elsewhere, too.) That said, EQ can be an essential sweetener. There are some great equalizers out there that add qualities no mic or preamp can.

DYNAMIC RESTRICTIONS

Compressing vocals to tape is standard practice—in fact, I'm hard pressed to think of an occasion when I recorded a vocal track without using some compression. The human voice's dynamic range is extremely wide, and some compression is usually necessary to get a reasonably consistent level on tape or disk (especially if the singer is inexperienced and doesn't have good dynamic control).

Sometimes extreme compression is an essential part of a vocal sound. Compression brings up the level of breaths, lip smacks, and other vocal nuances, and in some cases this effect is really cool. It can make the sound so intimate you'll think the vocalist is singing in your ear (which can sometimes sound a little creepy). It can also change the way a singer performs.

One vocalist I often record loves to have his voice squashed—usually though an Empirical Labs Distressor or a UREI 1176—and he performs better when he hears himself singing that way. It's his sound, and it works with his voice, but it's certainly not right for all voices. As always, if you are in any doubt, use a conservative amount of compression. It's easy to add more later, but you can't remove it if you use too much. And each compressor has its own sound, so factor that in; do you need warmth, detail, transparency, or just plain squish?

ON A ROLL

Once you've dealt with all the setup work, and you're ready to roll tape (or

TEN TIPS FOR SUCCESSFUL VOCAL RECORDING

1. Set up the studio environment to be as comfortable and relaxing for the singer as possible. Good ambience and good vibes will lead to a better performance.

2. Choose a microphone that's compatible with the singer's voice. Ask the singer in advance about mics he or she has had success with, and if you have that model available, use it. If you're not sure which mic to use, audition several before making a final choice.

3. Start out by positioning the singer four to eight inches from a condenser microphone, and one to two inches from a dynamic.

4. Use a pop filter to reduce plosives.

Metal filters are the best choice, and mesh models are a close second.

5. Consider using an omni-pattern mic if the singer has trouble staying still while recording. An omni will help keep the vocal tone consistent. Omnis are also good in situations when you want to eliminate the proximity effect.

6. Set up a headphone mix that's comfortable for the singer. Take out unnecessary instruments that could muddy the mix and make it difficult to discern pitch.

7. Try using different combinations of mics, preamps, and mic placement in order to get the sound you want before you resort to EQ.

8. Use compression when recording to keep the dynamic range of the vocal in check. But be careful not to over-compress; you can always add more during the mix if needed.

9. Try to record complete takes whenever possible and keep the punching to a minimum. This will enable you to get a better, more natural performance from the vocalist. If necessary, make a comp track composed of the best parts of several takes.

10. Write down all settings, mic and preamp choices, and positioning information. If you need to rerecord any of the vocals at a later date, your notes will make it possible to recreate the original vocal sound.

Tracking the Elusive Vocal

roll hard drive, as the case may be), there are still some things to keep in mind. Generally I find that singers do better taking complete passes to start with. Sometimes a singer will hit a wrong note and ask to stop and start over from that point. But more often than not, I find it's better to roll a complete take. That allows the singer to perform all the way through, and it tends to keep a better vibe through the track.

If the singer has the stamina (and if I have the tracks), I prefer to roll several complete takes and then comp together the best parts from each take to get one master vocal (see the sidebar "Comp It Up"). I used to do a lot more punching, and I still do it frequently, but I found that the more punching I did—constantly stopping, starting, and listening again—the less cohesive the overall vibe became.

Still, there are times when a vocal track is killer except for a few lines or words, in which case I'll happily punch. (As a geeky engineer I love the challenge of punching syllables.) But be mindful of how the singer performs. Some singers never sing a line the same way twice, and that can make it very difficult to do short punches; you might have to replace an entire verse to get one line right.

You're best off working with the singer. If he or she only wants to do one pass and then fix rough spots, that's fine. If he or she wants to do three takes and then comp, that's also fine, but do the comp soon while everyone still has what they liked fresh in their minds. Doing a comp a month later can be hard, because people might not recall which takes they liked best in what places of the song.

HEARING DOUBLE

Double tracking vocals is still a common practice. Although there are now a variety of boxes that can artificially double a vocal track—and some sound

quite good—there is still a certain magic to a naturally doubled vocal. When I know at the beginning of the session that a double track is called for, I like to get one completed vocal track with all punches and comps finished, and then record the double.

Before that second track goes down, it's helpful to fine-tune the headphone mix so that the singer can hear the first track clearly enough to follow and match it, yet also hear enough of his or her live voice. Some singers want the two signals panned apart, others want them at equal levels, and others want the live voice louder in the mix. Spend a few minutes and get it sorted out for optimum results.

When recording the double, I find most singers still prefer to do a complete pass and then go back and punch the spots that don't quite match up.

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It's kind of like buying a Ferrari to drive in Alaska.

Some singers can double themselves in their sleep, and it's a quick and easy process, but others aren't really aware of what they did on the first track, so they have to listen repeatedly to lines to match them. In those cases, doubling can be a long and painful process, and it can make you appreciate those neat processing devices even more.

VOX IN BOX

Speaking of magical boxes, I should acknowledge two devices that are becoming more common: pitch correctors and microphone modelers. I have worked with and reviewed boxes that perform both operations, and I think they are useful tools, especially in a personal studio. Pitch correction can save a fantastic vocal performance that might have a bad note or two, and it can also let you avoid a lot of retakes because you know that you can fix pitch mistakes later. The possible trade-offs are sonic artifacts and an overall loss of

quality because of the extra processing.

Microphone modeling is a technology that can also be useful for a personal-studio recordist with a minimal microphone collection. I find that it's more like creative equalization than dead-on emulations of classic microphones. (I've yet to hear a box that makes a Shure SM57 sound like a Neumann M 49.) Nevertheless, it's a great concept, and it's easier to afford than 20 different mics.

WRITE IT DOWN

Once you've chosen the microphone, pattern, placement, pop filter, mic preamp, compressor, EQ, and anything else, it's important to write it all down. Take notes of every detail of your vocal settings: preamp gain, compressor settings, pad settings, barometric pressure—everything. This is essential because there is a good chance you are going to have to redo a vocal or come back to it the next day or a month later.

Any change in the setup can drastically change the sound and make it difficult if not impossible to match it in the future. Chronicling your settings can be a lot of work, but it can also be a lifesaver. I've had to go back to a track months later to replace one line, or even one word, and by referring to my notes I've been able to re-create the vocal sound in a matter of minutes. Draw charts, graphs, take pictures, whatever you have to do, but make sure you record every detail.

Another reason for copious note taking is that another engineer might work on the project and have to match the vocal sound. I've had to do it by ear, when the previous engineer didn't take notes, and it's not easy. (Although the challenge can be kind of fun in that geeky, recording-engineer kind of way.) What's more, it's always useful, for future reference, to have a log of the sounds and approaches you've used. After all, if you got a good sound once, you'll probably want to use it again.

Sean Carberry is field producer for WBUR, Boston's NPR station. When he's not on the campaign trail, he tries to spend time in the studio recording music he actually likes.

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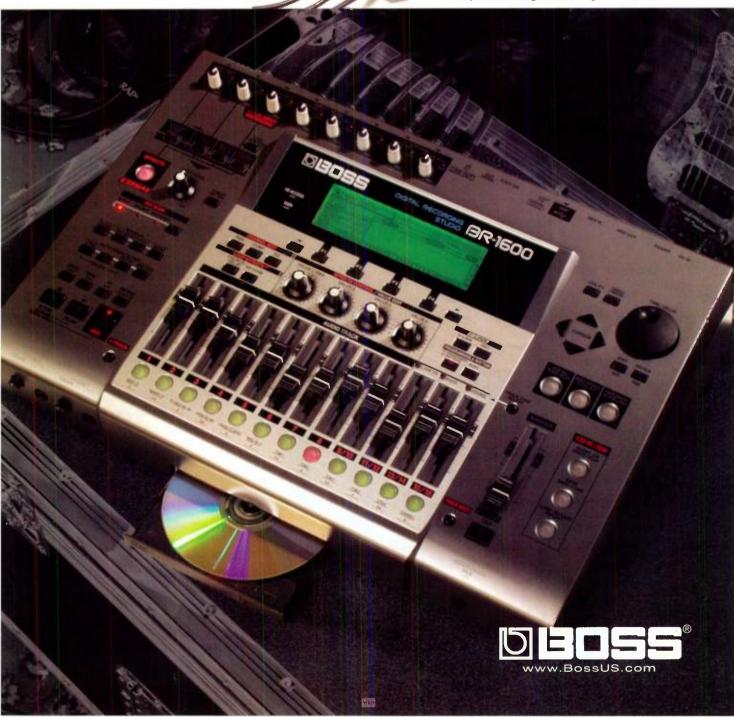
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Yes you counted right, there's eight XLR inputs.

Playing with Pd

Make your own real-time MIDI processors and soft synths for free.

By Jim Aikin
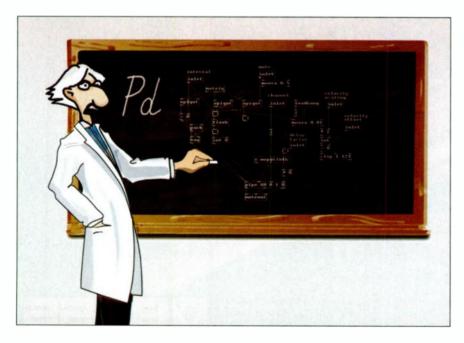
It wasn't too many years ago that musicians with more passion than money thought nothing of building their own music hardware. *Electronic Musician* founding editor Craig Anderton, who is handier than most with a soldering iron, once built an entire mixer for his home studio.
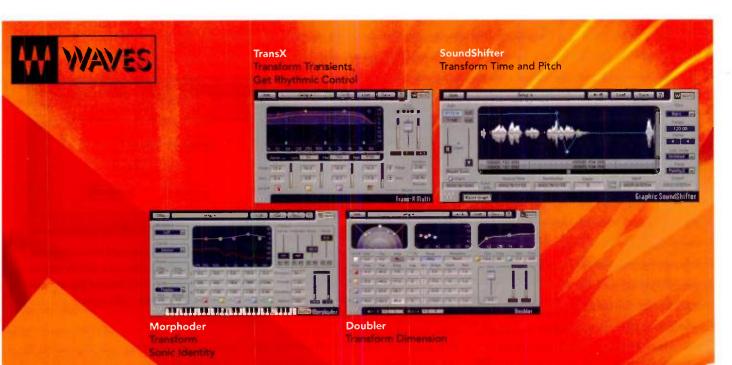
Now that the world is inundated with massively engineered gadgets such as cell phones and shirt-pocket computers, it's getting tough to build your own hardware. But creating your own music software is easier than you might think, and more affordable, too. In fact, you can do it for free.

Music software for DIY aficionados comes in several flavors. You can write your own sequencer or synth in a conventional programming language, but there are easier options. This month I'll take a close look at one of the more powerful and (relatively) user-friendly programming environments for music: Miller Puckette's Pd. You don't have to know anything about computer programming to use Pd, but some patience and the ability to think logically wouldn't hurt.

Given that so many wonderful music software tools are available with the wave of a credit card, why would you want to roll your own? For one thing, you may want to do esoteric things that no commercial software will do. With the aid of a program like Pd, you can generate and process complex streams of MIDI data for a live performance or build your own software-based synthesizer and effects.

Beyond that, solving technical problems in software design can be a fascinating challenge. You get a sense of accomplishment from creating something that works exactly the way you

DMITRY PANICH

envisioned. Then there's the money issue. Your personal Pd software synth may not sound quite as massive as Native Instruments Absynth or VirSyn TERA, but if you've spent your last dime on a fast computer and a decent sound card, tools such as Pd and Csound (see "Csound Comes of Age" in the July 2002 **EM)** are an unbelievable deal.

In this column, I'll tell you how to get Pd, and I'll provide a few insights into what it's like to work with it. (A full Pd tutorial is beyond the scope of this column.) To give you some ideas about what you might want to do with Pd, I've created two MIDI processing utilities and an unusual step sequencer. You can download them from the **EM** Web site. Once you've installed Pd, you'll be able to use them and customize them to meet your needs.

WHAT IS PD?

Pd is a freeware graphical programming environment for real-time MIDI and audio manipulation. (Extensions are also available for video processing.) To create a program in Pd, you connect little boxes to one another with the mouse using graphic "patch cords." Once you've created or loaded the patch, you can interact with it by sending MIDI messages to your computer and by using onscreen sliders and buttons.

If you've ever seen or used Cycling '74's Max, Pd will look eerily familiar, because to a considerable extent, Pd is

FIG. 1: Pd's main program window has an audio on/off checkbox, a File menu for opening your Pd patches, and not much else. Patches are created in other windows.

Max. Puckette created Max at IRCAM in the mid-1980s. At present, Max is a commercial application (for Mac and Windows), and Puckette has no direct involvement with it. However, he's actively developing Pd, which, in addition to Mac and Windows, also supports Linux and Irix operating systems.

One of the main differences between Pd and Max is that Pd's functionality is simpler than Max's, and its simple user interface, although adequate, is quite bare-bones. A third variety of the same software, jMax, is also available as a free download from IRCAM (www.ircam.fr/equipes/temps-reel/jmax/en/index.php3).

The fact that Pd is free has a number of implications. If you're a C programmer you can customize the source code from the ground up or write new Pd objects that will integrate with the programming environment. Also, there's no tech support. There is, however, an active mailing list (http://iem.at/mailinglists/pd-list) to which you can subscribe.

Technical questions that are posted to the list are usually answered within a day or two by more experienced Pd users. For lots of useful miscellany on Pd (including downloadable goodies designed by various people), the place to go is www.pure-data.org. Puckette, who now teaches at the University of California at San Diego, is not affiliated with pure-data.org, but he does monitor the mailing list.

GETTING AND RUNNING PD

All of the versions of Pd can be downloaded from http://crca.ucsd.edu/~msp/software.html. No special installation procedures are needed. The Windows version, which is what I use, arrives as a Zip file. After it's unzipped, you should place the folder named pd in the root directory of your C drive. (Mac users can get an OS X–style "package" created by Adam Lindsay at http://homepage.mac.com/atl/sw. It simplifies the installation process.)

In Windows, Pd runs from a com-
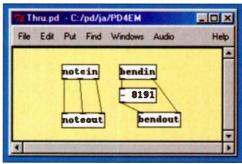
FIG. 2: Shown here are two extremely basic Pd patches. On the left is a MIDI Thru connection for note data; on the right is a Pitch Bend Thru connection. Signals come out of outlets on the bottoms of the objects and enter inlets on the tops.

mand-line interface. You can't run it by double-clicking on the program icon. Instead, you open an MS-DOS prompt window and navigate to the directory containing the program by typing this:

cd c:\pd\bin

The prompt will then say: c:\pd\bin>. That shows you're in the pd\bin directory, and depending on what MIDI and audio devices are available in your system, you then start the program with the needed command-line arguments. I have an ASIO sound card, so I start the program like this:

c:\pd\bin>pd -asio -midiindev 1

This tells Pd to use the ASIO driver and the first MIDI device listed in the Windows control panel.

Keeping the MS-DOS window open while Pd is running is both necessary and useful. For diagnostic purposes, Pd includes a handy "print" object whose output appears as text in the MS-DOS window.

Pd's main window hardly qualifies as a window at all, though it does have clipping indicators and a couple of checkboxes (see **Fig. 1**). When you open an existing file or load a new one, you'll be working in a separate window that looks more like **Fig. 2** or **Fig. 3**. New objects can be added to this window from a menu, or you can type Control-key commands, which is faster.

Pd comes with a set of tutorial files to get you started. If you're new to the whole idea of graphic programming, you should definitely spend a few days

working your way through the tutorials. Pd also has interactive Help files for most of its objects. Right-clicking on an object brings up a context menu from which you can select Help. That opens a new Pd window with information and an actual functioning Pd patch that demonstrates how the object works. Some of the discussions are a bit sketchy, but even so, the Help files are a great resource. You can also read an HTML manual in your Web browser.

PROGRAMMING IN PD

There are a few tricks to handling audio in Pd, so it's best to start with MIDI. Pd provides a number of objects for handling MIDI, the two most basic being notein and noteout. To use them, insert a new object box from the Put menu, and type "notein" (without the quotation marks) in the box. Follow the same procedure to make a noteout, and connect them as shown on the left side of Fig. 2.

The little black indicators at the top and bottom of each box are inlets and outlets. All objects in Pd receive signals at inlets (on the top of the box) and transmit signals from their outlets (on the bottom). A connection is made by clicking on an outlet and dragging with the mouse to the inlet of another box. Each type of object does something different, so each has a different combination of inlets and outlets (see **Fig. 4**). The inlets expect to receive cer-

tain types of signals, and if an inlet receives a signal that it can't handle, you'll get an error message.

The connection between notein and noteout in Fig. 2 does nothing except pass on to the output all of the MIDI Note On messages received at the input. It's a MIDI Thru connector for notes. The connection between bendin and bendout is a little more interesting. This patch subtracts 8,191 from the data coming out of the left outlet of bendin, and then passes on the result to the corresponding inlet of bendout.

Why do that? Because this simple patch fixes a bug in Pd. The bendin object is sending the MIDI channel of the incoming Pitch Bend message from its right outlet, and the value of the Pitch Bend message from its left outlet. MIDI Pitch Bend is a 14-bit message, which means the data can have any of 16,384 possible values. But as it happens, bendin transmits this data in the range between 0 and 16,383, while bendout expects to receive it in the range between –8,192 and +8,191. By subtracting 8,191, this patch puts the Pitch Bend data in the range that bendout needs. (Note: like most computer programs, Pd expects large numbers to be entered with no commas.)

Just about everything you do in programming Pd will be more or less like that: figuring out what range of values the receiving object needs to see, and then shaping the data stream so that it
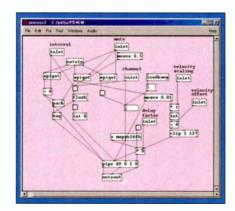
FIG. 4: Opening one of the "process" boxes in Fig. 3 reveals this user-created subpatch. The subpatch receives data through the inlets at the top, and the noteout object at the bottom transmits MIDI data.

looks right when it reaches its destination. In addition to simple arithmetic operations, you'll use objects like clip (which limits data to the range you specify), spigot (which opens to allow data through and shuts to cut data off), moses (which "parts a data stream," sending values above a threshold to one outlet and values below the threshold to another outlet), and value (which stores a numeric value for later use).

Web Clips 1, 2, and **3,** which you can download from the **EM** Web site, will give you a taste of what you can do. I don't have space here to go through each patch in detail, but I've included comments in the files themselves.

The xposer (transposer) patch (see Fig. 3) can turn each incoming MIDI note into a four-note chord. For each note in the chord, you can choose a MIDI channel and scale the Velocity up or down. Each note can also be delayed, turning the chord into an arpeggio. The delay time is set in 16th notes at the current tempo, and some simple chord presets have been provided. The ctrlprocess patch is for scaling, offsetting, limiting, and inverting MIDI Control Change messages. The stepseq patch can generate sequences up to 64 16th notes long, and you can interact with the sequence in numerous ways while it plays.

CLEAN CUPS, MOVE DOWN!

The key to Pd programming is understanding the order in which the
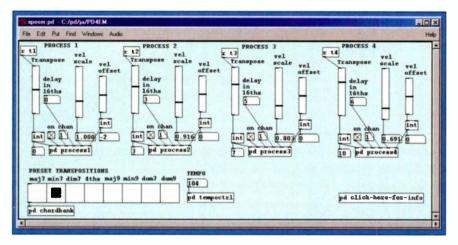
FIG. 3: The xposer patch (which you can download) outputs up to four MIDI notes for each note received. The sliders, checkboxes, and the button bar at lower left are all user-interface objects supplied with Pd.

messages are transmitted. A personal computer is extremely fast, but it does only one thing at a time. A Pd algorithm can work in an unexpected way, or fail to work at all, if you make a bad assumption about the order in which events take place.

Objects in Pd send "simultaneous" messages from their outlets in a seemingly counterintuitive way, servicing the outlets in right-to-left order. But what happens when several patch cords are attached to a single outlet? In Max, the objects attached to a single outlet receive their messages in right-to-left order based on the screen position of the receiving object. That has a good

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Creating your own music software is easier than you might think.

side and a bad side. The good side is that you can always see the order in which messages are passed by looking at a patch on the screen. The bad side is that you can "break" a patch if you tidy up the screen and in the process change the right-to-left orientation of two objects that are attached to the same outlet.

In Pd, this situation is turned on its head. When several patch cords are attached to a single outlet, the Pd object sends its messages in the order in which the patch cords were created. As a result, you can safely shuffle objects around on the screen (you'll be doing that a lot) without breaking the patch. But if a patch is malfunctioning, you can't necessarily tell what's wrong by looking at it, because there's no visual indication of the order in which the messages are being transmitted. Deleting the patch cords and reattaching them may be the only way to be sure of what's going on. In fact, I've found that when subpatches are copied and pasted in Pd, the order of the connections can change. After copying, I

sometimes have to delete patch cords and reattach them.

AUDIO IN PD

When it comes to synthesis and signal processing, Pd is very good at some things and not quite so good at others. It will do sample playback and FM synthesis, and the waveshaping tools let you easily produce a variety of tone colors. Pd's filters, however, don't compare with what you can get in commercial software. A basic ADSR envelope generator is part of the software library that comes with Pd, but creating a multisegment envelope with Velocity modulation inputs is a project that can easily take an evening if you're new to Pd programming.

Audio objects in Pd (more than two dozen are provided) have tildes (the ~ character) at the end of their names. The osc~ object, for example, is an oscillator. The mtof~ object translates MIDI note numbers into frequency values suitable for playing equal-tempered scales, the readsf~ object plays a sound file from the computer's hard drive, and the dac~ object sends audio to the sound card.

Audio data normally has a range from −1 to +1. If you accidentally multiply an audio data stream by a large value, Pd will not protect your speakers or your ears from the results. Nor does it have a DSP usage meter; in my experiments, however, Pd seems fairly efficient (though the screen redraws are slow).

PURE DIY

Currently at version 0.37, Pd is a work in progress, not a finished program. But while it may not do everything you might envision (at least not easily), it has enough tools to keep a do-it-yourselfer busy for a long, long time. Best of all, you don't have to worry about burning yourself with **EM WEB CLIPS** a soldering iron!

Jim Aikin writes about music technology for a variety of publications and Web sites. His most recent book is Software Synthesizers (Backbeat Books, 2003). You can visit him on the Web at www.musicwords.net.

Things I Wish I'd Known Then
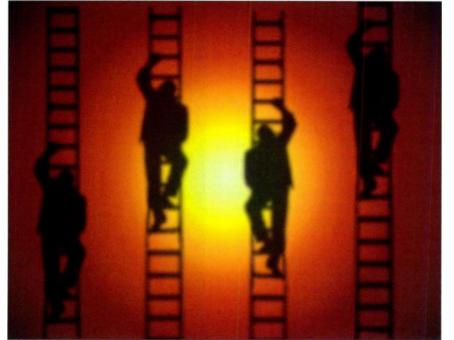
Ten lessons learned during 20 years in the business.

By Jeffrey P. Fisher

It's often a struggle for musicians to build the life they want to lead. The uncertainty and constant flux that characterize today's music industry only make achieving a modicum of success that much harder. For many of us, it's a long, arduous journey, full of challenges and setbacks—and hopefully a few bright, shining moments, too.

I wish that your career could be solely about the music. Unfortunately, commerce and the other demands of real life intrude on that ideal. While your primary focus should always be on your music—honing your craft, polishing your performance, augmenting your skills, and growing as an artist—you must also attend to other aspects of your career. Here are ten basic concepts that I wish someone had pointed out to me when I started out in the business more than 20 years ago. By implementing this advice, you can rapidly gain a distinct advantage in the music industry.

HOW TO BUILD A REPUTATION

Building your reputation is a crucial part of making your career better. As buyers, many of us are skeptical of new things because we've made poor purchasing decisions in the past. We don't want to repeat those mistakes, so many of us are less willing to take a chance. However, a good reputation instills confidence and reduces that fear of buyer's remorse. When people trust that you'll deliver what they want based on your track record, they are more willing to support you. People buy your rep, so spend your resources growing and selling it.

You build a reputation through image

DMITRY PANICH

and credentials. An image establishes what you are about, and it should appeal to the people you are trying to reach. Look at the acts popular today and see how image is fundamental. Savvy promoters know it's often easier to sell an image than content (a few "celebrit-neys" come to mind). Package your image through diligent and consistent presentation of your music style, song topics, attitude, dress, speech, graphics, and other visual material, including your Web site, video, and posters.

Credentials show people that you are legitimate. If you have an act, you must play live. I consult regularly with up-and-coming musicians who don't tour. I tell them that if they won't play live, it will be difficult to establish a fan base and even harder to make any money. Most acts earn their pay on the road, not selling CDs and downloads.

Even though touring can pay more than CD sales, having a CD creates genuine credibility. Showcasing your work on disc is the main way to demonstrate your skills and prove you are for real.

Other credentials come from third-party endorsements such as media reviews and testimonials from satisfied buyers, clients, and peers. This evidence of your success works its magic on the doubtful. Don't believe this is effective? Look at all the movie ads in the newspaper. Two thumbs up goes a long way toward getting people into the theaters. Finally, work to win awards such as first place in a song-writing competition, best band in a showcase, or, ultimately, a Grammy. Those are credentials you can take to the bank.

Even if you're not an act, if you're working in the music industry, this formula applies. Combine a meticulous image with specific credentials, and you'll earn a reputation that buyers will trust and support.

MAKE SUCCESS HAPPEN

How do you quickly make your music career better? Learn all you can about how the business really works. Read, take classes, find a mentor or two, and get real-world, hands-on experience. The more information you gather, the easier it will be to make good decisions.

Don't sit on your hands waiting for something to happen, either. Be proactive. Building a reputation, cultivating business relationships, and growing your career take time and effort. You need to push hard continually. What are you doing today to reach those people who want and need the music products and services you sell? What actions can you take that move you closer to your goals?

All aspects of the music industry are competitive, and you have to really want your success to achieve it. Many of us moan about those with lesser talent getting ahead, but they usually make up for their lack of chops with an abundance of ambition and hard work.

Success is self-perpetuating in a sense. The more you achieve, the easier it is to sustain. But that doesn't mean you can ever become complacent. Don't take your clients or your fans for granted or you'll risk quickly losing everything you've struggled to earn.

DON'T LIVE IN A BUBBLE

I often hear from would-be composers wanting to score independent films. I soon discover that they don't hang out with indie filmmakers, go to indie film festivals, or even read the indie film magazines. It's crucial that you start associating with the people who are either in a position to help your career along or ready to hire you. You can't make it completely on your own.

Join and participate in your music scene and start networking with industry people, media, and your peers. Start helping others in whatever ways you can. Ask about what they do and what they need. Let them know about your skills and what you are looking for, too. Networking is a form of barter. You want to build long-term, mutually beneficial relationships with people, not just take, take, take. When you approach and give assistance first, people will, out of a sense of obligation, help you in return with referrals, good word of mouth, and occasionally a paying gig.

BE MONEY SAVVY

If you're earning money from your musical pursuits, and I do hope you are,

get your finances together fast. If you don't keep track of what you make and spend, how will you know your true financial situation? How can you successfully complete your taxes?

If you are not an employee and instead manage your music career yourself, you are in business. As a small business, you are subject to many legal, financial, and tax regulations. Since finances and taxes interrelate, it's crucial that you establish a specific method for tracking income and expenditures. Setting up a proper bookkeeping system now will save many headaches later.

Get a business checking account and a credit card for your business. Deposit your music income into the account and pay all your business expenses from it. Use either a ledger or software to record all these transactions. The more detail you use, the clearer the picture. If you track specific income streams—differentiating between gigs, CD sales, and royalties, for example—you will see which activities are profitable and which are not. Also, carefully monitoring your expenses in detail reveals where you over- or underspend. (See "Working Musician: Hit the Books" in the January 2004 **EM**.)

DON'T MESS WITH TAXES

I was once stung by the bee of a substantial tax bill. It was my own fault, having jumped, young and green, into the waters of my first business without researching key issues. I collected check after check from client after client and failed to notice that nothing was being deducted from those checks. Self-employed individuals pay their own "payroll" taxes (federal and state income taxes, social security, and Medicare contributions). Consequently, I faced a substantial tax bill come April the next year, owing for the previous year and making my first estimated payment for the current year.

When all this came down, I made two important decisions: one, to research all the issues that affected my business, and two, to go out of my way to help other people avoid making the same mistakes. There are many distinct advantages to having your own business

and there are many pitfalls. Make sure you fully understand all the tax issues as they apply to your particular situation. (See "Working Musician: Tax Tips for Musicians" in the March 2003 issue.) Knowledge here benefits you greatly—you will save money and stay out of trouble. Also, examine any legal issues, especially liability, to protect your business and personal assets.

GETTING YOUR MESSAGE OUT
Plop down some change for an ad to promote your new album and people will beat a path to your door, cash in hand, ready to buy your latest opus. Right? Wake up! It just doesn't work that way. One ad, no matter how ingenious, will not turn an unknown into a celebrity. Unfortunately, too many people waste their cash on such pipe dreams. If any promotions you use don't immediately move more products or services out the door, rethink your approach.

When it comes to promotion, you can spend money or you can spend time. If funds are short, you need to get more creative with your promotions and devote more time to them. Usually these get-up-and-go tactics are both more effective and substantially more profitable than simply throwing money at the problem. Don't fall into the easy trap of relying on passive promotions, such as advertising, when what you really should be doing is being more active.

The personal touch really works in today's often sterile, anonymous world. No ad, letter, e-mail, or flyer is ever going to take the place of standing face-to-face with someone and making the sale. Back-of-room sales are always helped when the whole band participates and interacts with buying fans. In person, and to a lesser extent on the phone, you can build a solid rapport, address every sales objection, and win people over faster. Just finished work

for a happy client? Now is the perfect time to ask for another project or gig.

NARROW YOUR MARKET FOCUS
I'm sure you would agree that promoting your industrial band in a country-music magazine isn't the smartest idea. Yet every day brings evidence that people continue to use scattershot promotion when pinpoint accuracy is what works. Our world is deeply segmented, and the tighter your focus on a narrow market segment, the better your results will be. Trying to reach everybody with your message is a waste of time and money. Instead, find the people who already like what you do and concentrate your promotional efforts on them.

Do you really know who buys your music? If not, you'd better roll up your sleeves and find out. You must know specifically who these people are, what they want from you, how and where you can reach them, and which ways of doing so will appeal to them. Start

investigating your local scene and build from there. Go to clubs, hang out with patrons and other acts, read local newspapers and magazines, review radio, television, and film. Check out other acts' music, ask questions, and conduct market research. Do whatever it takes to get the information, and then use what you learn to advance your own career.

EXPAND YOUR BASE

If you sell a service (for example, a musical performance), you should also sell products (such as CDs). If you sell products, you need to provide services, too. Introduce new products and services regularly. Have a studio CD? Put together a quick compilation of live performances and sell that. Better still, sell a DVD with performances, interviews, demos, and other goodies. After that, work on the second studio CD, and so on.

Don't limit yourself to selling only the products and services you create. You can sell other people's stuff, too. Recommend other products and services that complement or augment your own work and then figure out how to profit from your advice.

For example, let's say you have agreed to promote CDs by other artists on your Web site. You can join the free Amazon.com Associates program and sell the CDs mentioned on your site. You put order links on your Web site and Amazon pays up to a 15 percent commission on orders that originate from your links. Best of all, Amazon handles the whole transaction and pays you quarterly. (See this concept in action at www.jeffreypfisher.com.)

INCREASE SALES

Everybody likes to save money, including your fans and clients. Stimulate some sales through an aggressive discount program. Give your loyal fans (or clients if you have them) exclusive offers, price breaks, and discounted advance copies. Put together packages that entice buyers to spend more money with you. For example, sell your independent CD for $10, and throw in a band-logo T-shirt for $10 more ($20 total). You've doubled your sales vol-

ume fast. (For more on boosting the profitability of your music career, see "Working Musician: Shake Your Moneymaker" in the June 2002 **EM**.)
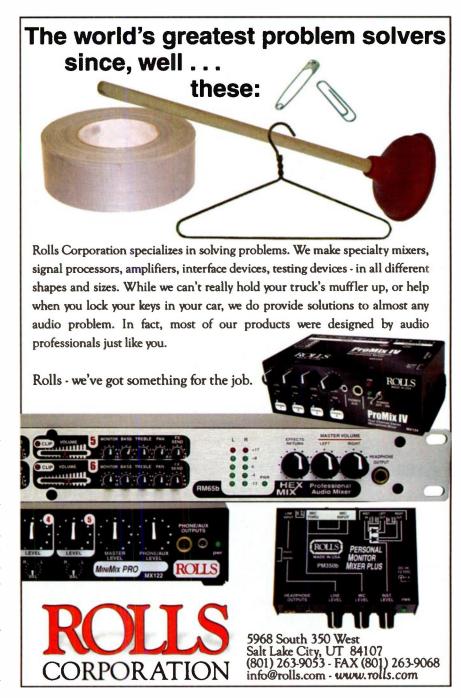
PATIENCE AND COMMITMENT

Just about the time you get tired of promoting, networking, and the other tasks that comprise an active music career, people will start to notice you. Don't give up now. Set goals and devise specific methods for achieving them. Trust that you've made the right decisions (and be willing to adjust and adapt as circumstances change). Commit yourself to all of these steps and diligently take care of yourself and your career.

Jeffrey P. Fisher's *latest book*, Moneymaking Music *(artistpro.com, 2003), is a detailed guide to starting, growing, and sustaining your music business career. See it and other music-success resources at www.jeffreypfisher.com.*

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REVIEWS

ABLETON

LIVE 3.0 (MAC/WIN)

Stay in the loop with this must-have upgrade.

By Len Sasso

The Ableton team has come up with another stellar upgrade to Live. The big news here is in the automation department, where Ableton has added individual Clip envelopes as well as improved drawing and editing tools. Other improvements include more flexible MIDI and computer-key Clip triggering, the ability to play Clips from RAM as well as from disk, new rendering and Scene-capturing capabilities, and more quantization options.

The Live loop library has been reworked and expanded, and four new DSP effects have been added. For a detailed look at Live from the ground up, see the Live 2.0 review in the June 2003 issue of **EM** and the Master Class "Going Live" in the December 2002 issue (available at www.emusician.com). In this review I'll give a brief overview of Live, then concentrate on the new features.

Live is a standalone audio sequencer especially designed for performance, but it's also well suited to recording, arranging, and remixing. It is available for all recent versions of Windows on the PC

FIG. 1: Live's Session view is set up like a typical mixing desk, with channel strips along the bottom. Audio clips are triggered from the Clip Pool in the center. Clips can be triggered individually or in rows (called Scenes), using MIDI, the computer keyboard, or the mouse.

and for OS 9 and OS X on the Mac. It supports ReWire, functioning as either master or slave (but not both at the same time), and it hosts VST effects plug-ins.

Live runs in one window but has two views corresponding to two modes of operation. The Session view (see **Fig. 1**) is laid out like a standard mixing desk with channel-strip controls along the bottom. Unlike a mixing desk, however, Session view has a Clip Pool in the center for recording and triggering audio clips. In a performance you would trigger individual audio clips or whole rows (called Scenes) using MIDI, a computer keyboard, or a mouse. The Arrangement view is laid out like a typical audio sequencer, with audio tracks running horizontally. You can capture a Session-view performance in the Arrangement view, drag clips there and arrange them manually, or record live performances there.

The Session and Arrangement views are connected in two ways: they share audio channels, and both are always active for playback, even though only one is visible. Initially all audio channels play Arrangement-view tracks, but as soon as a clip is triggered in the Session view, that view takes control of its audio channel. That lets you set up an arrangement of a song and then substitute clips from the Session view in real time—a key difference between Live and other audio-sequencing software. Live's unique setup offers many other performance, composition, and recording options as well.

CLIP AUTOMATION

In previous versions of Live, automation was limited to the Arrangement view, where any mixer or insert-effect parameter can be automated. That method, commonly called track-based

automation, is great for mixing and overall effects automation, but it is not particularly well suited for sound design on the individual Clip level, because any automation would need to be moved, copied, or re-created each time the Clip is used. In Live 3.0, you can draw automation envelopes directly in the Clip view, and that automation follows the Clip as it is moved or copied to other locations (see **Fig. 2**). If you move a Clip with effect automation to a new track that doesn't contain the effect, the automation is lost. That makes sense, but it would be nice if the automation were preserved and restored if the effect was again applied to the Clip.

Clip automation differs from track automation in several ways. For one, Clip automation is applied relative to current settings and to track automation. For example, if you automate track volume on the Clip level by setting alternate eighth-note segments to half volume, and then move the track volume slider or automate it on the track level, the alternate eighth-note volumes will always be half the current volume setting. Another difference is that Clip automation loops with the Clip, so if you stretch the Clip in the Arrangement view, causing it to repeat, the automation repeats as well. In effect, that makes Clip automation a form of nondestructive DSP.

In an interesting and useful twist, you can unlink the automation from the Clip, in which case the automation gets a complete set of position and loop parameters of its own. That lets you apply an automation loop of a different length than the Clip's loop length. Alternatively, you can create a long automation sequence applied to a short, looping Clip. The online MP3 example ResoDrum ap-

plies Sample Offset and Volume automation to a one-bar percussion loop, which is then processed by Live's new Resonators effect (see **Web Clip 1**).

On the Clip level, you can automate all track parameters, including plug-in settings. That includes Clip parameters, which can't be automated on the track level, such as Clip Volume, Transpose, Sample Offset, and—in Tone and Texture Warp modes—Grain Size and Flux. By automating Transpose in unlinked mode, you can adapt a short, looping musical phrase to match harmonic changes in other Clips. Sample Offset automation is intended for quickly rearranging the beats in a beat loop. Although limited to 16th-note steps and a one-measure range, it is quite effective. Using an automation loop of a different length than the Clip loop produces especially interesting results.

Other automation improvements include a Draw mode, which lets you draw stepped curves, and the ability to freely copy and paste automation between parameters. The new automation drawing tool creates automation steps the size of the selected quantization. You can copy and paste different parameters within the same Clip as well as between different Clips.

ALL KEYED UP

Another important new feature in Live 3.0 is the ability to assign a range of MIDI notes to trigger a Clip. Previously, only individual notes could be assigned to Clip slots. This change enhances Live's functionality as a multisample player, as does the new Clip Velocity parameter, which determines the extent to which Velocity affects Clip volume. But beyond standard multisample playback, Live's automatic time warping makes the ability to assign note ranges an ideal tool for live groove transposition. By assigning a

FIG. 2: Live's new Clip envelopes let you automate all playback parameters on the clip level. Clips, along with their automation, can then be freely moved or copied to other Session-view slots or Arrangement-view tracks.

> **Minimum System Requirements**
>
> **Live 3.0**
>
> **MAC:** G3/266 MHz; 256 MB RAM; Mac OS 9.2 or OS X 10.1.5
>
> **PC:** Pentium II/400 MHz; 128 MB RAM; Windows 98/2000/XP

range of notes to a looped groove, for example, you can transpose the groove without affecting its timing.

A new Legato option makes it possible to change Clips without retriggering. When Legato is engaged for a Clip, triggering another Clip on the same track in the Session view, or triggering the same Clip with a different MIDI note, starts playback at the current time position rather than at the beginning of the Clip. That feature is especially effective when

applied to an evolving ambient Clip over which you've defined an unlinked automation loop; it transposes the ambience relative to your key selections while keeping the animation and Clip playback continuous and in sync.

A number of improvements have been made to Live's already excellent user interface. The quantization options have been expanded to include triplets and lengths greater than one bar. Samples can be dragged directly

from the File Browser into the Clip view. This replaces the audio data while preserving all Clip settings and automation, which effectively allows Clips to function as automation templates. Management of VST banks and programs has been improved, and multiple VST plug-in windows can be open for editing at the same time.

Live now offers a play-from-RAM mode, useful if your disk drive is too slow to stream all the tracks in your song. This can also improve performance in Legato mode. The Consolidate command quickly renders the current Arrangement view selection into new Clips on a per-track basis. Rendering the entire song or loop is, of course, still available. And there is a Session-view Capture command that creates a new Scene from all currently playing Clips.

SOUNDS, EFFECTS

Four new effects have been added. Compressor II is a compressor/limiter with look-ahead capability. EQ Three is a DJ-style 3-band EQ with 24 and

FIG. 3: The Resonators effect applies five resonators to the incoming signal. The first determines the base pitch and applies to both stereo channels. The remaining four resonators are tuned relative to the first and apply alternately to the left and right channels.

48 dB rolloff. Utility is a multipurpose output utility with controls for gain, stereo channel suppression (allowing either channel to be sent to both sides of the stereo mix), stereo width, and phase inversion for each channel.

Resonators is a bank of five resonators, four of which are tuned to intervals relative to the pitch selected for the first resonator (see **Fig. 3**). You can use the effect to create chords as well as enharmonic sounds from pitched and unpitched material. Applying Clip automation to the individual resonator frequencies opens up a whole new range of possibilities. The resonators are preceded by a multimode filter allowing you to resonate only part of the signal, and the output has wet/dry balance, gain, and stereo-width controls.

The Live 3.0 sample library (available only with the boxed version) is packed with new samples from Big Fish Audio and PowerFX. As is typical, these collections contain examples from larger libraries available from those providers, but there are enough samples in each category to keep you busy. Especially notable is the PowerFX samples CD with over 500 MB of samples in virtually every instrument and style category.
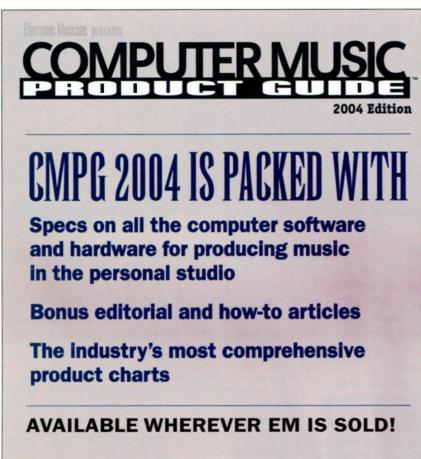
The Live 3.0 download upgrade comes with the manual in PDF format, which is a big improvement over the HTML format used in previous upgrades. The boxed version has a printed manual, but as in previous versions of Live, neither manual has an index.

LIVE AND LET LIVE

If you're already a Live user, there's no question that this upgrade is an incredible bargain. You can hardly afford not to get it. If you use other loop-based audio-sequencing software or are looking into Live for the first time, it should definitely be high on your list of packages to consider.

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

SONY PICTURES DIGITAL

SOUND FORGE 7.0A (WIN)

New owners keep this classic up-to-date.

By Dennis Miller

Version 7 is the first edition of Sound Forge to be released by its new developer, Sony Pictures Digital. This venerable audio editor was previously owned by Sonic Foundry, and though it is sad to see an old friend disappear, it's clear Sony intends to keep Sound Forge current and competitive.

Unlike the past few releases, which mostly played catch-up and had little in the way of visible interface changes, Sound Forge 7 adds many of the modern features found in the company's other software (Vegas and Acid, for example), and also includes a host of new options not seen elsewhere in the product line. There's a lot to get through, so let's get started.

RECORD READY

At the top of the new-features list is a host of recording options that make Sound Forge a much more flexible program for capturing audio. One of the more esoteric, though potentially very useful, options is the new time-based recording feature. If you're a soundscape composer and need to grab, say, a recording of some sonic environment one hour every day, you can set Sound Forge to record either indefinitely or for a set duration (two weeks, for example), then return later to retrieve your audio.

Another type of automated recording is possible using the new Threshold feature. This familiar function has Sound Forge scanning an incoming signal and starting recording when the signal reaches a level that you determine. In addition to setting the level (from −∞ to 0 dB), you can set the amount of time the program will continue to record after the signal moves below the threshold. You can also choose to rearm after each new recording completes if you want to make a series of recordings using the same threshold.

If you've ever missed the very be-ginning of a recording because your computer hiccuped or your mouse finger missed the mark, you'll also appreciate the new Prerecord Buffer, which constantly captures a user-defined amount of time from the audio input, even before you initiate recording. The buffer size can range from 0 to 30 seconds.

New metering options provide much better views of your signal. In addition to the Peak level meter found in older versions, you can now use VU and Peak Program meters with a variety of user-definable scales (including traditional and logarithmic VU; and UK, EBU, and DIN peak program) to view the average loudness of a signal. Many engineers prefer VU metering to peak metering because it's a better tool for matching track-to-track loudness. Sound Forge lets you view Peak level and either Peak Program or VU at the same time.

LIVE, FROM VEGAS!

If you're a user of Vegas or Acid, you're already familiar with some of the new interface enhancements. Sound Forge now includes an Explorer window, in which you can navigate to and preview any supported media files on your system (see **Fig. 1**). The Auto-preview option is similar to the Auto-play feature found in the File/Open dialog and will start playing back a file as soon as you click on it. However, even though I have a multiclient sound card, playback of the currently loaded sound file stopped when I clicked on a file name in the Explorer. I'd like the option to hear the new file mixed with a file I already had loaded.

If you drag a file from the Explorer onto an existing file, Sound Forge lets you mix, crossfade, or paste it into the currently loaded file. Dragging directly from the Explorer means you don't have to first open a file, copy it to the Clipboard, then select which method

FIG. 1: Sound Forge 7 has acquired many of the interface elements of its sibling products, Acid and Vegas. You can now dock an Explorer window anywhere on the screen and draw volume and pan envelopes directly on the waveform view.

of combining the two files you want. When you mix or crossfade, a dialog box pops up in which you can adjust the balance between the existing and the new file. Sound Forge will also mix stereo and mono files automatically.

Also imported from Acid and Vegas are new envelope options, including a dedicated envelope-editing tool. You can add a pan or volume envelope and modify it directly in the main waveform display, and you can automate many (but not all) DX effects. Rather than limit you to linear segments, your envelopes can now have one of six different shapes, selectable on a per-segment basis. (The new envelope types also work in the FM-synthesis tool and in several of the Processes and Effects.)

When you first load an automatable plug-in in the Chainer, you'll see a list of its parameters on the right of the screen (there's a folder in the Plug-in Manager that contains all the automatable plug-ins). You can enable automation for each parameter individually, and you can choose whether or not automation envelopes will be displayed on the screen, also on a parameter-by-parameter basis; very flexible indeed. In the original version 7 release, when you looked at the envelopes superimposed over the waveform display, you couldn't tell which envelope controlled which parameter until you clicked an envelope and

started to drag it. Fortunately, like Acid, version 7a now shows you the parameter's name when you point to it with the mouse (be sure to update online if you have an early release).

VISUAL MUSIC

Sound Forge has an enhanced spectral-analysis view with a vast range of new options. Though the display itself isn't up to the level of my all-time favorite analysis view, which is in Steinberg's WaveLab, there are many new features that can give you a lot of useful information about the content of your audio (see **Fig. 2**). A tabbed interface takes the place of the old drop-down menus, making it easier to change settings, and the Spectrum window is now dockable, so you can leave it open as a file plays back. Equally important, you can use the Real-time monitoring option to see a moving picture of a sound as it plays, and the Hold Peaks feature makes it even easier to see where the action is in the spectrum. You can also save up to five Snapshots (custom arrangements) of the Analysis window.

Speaking of seeing, Sound Forge 7 now supports one of the hottest new video formats, which is known as "24p." This format is used for high-definition broadcasts and is especially useful if you are transferring film to video (or vice versa), because both film and 24p run at a rate of 24 frames per second. As a result, there is no conversion needed as

there is with standard NTSC video, which runs at 29.97 frames per second.

AND THAT'S NOT ALL!

Other handy new features appear throughout the program. For example, in previous versions, you could drag any audio file from the Windows Explorer directly onto the Sound Forge interface, and it would open the file. Now you can even drag a track from an audio CD and Sound Forge will extract the audio automatically.

The new Project file format lets you save a record of all the edits you make in a work session even after you close the file and the program. You could then go back to a saved file and undo any edits you might want.

With the exception of the ExpressFX Vinyl Restoration plug-in (available with 7.0a), there's nothing new in the Processes and Effects categories, places I turn to first with every new release. It might be time for Sony to think about adding some new effects (vocoding, for example?) or to add support (available in Acid) for VST-format plug-ins. (You can use a wrapper, such as Cakewalk's VST Adapter, to get around this limitation.) There are a few new options in the Synthesis menu—for example, four new noise generators (filtered, pink, brown, and white)—but here again, perhaps Sony would consider an additive-synthesis tool for generating static waveforms more complex than the ones currently available.

Though not new, Sound Forge's Playlist feature deserves mention as perhaps the least-heralded feature in the software. Sound Forge doesn't have Acid's looping capabilities, but if you have any interest in creating music with highly rhythmic, repeating segments (not to mention stuttering and other modern effects), give the Playlist a good look.

HELP IS ON THE WAY

Sony has provided a number of support resources for both new and experienced users. Among these are a 23-minute comprehensive online video tutorial and a 23-page PDF describing the new features, complete with usage tips. Alas, the
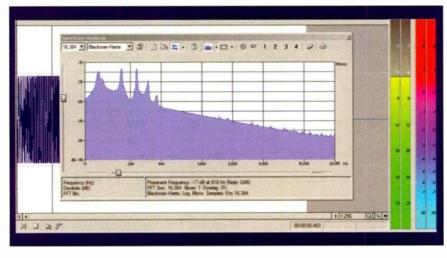
FIG. 2: The new spectral-analysis options in Sound Forge allow you to get an inside look at your audio. The real-time monitoring feature provides an animated display while a sound plays back.

bound, printed manual, available with early versions of Sound Forge, has not magically reappeared. But a 300-page PDF manual is included with the release (and available for download). An active online users forum is also a great resource for anyone with questions.
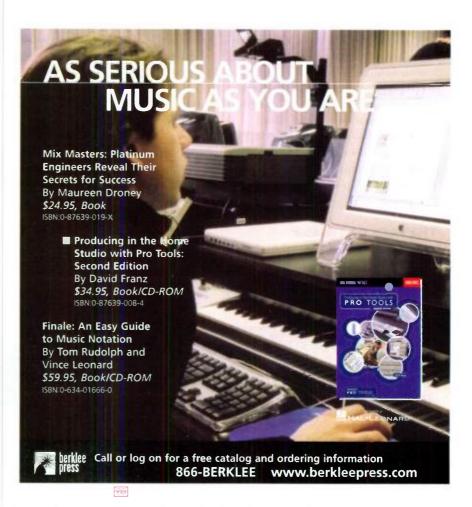
It's worth mentioning that though Sony will only support users running Sound Forge 7 under Windows 2000 or XP, I used both an XP laptop (with an Indigo I/O interface) and a 98 desktop machine (with a MOTU 2408mk3) and ran into no problems whatsoever.

Though still only a stereo editor, Sound Forge remains an indispensable tool for audio editing on the PC. No matter how fancy the audio-editing features get in Windows digital audio sequencers, I still would never undertake a serious project without Sound Forge close at hand.

Perhaps someday, Sony will integrate the features of all its many multimedia applications—consolidation increases with every release. But of all the tools in the company's catalog, Sound Forge is the one I turn to most.

Dennis Miller *is an associate editor of* **EM**.

Y A M A H A

SPX2000

A signal processor upholds and improves upon its heritage.

By Larry the O

The Yamaha SPX90 was like the Model T of digital multi-effects processors. It wasn't the first digital effects unit capable of doing more than one thing, but it had the most-used effects of all types, was priced well, and held up reasonably well against the rigors of constant use. It ultimately became a de facto standard, which Yamaha wisely chose to perpetuate rather than abandon. Successors such as the SPX90II, SPX900, SPX990, and SPX1000 added refinements and better audio quality as the technology improved.

The most recent offering from the SPX line is the SPX2000. Like its forbears, the SPX2000 is a 1U device with stereo input and output and all the standard effects programs: reverbs, delays, choruses, pitch shifters, compressors, noise gates, EQs, and more.

The SPX2000 features 24-bit, 96 kHz operation and a new set of high-quality reverb algorithms called Rev-X. It also includes a USB port for connecting to computers.

THE FACE OF THINGS

The SPX2000's front panel and software architecture stick closely to the classic SPX configuration, but they do deviate to accommodate some improvements. The front panel has a concentric input-level knob for the two inputs (see **Fig. 1**). It's occasionally pesky to use but offers welcome flexibility, especially if you use the inputs as two independent mono inputs feeding a single effect.

The familiar LED-ladder meters are just to the right of the input-level knob, and to their right are a group of status LEDs and two mode buttons. The buttons themselves are quite small but feel more solid than I expected.

The backlit LCD now includes a new feature. Its background color changes to show what type of effect program is running or to indicate a problem (such as losing the clock when slaving to an external input).

A two-character program-number display supplements the LCD, and two sets of four large buttons are used to select programs and edit parameters. The button area also includes smaller Undo and Compare buttons and three buttons labeled Parameter, Fine Param, and Utility. Parameter gives you access to the basic parameters, and Fine Param (a misleading name) provides additional parameters. With these two buttons, the SPX2000 offers a solid set of editing controls for each effect program.

The Utility button provides configuration settings for MIDI, input-source selection, display color, and program naming. The front panel also includes handy Bypass and Power buttons along with a footswitch jack and a button for use with the Tap Tempo feature.

BRINGING UP THE REAR

The rear panel features XLR and TRS jacks and a +4/–10 level switch for inputs and outputs (see **Fig. 2**). Curiously, Yamaha references both levels to 0.775V (both are dBu), though most audio devices reference 1V (dBV) for a –10 dB nominal input or output level. The SPX2000's –10 dBu output is actually –12.2 dBV, and therefore enters a –10 dBV (nominal input level) destination device with 2.2 dB less level than if both devices used the same voltage reference for a –10 dB signal. That creates a noticeable, if not critical, level difference.

The rear panel's other connectors include XLR connectors for AES/EBU digital audio input and output with a maximum sampling rate of 96 kHz. A BNC word-clock input resides next to the aforementioned USB port, which is provided for use with the editor-librarian software that should be available from Yamaha's Web site by the time you read this review. The SPX2000 also has MIDI In and Out/Thru connectors; most effects parameters can be controlled dynamically by MIDI. I find combination Out/Thru jacks inconvenient when I have several processors in a single rack and want to daisy-chain Thru to In while reserving Out for editing from the computer. However, for most applications, it's a tolerable setup.

On the far left of the rear panel next to the IEC power jack is a grounding screw. This last item is a thoughtful recognition of how often SPX-family devices are used in rooms with less than optimal AC wiring.

ARCHITECTURAL DIGEST

The SPX2000's hardware design incorporates several thoughtful features, such as the front-panel mono/stereo switch and switchable input/output metering. Combining the mono/stereo switch with the choice of a one-in/two-out or two-in/two-out effects program yields any combination of mono or stereo processing. You can even mix

FIG. 1: The front panel on Yamaha's SPX2000 is generally consistent with earlier SPX models.

FIG. 2: The SPX2000's rear panel includes digital and analog audio I/O, MIDI I/O, a word-clock input, and a USB port.

the two inputs to mono using the front panel level controls.

Like other SPX units, the SPX2000 has a single processing engine. It has no chains of modules and no separate programs for the left and right channels. Yamaha has apparently decided to emphasize fidelity rather than fancy features, which is not surprising—the SPX series has never been about flash and megatweaking, it's been about bread-and-butter utility and a broad range of applications.

The effects programs are organized into three banks. The biggest of the three, the User bank, stores up to 99 custom programs. Unfortunately, this bank ships empty, which is a missed opportunity; it could have been used to deliver a larger preset library. The Preset bank contains 97 programs, of which 17 are the new Rev-X reverb. The other 80 are mostly new versions of the standard array of effects, but there are a few new entries, including a multiband dynamics processor and a multifilter.

The Classic bank contains 25 "best of SPX" programs. It is worth noting that each of the Classic programs, and quite a few of the Preset programs, are the only programs using their respective algorithms.

The SPX2000 is a breeze to use, and it takes very little effort to figure out. Far be it from me to suggest that you skip reading the manual, but this is one device where you could probably get away with it. The only usability problem I found vexing was that the fast parameter scrolling is extremely fast. It moves from one end of the range to the other before you know it. You can go nuts zipping back and forth, or you can give up and do it slowly.

REVERB REVELATION

I tried the SPX2000 on a number of sources (primarily vocals, guitars, drums, vibes, percussion, and synthesizers) during overdubbing and mixing. I also compared the SPX2000's reverb against several others. The most interesting comparisons were with a

SPX2000 Specifications

Digital Audio I/O	(1) AES/EBU in; (1) AES/EBU out
Analog I/O	(2) ¼" TRS in; (2) ¼" TRS out (–10/+4 dBu); (2) XLR in; (2) XLR out
MIDI	(1) In, (1) Out/Thru
Other Connections	(1) BNC word-clock in; (1) USB; (1) footswitch
Resolution	24-bit
Sampling Rates	44.1, 48, 88.2, 96 kHz
Frequency Response	20 Hz–20 kHz (0 dB +1.0, –3.0) @ 48 kHz; 20 Hz–40 kHz (0 dB +1.0, –3.0) @ 96 kHz
Dynamic Range	106 dB (A/D+D/A)
Total Harmonic Distortion	0.01% @ 1 kHz (96 kHz sampling rate)
Memory	Preset bank: 97 programs; User bank: 99 programs; Classic bank: 25 programs
Displays	16-character × 2-line LCD with 5-color backlight; 2-digit program-number LED; (2) 12-segment LED ladders (left/right)
Dimensions	1U × 14.7" (D)
Weight	8.8 lb.

Lexicon PCM80 and Audio Ease's Altiverb plug-in.

The SPX2000 sounds quite good in a wide range of applications. The audio quality of the processing is better than average for a product at this price, thanks to the analog-to-digital converters, internal processing power and resolution, and the effects algorithms. Yamaha has been making reverbs and effects for a long time, and its refined algorithms show it.

The new Rev-X algorithm is the jewel in the crown. It is very dense and has less coloration than older Yamaha reverbs (such as the first- and second-generation REV series). It also has a healthy number of adjustable parameters. Rev-X reverb isn't as versatile as the PCM80's or as dense as Altiverb's. However, I spent some time matching algorithms and parameters as closely as possible, and I was able to get the SPX2000 sounding so much like those other reverbs that you couldn't hear the difference in a mix.

When soloed, the SPX2000 still held up well in direct comparisons, but its limitations did show, primarily as some graininess in the high frequencies. That would only occasionally be a problem for me; most of my applications that require high frequencies in the reverb are for small spaces with decays short enough to hide the graininess. I generally got the best results setting the SPX2000's high-frequency decay time much shorter than the low-frequency decay time. The bottom line, though, is that the Rev-X reverb is a great choice for many uses.

Possibly the most demanding and revealing pair of sources I compared was solo vibes and vocals. A somewhat spacious, though not cavernous, reverb was needed, and the Rev-X Hall algorithm did a great job of producing an airy, shimmering sound. I tried all of the Rev-X presets, but for that application, the Hall preset was it.

NEW AND OLD

After exploring the Rev-X, I worked my way through nearly all of the other presets. As I mentioned, many presets are the only ones for a particular algorithm, so the total number of algorithms available is large. For vibes and vocals, I liked the phasers, flangers, and choruses, including the trademark Symphonic chorus, in addition to the Rev-X.

The SPX2000's results were not quite as spectacular on drums. I was not wild

about the plate programs on any source material. (In fairness, I have yet to find any digital plate program that I really like, so I'm hard to please on that point.) Likewise, the SPX's chamber program didn't impress me on drums; it was a bit boxy and not smooth enough.

My usual choice for drums would be a room program, and there I met with some success. Almost all of the presets had decay times that were too long for drums, but some yielded good sounds with a little tweaking. The SPX2000 was adequate for drums, which ought to be fine for the many users who will buy this box for the vast array of things that it can do. It wouldn't be my first choice for drums, however, if I had the PCM80 or Altiverb available (just to name two that I had on hand).

Most notable among the other new programs in the SPX2000 are the dynamics algorithms, which process audio based on input level or MIDI input. Also noteworthy are the dynamic multiband compressor, triggered filter, and multifilter. This last program consists of three filter sections, each of which can be set as a lowpass, bandpass, or highpass filter.

I tried all of those along with other effects, including the ring modulator and the rotary-speaker emulation (which features separate rate settings for the high and low spin speeds). All were at least usable, and some were a lot of fun and capable of very cool sounds. There are way too many algorithms in the SPX2000 to give individual report cards, but I can report that the embarrassingly warbly pitch shifting of the original SPX90 is a thing of the past.

NOBLE HERITAGE

All in all, the SPX2000 is exactly what it should be: a device that maintains its heritage while moving it forward. Like all SPX units, it does a great many things, with varying degrees of success. It may not be the most exotic, flexible, powerful, or best-sounding kid on the block (though the Rev-X reverbs compete well). It does, however, offer a lot at a reasonable price, and the current model's performance is clearly an improvement over earlier generations.

While the SPX line has remained consistent in its approach, the market around it has changed. Today, there are multi-effects processors galore in the $200 to $500 price range, although truly excellent fidelity still typically costs $1,500 or more. That leaves the SPX2000 curiously alone at its price point. The closest competitors are a few hundred dollars on either side, and that could give this device a nice niche as the entry point into the world of upscale audio processing.

The SPX2000 sounds good, sometimes excellent, and fulfills all of the basic needs for a studio or live rig. It's also easy to use and has some nice bells and whistles, such as its 24-bit, 96 kHz digital I/O. (You could even use it as a 96 kHz A/D converter.) The SPX2000 is a very solid performer that will meet many needs just as its predecessors did, only this multi-effects processor does it better. The Model T is now a latter-day Mustang. Hop in and take a ride.

Larry the O has yet to encounter a singing folk vibist other than himself, but there must be another one somewhere.

IK MULTIMEDIA

SAMPLETANK 2 (MAC/WIN)

Make way for the virtual rompler.

By Geary Yelton

The best thing about *romplers*—sample-playback synthesizers with a huge selection of recorded sounds in ROM—is their immediacy. Switch on the power, turn up the audio, select a sound, and start playing. Ever since E-mu introduced the first Proteus, romplers have ruled the roost, onstage and in the studio. Even today, when synthesists can choose from a multiplicity of sound engines that encompass physical modeling and real analog synthesis, the most popular instruments are still romplers: the Korg Triton, the Kurzweil 2600, the Yamaha Motif, and the Roland Fantom series are just a few.

Until recently, the closest that soft synths came to such ease of use was IK Multimedia's sample-playback plug-in SampleTank. Fire up your sequencer, open a plug-in, select a MIDI track, pick a sound, and start playing. In Sample-Tank 2, those capabilities have been expanded and enhanced. SampleTank 2 is available in two editions: XL and L (that's down from three editions of the original). One of XL's many improvements is a library of Instrument Presets that's more than twice as large as before with no increase in price.

Foremost among SampleTank 2's new features is the ability to import samples from nonnative formats without the separate ST Converter program previously included with SampleTank XL. SampleTank 2 also provides three sound engines: one based on sample playback, another based on pitch shift and time stretching, and a third called SampleTank Time Resynthesis Technology, or Stretch.

SampleTank 2 now functions more like a traditional synthesizer thanks to over 50 new user parameters that encompass LFOs, 6-stage envelopes, and multimode filters. The effects section has been beefed up considerably, and the user interface is greatly improved. All told, SampleTank 2 is a much better substitute for a rompler than the original ever was.

LET'S GET TANKED

For this review, I used the Audio Units version of SampleTank XL 2.0.3 in MOTU Digital Performer 4.12 and Emagic Logic Platinum 6.3.3, and the VST version in Steinberg Cubase SX 2.0.0. (SampleTank 2 also supports RTAS in Mac OS 9 and OS X, MAS and VST in Mac OS 9, and DX, RTAS, and VST in Windows 2000 and XP.) My computer was a dual-processor Power Mac G4/1 GHz with a gigabyte of RAM and over half a terabyte of hard-disk space, running Mac OS X 10.3.2. The audio interface was a MOTU 2408mkII, and the MIDI interface was an Emagic MT4.

An installer program took care of installing the plug-in, which initially opened as a demo. When I entered my serial number, a window displayed my computer's digital ID. I entered that number on IK Multimedia's registration Web site, and my authorization code arrived almost immediately by e-mail. Installation to authorization took only minutes. You are allowed three authorization codes, so you can install SampleTank 2 on your notebook and desktop computers and still have an authorization left for the day you upgrade to a newer model.

The sample library arrives on eight CD-ROMs. Each disc has its own installer, so I installed the entire library one disc at a time. I would have preferred to install the contents of all the discs in one pass, but I can see the logic of this one-installer-per-disc system if you're short on hard-disk space. The entire XL library is over 4.5 GB. SampleTank 2 L has a 2 GB sample library, the same size as the previous XL's.

ONCE AROUND THE BLOCK

Compared with the previous version, SampleTank 2 has more than twice as many knobs and buttons on its bright-red virtual front panel (see **Fig. 1**). The group of four Instrument-specific parameter knobs is now labeled Macro, and eight dedicated knobs appear in the Effects section. In the new Synth-Sampler section, eight additional knobs control parameters that depend on which button you click on—Synth, Filter, Envelope 1 or 2, LFO 1 or 2, or Velocity. An onscreen keyboard is displayed along with pitch-bend and mod wheels.

FIG. 1: SampleTank 2 offers dozens of improvements over the previous version, especially with regard to its program-editing capabilities. In addition, the XL version's sample library has more than doubled in size without an increase in cost.

You can control any knob with any MIDI Control Change (CC) message, but making assignments isn't as simple as selecting MIDI Learn, clicking on an on-screen knob, and moving a control on your hardware. Instead, when you click on the MIDI Ctl button and then a knob, a pop-up window appears for you to define your MIDI controls (see **Fig. 2**). Using that technique, I was able to control 16 independent parameters with an Evolution UC-16. You can also assign multiple controllers to control one parameter, or one controller to control multiple parameters.

The original SampleTank had 16 buttons for selecting which multitimbral part to display for editing. SampleTank 2 displays all 16 parts simultaneously in the Mix window, and you select a part by simply clicking on it. In addition to the MIDI channel and Instrument name, each row in the Mix window lets you change the Preset's maximum polyphony, volume, pan, solo or mute status, and audio-output assignment. The MB column displays how much RAM a Preset uses, and clicking on a button at the top of the column displays SampleTank's total RAM consumption. Each part also has an Empty button for deleting the current Instrument, and a tiny graph displays the Part's output level.

Red is no longer the only color available for SampleTank 2's user interface. Tiny, continuously variable Color, Luminescence, and Saturation knobs let you choose from a seemingly infinite palette of colors and shades—even black and white.

JUST BROWSING

The Browser window is an improvement over the original, but it's still a little cumbersome, mostly because it's so small. Previously, you double-clicked on folder icons to reveal their contents and clicked on a single triangle to move up through the folder hierarchy. Now
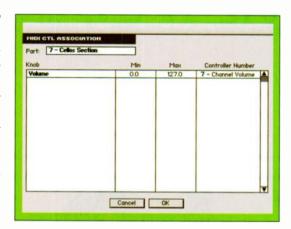
FIG. 2: To assign a MIDI CC to any onscreen knob, click on the MIDI Ctl button and then a knob. Scroll through the controller numbers in the pop-up window that appears to define minimum and maximum values.

you click on triangles that open and close directories within directories, as you would in the Macintosh Finder's List view (see **Fig. 3**).

To assist in locating the Instrument you want from a choice of hundreds, SampleTank 2 provides a Search function. Just click on the Search button and type in a keyword, and all the Presets associated with that keyword will appear in the Browser. If a factory keyword won't get you where you want to go, you can also assign user keywords by typing them into the appropriate field when an Instrument is selected. The Search function isn't perfect, though; when I searched for the word "old," expecting to find a Preset called Old Orchestra Recording, it wasn't among the four Presets that appeared in the list.

NOW I'M DOWN IN IT

Unlike its predecessor, SampleTank 2 can sync loops to sequencer tempo. To change tempo without transposing pitch, though, you need to switch the sound engine from straight sample playback (which IK Multimedia calls resampling) to Pitch-Shift/Time-Stretch (PS/TS). I was pleased that the sync function worked with user samples as well as factory loops.

However, as soon as I imported a loop containing pitched material and clicked the PS/TS button, I heard harmonic distortion and other undesirable artifacts. They became even more

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FIG. 3: SampleTank 2's Browser window reveals Instrument Presets within directories in a hierarchical fashion.

pronounced when I changed the sequencer's tempo. On the other hand, the artifacts were barely audible on drum and percussion loops. I wasn't surprised, then, to find that all the factory loops were drums and percussion. Obviously, PS/TS isn't meant for changing the tempo of pitched loops.

I was surprised to discover that if I set the tempo to a multiple of the loop's original tempo, the loop played at its original tempo rather than at the multiple. For example, if the loop's original tempo was 80 bpm and I set the sequencer's tempo to 40 or 160 bpm, the loop played at 80 bpm, regardless of whether it was a factory loop or one that I had imported. IK Multimedia is working on a solution.

I'd been looking forward to trying out SampleTank 2's much heralded other new sound engine, Stretch, since it was announced a few months ago. It's designed to smoothly transpose pitch with the tempo and formants intact, making it possible to play instant vocal harmonies, for example. Stretch works quite well as long as the source sample is a sustained monophonic tone. When I stretched one sample of a synthesized sax tone, for example, I was astonished when it transposed convincingly up and down the entire keyboard with no apparent artifacts (see Web Clip 1). When Stretch works, it's a remarkable feat of technology.

The moment I loaded up a loop and clicked on the Stretch button, though,

it was transformed into something that sounded similar but different, full of odd noises and thumps that weren't present in the original loop (see Web Clip 2). I was also disappointed that the stretching process introduced similar thumps to a sweeping synth tone that I'd sampled. Obviously, you need to experiment to find sounds that are appropriate for stretching.

CHECK OUT THE LIBRARY

Anyone composing music with computers appreciates quick and easy access to an array of sounds. SampleTank 2's library is vast enough to encompass a broad palette of traditional musical timbres that represent quite a few instrumental families and provide hundreds of individual Instruments. No matter which Preset you select, load time is minimal. The Browser divides Instrument Presets into 14 categories. You can also create your own categories by simply placing a new folder in the SampleTank Instruments folder on your computer desktop; likewise, you can rearrange factory Presets into whatever folders you prefer.

A Program Change Association window lets you assign a MIDI Program Change to any Preset. Simply click on an Instrument name and then on the Program Change button; a Program Change Association window will appear in which you can click and drag in a field to scroll through patch numbers.

Besides Bass, Drums, Guitar, Piano, and Organ, other categories include Brass, Percussion, and Woods and Winds. The Ethnic category features nonorchestral instruments that don't fit comfortably in other categories, such as accordions, harmonicas, mandolin,

and whistles, as well as world instruments such as bamboo and Indian flutes, kalimba, shakuhachi, and sitar. Strings are divided into Cello, Double Bass, Viola, and Violin folders that contain solo and ensemble instruments. Multistring ensembles are in the Orchestra folder, along with various full orchestras, orchestral percussion, and a few more instruments such as harpsichord and glockenspiel that are difficult to fit into other categories.

The quality of the samples is consistently excellent overall (see Web Clip 3). Five years ago, the pianos would have been some of the best you could buy, and they still sound quite good. I especially like the Acoustic Grand 1 and 3; to my ears, both have a very nice tone and a decent touch response, though neither responds realistically to playing forte. The Strings don't provide as many articulations as a dedicated string library, of course, but they sound very nice. Guitars are a mixed bag, and I was generally disappointed with the synths. For electronic sounds, though, I always prefer a synth plug-in to samples. SampleTank's useful collection of drum and percussion Loops covers a variety of musical styles and tempos (see Web Clip 4).

Most but not all of the Presets manage to avoid uneven multisampling. Some Instruments have certain ranges of pitch that stick out a bit, but none are terribly serious. As is often the case with sample-playback synths, at least half the Instruments are not multisampled as much as I would have liked. The transposition artifacts that result from too few samples were often obvious, especially for Instruments with sampled vibrato. Still, with more than

FIG. 4: Clicking on the Zone button displays an Instrument's keyboard mapping, but SampleTank 2 doesn't offer the means to edit split points.

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–Tim Hauser, Vocalist, Manhattan Transfer

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–Joel Rosenblatt, Spyro Gyra Drummer

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M - A U D I O

LUNA AND SOLARIS

A pair of low-cost large-diaphragm condensers.

By Karen Stackpole

Not long ago, M-Audio introduced its first large-diaphragm microphone, the Luna (see **Fig. 1**), a condenser with a classic lollipop-shaped capsule. Several weeks after releasing the Luna, M-Audio brought out the Solaris (see **Fig. 2**), a nearly identical microphone that adds
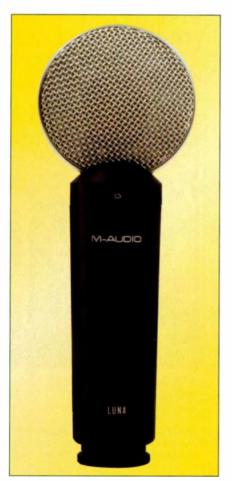
FIG. 1: M-Audio's Luna is a single-pattern large-diaphragm microphone designed for vocals, percussion, and stereo configurations.

three selectable polar patterns, a switchable –10 dB pad, and a highpass filter.

M-Audio recommends both microphones for use on vocals, guitar, piano, and other acoustic instruments, and for stereo recording. Naturally, I was curious to see how such affordable microphones (both are priced under $350) would hold up in professional applications.

THE SUN AND THE MOON

Weighing a substantial 1.4 pounds each, the Solaris and the Luna are built like tanks and feature sturdy machined-metal housing and Class-A FET electronics. In both microphones, the 1.1-inch evaporated-gold diaphragm sits inside a 3-inch brass capsule, and the open-weave grille is internally reinforced with a fine mesh screen. The XLR connector is recessed under the base of the body. The Luna has a black finish; the Solaris is a matte silver.

To identify the cardioid address side of the Solaris, the M-Audio logo is etched in black into the silver housing along with the model name. Below that are switches for the –10 dB pad and the bass rolloff. The rolloff slopes below 125 Hz at 6 dB per octave. A three-position switch on the back of the Solaris allows you to select the pickup pattern: omnidirectional, cardioid, or figure-8. The Luna simply bears the logo and model name in white on the address side.

I received a pair of the Solaris microphones and two pairs of the Lunas, which enabled me to check M-Audio's claim that the frequency responses of the microphones fall within ±1 dB of one another. Each microphone comes in an aluminum carrying case with rounded edges, textured sides, reinforced corners, and a lock, and each is accompanied by an SM-4 shock-mount in a separate cardboard box. I was pleased to discover that the shock-mounts are made completely of metal rather than plastic. Either microphone can be screwed securely into the mount at the base of the suspended four-posted basket.

The SM-4 allows for a single-axis,

FIG. 2: The Solaris is the same size as the Luna and nearly identical in appearance, but it offers three polar patterns, a –10 dB pad, and a highpass filter.

180-degree vertical rotation on a hinge with a smallish, angular metal knob that adjusts and holds the mic in position. The device is hard on the fingers, and it is difficult, though not necessarily impossible, to clamp it down enough to prevent the microphone from drooping.

Although the all-metal mount seems quite hearty, the elastic band on one of the mounts broke at the seam soon after setup when subjected to stress from an unintentionally pulled cable, and the microphone it had been holding dropped to the floor. It may have been a defect in that particular piece of webbing, but it alarmed me. It's a testament to the durability of the Solaris that the only damage it sustained was a slight ding on the edge of the grille, and more importantly, the microphone's operation wasn't affected. Misgivings about the SM-4 notwithstanding, the entire apparatus, mic

and mount, looks solid and has class-act appeal.

STUDIO TAN

I used the Solaris and Luna microphones to track male and female vocals, acoustic and electric guitars, and cello; as overheads and stereo room mics (in an XY-coincident setup) to record drums; and as close-mics on kick drum and floor tom. In addition, I put the mics through controlled loudspeaker tests to check consistency and features and conducted an A/B comparison with two popular midpriced large-diaphragm condensers, the Blue Dragonfly and the AKG C 3000 B. While the comparison microphones are both pricier than the M-Audio mics, this test helped me evaluate the sound quality of the review microphones.

Steve Orlando, a fellow engineer and instructor at Ex'pression Center for New Media, helped me out with testing during several sessions with his band, the Jingle Punx. We put the microphones through a Studer D950S console and a Digidesign Control|24. I also performed several tests using an FMR RNP8380 Really Nice Preamp at Guerrilla Recording with Myles Boisen, and at my own studio through an Allen & Heath WZ16:2DX. I recorded the results to Digidesign Pro Tools (at 24-bit, 44.1 kHz resolution); to 16-track, 1-inch analog tape; to 20-bit ADAT; and to CD-R.

Vocals. Both M-Audio mic models performed well on male and female vocals for punk, funk, and jazz. The subtle high-end edge of the microphones enhanced the grit of the male punk-anthem vocals and lent some pop to the male funk vocals. However, the Solaris and the Luna both emphasized an undesired raspy quality in the smoky voice of a female jazz singer.

The highs of the Solaris weren't quite as smooth as those of the Dragonfly and the C 3000 B, but the overall sound was clean and clear. The Solaris was especially nice on a soft-spoken female vocalist reading text, because it provided good definition on a quiet, smooth voice without emphasizing sibilance.

Drums and stereo. The mics provided significant definition to the cymbal crashes and hi-hat when used as overheads, although the emphasis in the high end sounded a little harsh at times. While the mics weren't the most flattering on bass drum, they sounded absolutely fantastic on the floor tom, capturing the rich tone of the drum and the crisp attack of the sticks on the head.

I set up two Solaris mics as an XY stereo pair (using the cardioid pattern setting on both) in the drum room during a session, to capture the overall sound of the kit. The large size of the lollipop grille prevented me from getting the capsules as close together as I usually prefer, and it took some time to overcome the drooping shockmount. But once they were set up, the mics performed well.

The cardioid pattern of the Solaris is fairly wide, so it picked up a good deal of the room, which worked quite well for the sound of the kit. The snare and the hi-hat sounded a little bright, but the attack on the kick drum was well defined, and the low end was beefy and well represented. With its multiple patterns, the Solaris would be an excellent microphone to try in middle-side and Blumlein-array stereo-miking configurations.

Guitars and strings. The Solaris really shone on classical guitar, capturing the instrument's rich tone with plenty of depth and definition. It actually sounded better than the mics from Blue and AKG for this application during the comparison tests. And while it offered slightly less sparkle than the Dragonfly on a steel-string guitar, the Solaris still picked up plenty of tone and pick sound. Placed in front of the amp of a distorted electric guitar, the Luna picked up the full, chunky tone and plenty of grit.

The Luna's high-end edge was a bit harsh on acoustic cello. However, the overall tone was good and definitely usable.

IN THE LAB

To closely check out the features and consistency between the microphones, I set up a controlled loudspeaker test in my studio, placing all six M-Audio mics in a tight cluster about six feet in front of a speaker system. Most of the microphones sounded very similar, but one of the Solaris and one of the Luna mics exhibited an audible difference in high and midrange response. However, these two microphones were similar in response to each other.

When switched to omnidirectional, the Solaris had an even pickup around the microphone, and the sound was open and clear. The figure-8 pattern setting offered good off-axis rejection and gave a similar response on either side of the capsule. The cardioid pattern is fairly open, and I didn't notice

Luna and Solaris Specifications

Element	condenser
Polar Pattern	Luna: cardioid; Solaris: cardioid, omnidirectional, figure-8 (selectable)
Frequency Response	20 Hz–20 kHz
Output Impedance	200Ω
Self-Noise	14 dBA
Maximum Input Sound Level	130 dB (@ 0.5% THD)
Highpass Filter (Solaris only)	–6 dB @ 125 Hz (switchable)
Attenuation Pad (Solaris only)	–10 dB (switchable)
Power Requirements	48V phantom power
Accessories	SM-4 shockmount, aluminum case
Dimensions	3.00" (W) × 8.25" (L) × 2.00" (D)
Weight	1.4 lb.

a lot of proximity effect when the microphone was close to a source. The low-cut filter was subtle yet effective, as was the 10 dB pad.

SUNRISE, SUNSET

I'm impressed with the full sound and good definition of the Solaris and the Luna. Although I could hear a bit of an edge in the high-end response that sounded a bit harsh in some instances, the overall performance of these microphones is solid and clean. The Solaris in particular is quite versatile, with its three patterns, the pad, and the low-cut switch. The Solaris and Luna are attractive options for budget-conscious project studios and for engineers looking to expand their mic palettes without draining their resources.

Karen Stackpole *teaches sound arts at the Ex'pression Center for New Media and operates Stray Dog Recording Services. Special thanks to Steve Orlando, Myles Boisen, and Ann Dentel.*

"CLARITY!..."

Howard Burke, Front of House, Little Feat

AUDIX OM-5

"WITH 3 STAGES RUNNING SIMULTANEOUSLY 7 NIGHTS A WEEK, IT'S COMMON FOR US TO HAVE 15 OR MORE SEPARATE ACTS IN A SINGLE NIGHT. WHEN THE SMOKE CLEARS, IT'S THE OM-5 YOU SEE ON THE MIC STANDS. THE OM-5'S INDESTRUCTIBLE CASING IS ESSENTIAL, GIVEN THE UNENDING WEAR AND TEAR WE PUT OUR GEAR THROUGH, AND THE WIDE DYNAMIC RANGE IS PERFECT FOR THE VARIETY OF VOCALISTS WE HAND IT TO."
BRUCE DUFF
PRODUCTION MANAGER, KNITTING FACTORY HOLLYWOOD

VOX AMPLIFICATION

TONELAB

A modeling processor that packs a punch.

By Orren Merton

The field of desktop guitar-amp modelers has become quite crowded of late. It seems as though each new unit promises more emulations, more effects, more connectivity, and a lower price. Vox's new ToneLab—a direct descendant of the company's Valvetronix modeling amplifiers—takes a different approach. Instead of trying to outdo its competitors with sheer number of sounds, effects, and connectors, Vox has opted for superior quality. The ToneLab is designed to provide highly accurate emulations and to deliver real tube-amplifier reactivity between tone and player—undoubtedly the holy grail of modeling amp features.

BIG, BLUE, AND BOXY

The first thing you notice about the ToneLab is that it's not your average swirly-shaped, plastic-housed modeler. The solidly built unit is over a foot wide, with unpainted metal sides and bottom and a blue-metallic front, back, and top. Its array of chicken-head and vintage-style rotary knobs give it something of the look and feel of a classic amplifier (see **Fig. 1**). The gold-on-blue graphics are reminiscent of modern Vox amplifiers, and the staggered, diagonal knob positions give the unit a distinctive appearance.

Form follows function on the ToneLab; the left-to-right order of the dials and buttons corresponds to the unit's signal flow. The small display screen sits above ten programming and function buttons (actually called the Bank/Manual/Write/Tuner/Channel Selection buttons) at center right. In the top right corner, a clear plastic window gives a glimpse of the 12AX7 tube that powers the ToneLab's Valve Reactor power amp (which I will discuss further in a moment).

The two connectors on the front of the ToneLab are a ¼-inch stereo headphone out and a ¼-inch mono guitar-

input. Both of these connectors are securely bolted onto the chassis and look capable of withstanding any amount of abuse.

The rear of the unit houses the remainder of its connections: Left/Mono and Right ¼-inch output jacks, an S/PDIF digital output, a Vox bus pedal-controller output jack, and MIDI In and Out jacks. There's also an output-level knob, a Line/Amp level selector switch, a Standby (power) button, and a ventilation port for the tube (see **Fig. 2**).

Although the level knob, switches, and buttons feel secure enough, none of the rear jacks are bolted to the chassis. No doubt this was done to conserve space and save money, and in fairness, the outputs shouldn't see nearly the amount of cable swapping that their front-panel counterparts do. Even so, I would prefer to have all the connectors bolted on.

I also wish there were a word-clock input, which would have made the digital output far more useful. As it stands, the S/PDIF output has limitations for professional applications, because unless your hardware offers on-the-fly sampling-rate conversion, the ToneLab must be the clock master, and it functions only at 44.1 kHz.

IN THE LAB

Operating the ToneLab is straightforward. The 13 rotary knobs on the face of the unit adjust the models and their parameters—there's no need to go digging in hidden menus to tweak your sound. The Pedal, Modulation, Delay, and Reverb effects sections share the same three knobs to control their adjustable parameters. (With the press of a button, you select which section's parameters are active.) A silk-screened chart above the knobs clearly shows which parameters are available for editing. Finally, there's a dedicated Tap button for programming delay time manually; this button also functions as an effects bypass when it's held down.

The ToneLab has 96 rewritable programs (48 of which are factory presets), broken up into 24 banks of 4. The programming and program-select buttons
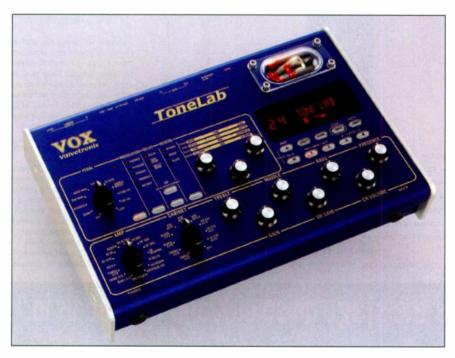
FIG. 1: The Vox ToneLab's vintage-style knobs and buttons offer convenient control of the unit's amp, cabinet, and effects models.

are in two rows of five buttons beneath the LED screen. The leftmost buttons let you scroll up and down through the program banks, and the four remaining buttons in the bottom row are numbered to let you choose a program from within the current bank.

The button scheme makes it easy to switch between programs within a group, but bank switching—especially if you're switching between noncontiguous banks—requires a little more effort. However, holding down a bank button puts you in Fast Scroll mode, which speeds up the process.

The Utility button allows users to adjust the noise gate, give names to programs, and set up pedal parameters for one of the optional foot controllers (see the sidebar "A Choice of Pedals"). The Global button accesses MIDI channel and dump functions as well as the S/PDIF level.

I wish that Vox had included a function to let you do an A/B comparison of the current edited parameter value with its original value from the saved or factory preset. The display does have an indicator light to let you know that you've returned a knob to its original state, but by going back you lose your edited setting. It would be nice to have a button that easily compared the two. (As a work-around, you can save your edited preset to a new location and compare by toggling between the presets.)

The LED display is relatively spartan. An eight-character top line shows program and parameter names. The lower line tells you the Valve Type of the original amplifier being emulated, whether the current parameter value is the original or an edited value, and what the current parameter value is.

If you prefer computer-style editing, the Vox Amps Web site (www.voxamps

.co.uk) offers a useful ToneLab editor-librarian application free for the PC. Unfortunately, there is no Mac version.

THE VALVE REACTOR

Explaining the technology of the ToneLab's Valve Reactor section requires a bit of background regarding the interaction between a guitar and an amplifier. A preamp alone does not generate the tone of a guitar amp. The ways the tubes of the power amp interact with each other and with the playing style of the guitarist—aka the "feel and touch dynamics"—are integral components of the final sound.

As any tube-amp aficionado will attest, the feel and touch dynamics of a tube amplifier depend on the number of tubes, on whether the amp is a Class A (meaning the tubes receive a constant current) or Class AB (meaning the tubes receive current in a "push-pull" grid), on the size of the output transformer, and so forth.

While other modelers on the market attempt to approximate this interaction digitally, Vox has gone the opposite direction, using a 12AX7 (sometimes referred to by its British name, ECC83) tube as a functional 1W power amp. And because the 12AX7 is a "dual-triode" tube (meaning it can function as two tubes in one), the ToneLab uses it as both a 1W Class-A and a 1W Class-AB power amplifier. After the power amp, ToneLab features what Vox calls a Virtual Output Transformer. So not only does the Valve Reactor model the sonic characteristics of the tube amps it emulates, it delivers actual interaction between the tube power section and the output transformer as well.

STAGES OF GAIN

One significant effect of the Valve Reactor technology is that the gain staging on the ToneLab is far more complex than on other modelers. In addition to the master Output Level knob on the rear of the unit, the ToneLab has three parameters that affect the final output level and sound: Gain, VR Gain, and CH Volume. The parameters' functions change depending on the model selected. In the emulations of vintage amp models that didn't offer a master

A CHOICE OF PEDALS

The ToneLab offers excellent expression-pedal functionality. Pedals can be used to continuously control wah-wah, volume, modulation speed, flanger frequency, and delay or reverb level within user-specified boundaries. If you're using one of Vox's two optional foot controllers, you can also select programs, turn individual effects on or off, activate the tuner, and control delay time using a tap tempo button.

The less expensive of the two controllers is the VC-4 ($200). It lets you choose between 16 program presets and to turn individual effects on

FIG. A: The VC-12 is the more fully featured of the ToneLab's two optional foot-controller pedals.

or off, and it offers a dedicated tap-tempo button and expression pedal.

The VC-12 has even more features ($350; see **Fig. A**). This solid-steel pedal has the same functions as the VC-4, and it gives you control over 48 presets, a tuner, and a second expression pedal.

volume, the Gain knob functions as a volume control, and the VR Gain serves as the volume into the Valve Reactor power-amp circuit.

For models of amplifiers that did offer a master volume knob, the Gain knob affects preamp gain, and the VR Gain affects the power-amp gain. The CH Volume knob acts as the attenuated output—in other words, the output that your speaker or line input sees. Although this control scheme might seem confusing at first, it makes perfect sense when you start using the ToneLab, and it adds to the realism of the models.

SHOOT-OUT

To test Vox's claim of authenticity, I compared the ToneLab with a vintage Fender Blackface Twin, a 1968 Marshall JCM Plexi, a modern Marshall DSL50, a '90s Mesa/Boogie Dual Rectifier, and a modern reissue Vox AC30—all amps that the ToneLab emulates. For these tests, the ToneLab was patched into my recording rig and played back through my Dynaudio BM6A active studio monitors.

I didn't expect the ToneLab to be sonically identical to the tube amps—even two Plexis won't sound exactly the same—but I wanted to see if it captured the texture, feel, and interaction of the originals. The well-written and comprehensive manual was a huge help in making a fair comparison possible. It lists not only which amps were emulated by the ToneLab's models, but also which specific inputs and channels were simulated. This information allowed me to duplicate the channel and input assignments when comparing the real amps to the ToneLab.

ToneLab's Blackface 2×12 captured the tone and feel of the Fender Twin (see **Web Clip 1**), although it sounded a bit thinner. The ToneLab's Plexi model—Vox calls it UK68P—was also quite good (see **Web Clip 2**). The feel was dead on, and the tone captured the most recognizably "Marshall" characteristics. The Marshall DSL and Vox AC30 (see **Web Clip 3**) were emulated nearly perfectly—I was hard pressed to tell the difference, even when staring at the unit being tested. I can't imagine trying to tell the difference in a blind test! ToneLab's Recto emulation accurately captured the feel and distortion characteristics of a Mesa/Boogie Dual Rectifier (see **Web Clip 4**), although I didn't feel the ToneLab reacted to different preamp settings the same way the Mesa did.

Most importantly, the ToneLab was more reactive than any other modeler I've tried. It felt quite touch sensitive, and the accuracy of the amp models didn't change as I rolled off the volume or switched guitars and pickups. I was impressed.

CALL ME A CAB

I'm always a bit suspicious of cabinet emulations. Considering that even with the same speaker configurations (including speakers of the same model), cabinets from different manufacturers sound completely different, how accurate can any simulation really be? Vox was smart enough to specify in the manual the precise brand of cabinet as well as speakers used in their various emulations.

I compared the ToneLab against two actual cabinets: a custom Black 2×12 with Celestion G12H30s and a Marshall UKV30 4×12. (For these tests, I first listened to the ToneLab—with its cabinet simulation on—plugged in through my studio monitoring system. I then compared that with the sound of the ToneLab—with the cab simulation turned off—plugged directly into the real guitar cabinets.) The ToneLab's simulations were better than what I've heard on other modelers, but still a bit thinner-sounding than

ToneLab Specifications

Inputs	(1) ¼" unbalanced
Outputs	(2) ¼" balanced/unbalanced; (1) ¼" TRS headphone; (1) RCA S/PDIF (44.1 kHz)
Maximum Input Level	+5 dBV
Additional I/O	MIDI In, Out/Thru; Vox bus (footpedal)
Presets	48
Amp Models	16
Effect Models	21
Mic/Cabinet Models	16
A/D/A Converters	20-bit
Sampling Frequency	44.1 kHz
Power Supply	9 VAC; 3.0A line-lump power supply
Dimensions	12.56" (W) × 3.11" (L) × 8.39" (D)
Weight	5.5 lb.

the real cabinets. However, choosing a matching amp and cabinet simulation (such as the Vox cabinets with Vox amps or the Black cabinet with the Blackface simulation) made the simulation even more effective.

On the other hand, turning off the speaker simulator and running the ToneLab through a power amp (I used the power section of a Rivera Knucklehead II) and a cabinet gave me an excellent sound. Even if it didn't precisely match the specific modeled amp, it did sound like a tube amp through a power amp, not like an amplified processor.

DONE FOR EFFECT

Like the amps and speakers, the ToneLab's effects are not simply digital multi-effects, but emulations of classic guitar pedals and effects. There are four effects sections: Pedal, Modulation, Delay, and Reverb. You have no control over the order of these effects,

and they all come before the amp and speaker simulations. You can, however, use any combination of the four sections simultaneously. The ToneLab also allows you to bypass any or all of the effect sections.

I didn't have access to all the original stompboxes for an A/B test, but I would definitely say that the Vox Wah and U-Vibe (Univibe) effects sounded quite good. I wasn't that impressed with the sound of the Fuzz effect, and the rest of the compressors and boosts in the Pedal section ranged from decent to good.

The Chorus, another standout effect, sounds very rich. The other Modulation effects (Phaser, Flanger, Tremolo, and Rotary) and the Delay sound good in the context of the various amp models—they sound more like guitar effects than digital effects. That's a good thing for a guitar modeler.
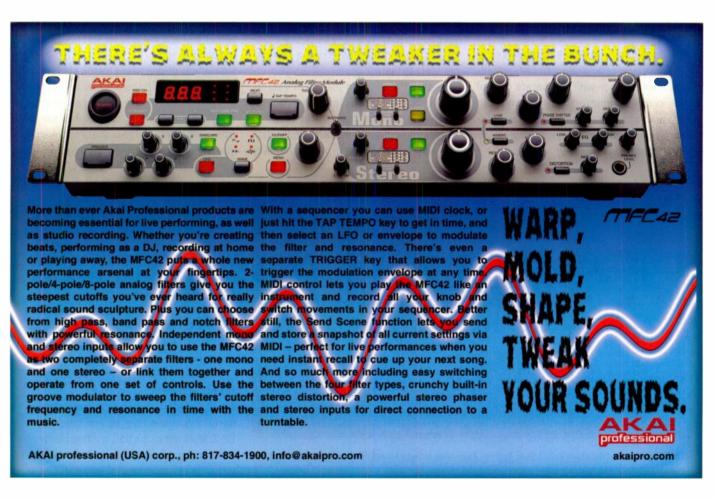
The Delay is useful, especially considering that you can control it with the Tap Tempo button on the face of

the unit. The Reverb section—which includes Spring, Room, and Plate emulations—was my least favorite of the effects groups. To my (admittedly picky) ears, the Reverb models lacked the density of others I've heard.

A MODEL CITIZEN

The Vox ToneLab is solidly built and easy to use. What it lacks in bells and whistles it makes up for in quality. If your budget is limited and you're looking to squeeze in as many features and sounds as possible for the lowest price, you might be happier with a different unit. But if authentic amplike tone and feel are what you're after, the ToneLab is sure to satisfy. **EM WEB CLIPS**

Orren Merton is the author of Logic 6 Power *(Muska & Lipman, 2003), and can usually be found playing guitar through one of his many of vintage and modern amps.*

APPLIED ACOUSTICS SYSTEMS

Lounge Lizard EP-2 (Mac/Win)

By Nick Peck

● ●

Applied Acoustics Systems has released a new version of its Lounge Lizard virtual piano, updating it from EP-1 to EP-2 ($249). Like the original, EP-2 emulates electromechanical pianos using physical modeling, which sounds realistic and lets you change timbre radically as you're playing.

Several versions of Lounge Lizard are available. The cross-platform CD is compatible with Windows 98, 2000, ME, and XP and with Mac OS X 10.2 or later. For Windows, VSTi and DXi plug-ins and a standalone version are supplied, and Mac users get Audio Units, RTAS, VST, and standalone versions.

No Nonsense

EP-2's slightly retro-looking interface is straightforward and uncluttered, providing a single page with a separate control for every parameter. The controls are grouped and laid out coherently by function, which makes it simple to program the instrument. Every control is mapped to a MIDI CC value, allowing you to tweak the tone in real time from your MIDI keyboard or controller. A Learn function lets you easily reassign MIDI CCs.

EP-2 offers four groups of parameters for tweaking the mechanics of the modeled piano: Mallet, Fork, Pickup, and Release. The Mallet parameters model the behavior of the mallet that strikes the tine when a key is pressed. The Fork parameters model the metal bar that generates sound by ringing when struck by the mallet. The Pickup parameters model the magnetic coil that transduces the fork's vibration into an electric signal; you can adjust the pickup's position and amplifier characteristics. The Release controls affect the behavior of dampers that mute the fork when the note is released.

Each group of parameters is subdivided into smaller groups that model particular characteristics. For example, the Mallet group is further broken down into Stiffness, Noise, and Force subgroups, each of which has a few controls of its own. All told, the Mallet section has 10 parameters, and the complete piano model has 24 different settings. That may sound a bit daunting, but I got good results by simply selecting a factory patch and then tweaking a few parameters to taste.

Just as the electric pianos of yesteryear were often connected to effects pedals, EP-2 features four effects wired in series: wah, phaser, tremolo, and delay. The wah is an LFO-controlled bandpass filter sweep, often called an auto-wah. It sounds good, but an envelope follower function would be a welcome addition. The phaser adds some nice animation to the signal but is mono only. Tremolo adds amplitude modulation with two LFO shapes. Tremolo does have a stereo mode, which enhances the sound considerably by adding a wide soundstage. The delay effect has individual delay-time and feedback settings for the left and right sides, but no modulation parameters; delay times range from 10 to 1,500 ms. All four effects sync to MIDI, allowing you to control modulation or delay time from a host sequencer. Last in the signal chain are bass and treble shelving-EQ knobs and a big master Volume knob.
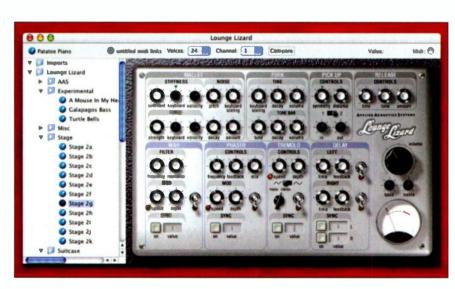
In Practice

I was pleasantly surprised at the realism and depth of many of the Rhodes patches. Both Suitcase and Stage models are portrayed, capturing the velocity-based gritty growl that is missing from many sample-based Rhodes simulations. EP-2's understated, groovy character truly evokes the real instrument. In a recorded setting, I would be hard pressed to distinguish between EP-2 and a real Rhodes. In contrast, simulations of Wurlitzer electric pianos are basically variations on the Rhodes sounds and not dead ringers of the real instruments. Additional factory patches range from new electric pianos that don't sound like Rhodes instruments to otherworldly synthlike chimes and ambiences.

EP-2's latency was very low on my dual-processor Power Mac G4/867 MHz. When I used the RTAS version of the plug-in, I was delighted to see that all parameters were available for editing from my Digidesign Control|24. Unfortunately, I had several crashes and MIDI glitches using the standalone and RTAS versions from within Mac OS X. However, I never resolved whether the problem was in my MIDI interface, the Mac's low-level MIDI software, or Lounge Lizard itself.
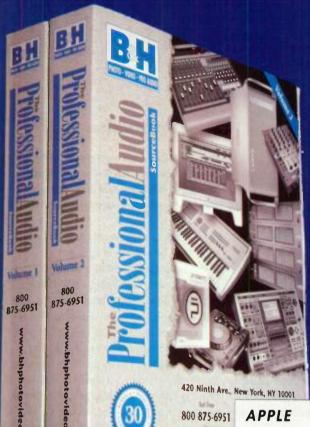
Hands Down

Applied Acoustics has developed a wonderfully realistic Rhodes emulation with EP-2. Right out of the box, it offers plenty of parameters to tweak, good effects, and lots of great-sounding factory patches. If

The plug-in and standalone versions of Applied Acoustics Lounge Lizard EP-2 capture the sound and the spirit of the Rhodes and other electromechanical pianos.

you want to add a Rhodes to your recording rig without the hassle and maintenance of the real thing, Lounge Lizard is worth a serious listen.

Overall EM Rating (1 through 5): 3.5
Applied Acoustics Systems DVM Inc./Ilio (distributor); tel. (800) 747-4546 or (818) 707-7222; e-mail info@applied-acoustics.com; Web www.applied-acoustics.com or www.ilio.com

MASTERWRITER
MasterWriter (Mac/Win)
By Michael Cooper

● ●

A module button at the top of the screen lets you access each of MasterWriter's dictionaries. Here, a search for imperfect rhymed phrases for *night* yields well over a dozen common phrases.

MasterWriter ($289), offered by a company of the same name, is a comprehensive collection of songwriting tools for the Mac and PC. The software mostly aids in writing lyrics, and it includes separate dictionaries containing over 100,000 rhyming words, 36,000 rhyming phrases, 33,000 nonrhyming phrases, and 11,000 references to pop-culture icons. Also included are an alliterations dictionary, the *American Heritage Dictionary*, *Roget's II* thesaurus, a database for storing both creative data (such as tempo) and administrative data (such as publishing information) for your songs, and a rudimentary hard-disk recorder for saving your melodies and arrangements.

MasterWriter requires either a 500 MHz or faster Mac G3 running OS 9.2.2 or OS X 10.1.5 or higher (10.2 is recommended), or a Pentium III or faster and Windows 98SE, 2000, ME, or XP. Whatever your platform, you'll need at least 128 MB of RAM (MasterWriter recommends 256 MB) and QuickTime 6.0.2 (which is included). I used MasterWriter 1.0.2.25 with a dual-processor Mac G4/867 MHz running OS 9.2.2.

Search Me
After you type some lyrics into MasterWriter, you can select any word and perform a search for a rhyming word or phrase or find a nonrhyming phrase, definition, synonym, or antonym for that word. Search results include perfect and imperfect rhymes. MasterWriter even provides rhymes to slang pronunciations (using *achin'* instead of *aching*, for example). To preserve the meter of your lyrics, you can filter your search results to show only words containing a specific number of syllables.

The separate pop-culture dictionary displays culturally significant names, places, people, events, and products that rhyme with your selected word. Using the dictionary of rhymed phrases, a search for the word *night* reveals dozens of fragments, including "by the dawn's early light." Yet another dictionary yields an abundance of non-rhyming phrases that include your selected word (such as "a night on the town"). MasterWriter plans to offer free annual updates to the phrase and pop-culture dictionaries.

For each word you search, the rhymes you subsequently choose to keep track of are organized into a separate Favorites list, with one list for each word search. Every word for which you performed a search—in any of the dictionaries—is subsequently available in a drop-down menu, providing a useful history of searches you can revisit.

Beyond Words
MasterWriter includes a bare-bones hard-disk-recording feature that's useful for spontaneously recording melodies and arrangements (using your computer's built-in mic or an external mic). It can also import MP3, WAV, AIFF, or QuickTime Movie files. MasterWriter records mono or stereo audio files at sampling rates of 11.025, 22.05, or 44.1 kHz, and it can export them in WAV or AIFF format. The recorder provides basic transport and loop controls, markers, and more than 250 MIDI drum loops. The drum loops encompass various performance styles, and you can adjust their tempos.

Because MasterWriter supports neither multitrack recording nor punching, its recorder is no substitute for a DAW. However, it provides an alternative to using a cassette recorder for saving quick sketches of your songs. It is also useful for prepping songs for registration with MasterWriter's Web-based Songuard service. Songuard is a date-of-creation registration service that offers protection for your lyrics and melody during the development stage, before you register them with the Register of Copyrights. A one-year subscription to Songuard comes with your purchase of MasterWriter.

The included 18-minute Flash demo covers all of MasterWriter's main features and, combined with the onboard Help, serves as the program's operating manual. I liked that the Help window could stay open while I worked on song lyrics and that it would automatically update to show relevant content as I navigated through MasterWriter's various modules (windows).

Acid Test

Whenever I tried to rewrite a certain song over the past year, I got stuck on one phrase I knew could be stronger. MasterWriter gave me three pages of rhymes, and one inspired the perfect substitute phrase for my song. That took the first verse to a higher level.

MasterWriter's greatest strength is that it aids helps you stay with the creative muse instead of switching your brain to analytical thinking to come up with rhymes. The program is comprehensive, well organized, and easy to learn. Its price is hefty, but ASCAP, BMI, SESAC, and NSAI members receive a professional discount. I'm sold!

Overall EM Rating (1 through 5): 4.5
MasterWriter; tel. (866) 892-8844; e-mail support@masterwriter.com; Web www .masterwriter.com.

DISCOVERY FIRM
Bom Dia Brazil (Acidized WAV, REX2)
By Marty Cutler

· ·

Elements of Brazilian music permeate much of our jazz and pop landscape. Many of us instinctively equate Brazilian music with the samba, but Brazil is rich in musical
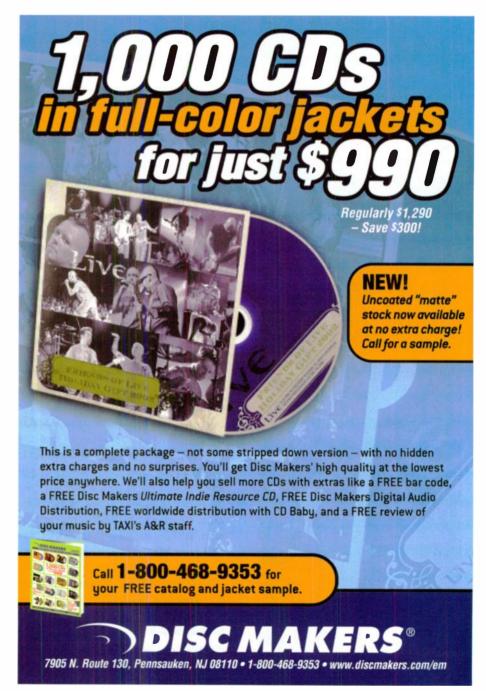
variety: hybrids of Portuguese, European, and African music abound. With the fervor of a dedicated ethnomusicologist, Discovery Firm delves deep into Brazil's musical heritage in *Bom Dia Brazil* ($55), a CD-ROM filled with 16-bit, 44.1 kHz loops of traditional Brazilian instruments and grooves.

The single CD-ROM features a folder each of Acidized WAV and REX2 files, covering two significant bases for importing and playing the loops. Subfolders contain an assortment of grooves and single-

instrument percussion loops. The colorful booklet gives the package a folky vibe, but don't let that lower your expectations: much of the content would serve well in jazz and pop compositions. I tested the files on my dual-processor Mac G4/1.42 GHz using MOTU Digital Performer 4.11, Ableton Live 1.5, and Propellerhead Reason 2.01.

Different Drummers
For each format, you get five main folders: Brazilian Groove; Cabacal Style, which is a

festive parade-style music populated with flutes and multiple types of percussion; Capoeira, which derives from the Brazilian martial-arts practice; a folder of individual-instrument percussion loops; and a folder of outtakes. Within each of the 15 Brazilian Grooves subfolders are three to eight loop variations representing different styles from various regions of the country, including *baiao, carimbo, choro, coco, forro, maracatu*, and four different types of samba. Although most are full drum-kit grooves, some of the loops have traditional percussion instruments mixed in.

Among the single-instrument percussion loops, my favorites come from the thunderous sound of the *zabumba*, a double-headed drum played with a mallet on one side and with a stick on the other side to produce higher pitches. I also liked the low, buzzy *surdo* and the *repinique* loops. Other looped percussion includes the *berimbau*; the *tamborim*; shaker-type instruments such as the *caxixi*, the *ganza*, and the *chequere*; and talking drums.

Among the Brazilian Grooves, I loved the rolling half-time feel of the Maracatu files. The Angola Capoeira files, with their vocal chants and marchlike feels, are spooky and atmospheric. The Capoeira Maculele loops manifest the music's African influence with a driving three-against-four rhythm. Generally, the loops have a minimally perceptible ambience, allowing them to sit well in tracks with little or no need for added reverb.

I was eager to test these files on a guitar-oriented piece I composed in the Latin-flavored style of the David Grisman Quintet. The samba grooves had the wrong feel for the song, so I navigated to the Baiao subfolder. Baiao is a syncopated style that originated in Northeastern Brazil and was popularized by Luis Gonzaga. I replaced my MIDI-programmed drum and percussion tracks with three REX-file variants of baiao grooves, and the track immediately sprang to life, taking on new dimensions of forward movement (see **Web Clip 1**).

Bom or Bomb

As much as I love this collection, I must also admit that it has some major flaws that limit its usefulness somewhat. To

Whether you're working with jazz, pop, or Latin styles, the wide range of grooves on Discovery Firm's *Bom Dia Brazil* may be just what your project needs.

begin with, the documentation may generously be described as sketchy. The booklet lists the contents by single-instrument loop and groove folders and gives the number of files in each folder, but that's all.

The worst thing about *Bom Dia Brazil* is the complete absence of tempo information for the files. If you are lucky enough have a digital-audio sequencer that can automatically adjust or reveal the original tempo of a file—as Sony Pictures Digital Acid and Ableton Live are able to—this won't be a problem for you. Otherwise, expect to have to do some educated guessing when you're working with the WAV files.

In addition, a few of the REX files are poorly sliced. When I dropped them into tracks, I noticed huge gaps between the slices that resulted in intermittent playback. That's too bad, because otherwise the quality and variety of the content make it extremely useful.

Nevertheless, *Bom Dia Brazil* provides a veritable aural education with its well-recorded examples of authentic Brazilian roots music, including many esoteric styles—and they groove like crazy. Only the poor documentation and sloppy editing prevent me from awarding this project **EM**'s top rating. ⊛

Overall EM Rating (1 through 5): 3

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132 Electronic Musician April 2004

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BUSINESS OPPORTUNITIES & CONSULTING

CASES

The MOTU Virtual Instrument Studio

"...another winner for MOTU."
— *Electronic Musician Magazine*

Electronic Musician 2004 EDITORS CHOICE

EQ AWARD

"Digital Performer 4 is better than ever."
— *Electronic Musician*

Electronic Musician 2004 EDITORS CHOICE

"MachFive is unquestionably a winner."
— *Electronic Musician*

Electronic Musician 2004 EDITORS CHOICE

Use DP4, the 828mkII and the G5 Power Mac to run MachFive, MX4 and a wide array of virtual instruments

Apple Power Mac G5
The world's fastest personal computer

Your all-native MOTU desktop recording studio just got bigger. A LOT bigger. The new Power Mac G5 is like doubling your studio's square footage, and then adding several additional floors. Run more virtual instruments, more plug-in effects, more tracks, more busses, more processing, more everything than you ever thought possible. Yes, it's time to bask in the glory of your MOTU native studio. Starting at just $1999, the G5 Tower transforms DP4 into a production powerhouse.

MOTU MachFive™ and MX4™
Put this universal sampler & unique multi-synth in your DP4 arsenal

MachFive could very well be the last sampler you ever purchase. Consolidate all of your sample libraries for instant access in DP4, including Giga, Sample Cell, Akai, K2xxx, Roland and others. Play up to sixteen different parts per instance, and apply automatable synthesis effects powered by the renown UVI™ engine. To add vintage and fresh analog synth sounds, look no further than MX4, MOTU's new multi-synth. Inspired by legendary subtractive synthesizers, MX4 combines several core synthesis techniques in a unified, hybrid synthesis engine that delivers fat basses, nasty leads, analog pads, vintage electronica — it's all at your fingertips.

Studiologic TMK-88
88-note semi-weighted MIDI keyboard controller

Don't let the size, weight, and price fool you. The Studiologic TMK-88 is a professional MIDI keyboard controller for serious MOTU users, but the amazing price makes it a technological breakthrough that's affordable for a MOTU studio of any size. Boasting full size keys with semi-weighted piano action, this is an 88-note MIDI keyboard controller that weighs in at a mere 13 lbs. Its lightweight and highly durable design makes it perfect for studio or stage. The keyboard action is light, but nicely weighted giving it an expressiveness that must be played to be appreciated. It is the perfect hands-on control for your MOTU studio virtual instruments. The TMK-88 is velocity sensitive with a mod wheel, one MIDI output, program change and bank select. Suggested retail is $399.95.

CALL **Sweetwater** music technology direct
800-222-4700
Authorized Reseller

UltimateSoundBank Charlie™
Virtual retro organ module

Charlie delivers famous electric organ sounds to your Digital Performer desktop studio via a 3 GB sound library that captures the real sound quality of genuine organs recorded with the vintage equipment favored by purists. Charlie is powered by the UVI Engine™, allowing you to play complex parts with unlimited polyphony. A gorgeous, clearly-designed, feature-rich synth interface, including amazing filters and mono/legato modes, lets you customize the sounds, or completely mangle them. Most patches are available with slow and fast rotary speaker effects. Use real-time MIDI control of every parameter to enhance expression and live use. Included instruments cover the gamut of vintage, classic American and European organs. Enjoy unsurpassed realism for your DP4 organ tracks.

Native instruments Pro-53™
The second coming of a virtual synth legend

The PRO-53 carries on the tradition of the legendary days of vintage cult synthesizers. Fashioned after the unique original Prophet Five, the PRO-53 casts in software those qualities which have been a major influence on popular music in the past twenty years: brilliance, power, warmth and beauty. Through Native Instruments' creative development philosophy, these timeless aesthetics have now reached the next step in their evolution. The result is the manifest re-definition of an original that was regarded as unrivalled until now. Call Sweetwater and add Pro-53 to your DP4 studio today.

East West / Quantum Leap Symphonic Orchestra™
World class strings / woodwinds / brass / percussion

This stunning new 24-bit orchestra sample library was recorded in a state of the art concert hall by GRAMMY award-winning classical recording engineer Keith O. Johnson with custom-designed recording equipment. And now it can be at your fingertips in DP4. Just open the included Kompakt™ sample player and then mix together any combination of three recorded mic positions (close, stage and hall) to alter the tone and ambience of any instrument or section. For example, you could use the stage mics for that big Hollywood sound, boost certain instruments with a hint of the close mics, and bring in a touch of the hall mics for reverb or even surround mixing. This library was produced by Doug Rogers and Nick Pheonix, recipients of over 30 international awards.

Arturia CS-80V™
Reproduction of the legendary Yamaha® CS-80 polysynth

In 1976, Yamaha introduced the CS-80. The price tag ($6,900) put it out of reach of most musicians... But the qualities of the CS-80, considered as Japan's first great synthesizer, made it immediately famous in the music industry. Popularized in the late 70's and early 80's by artists and groups like Toto, Jean-Michel Jarre, Keith Emerson, Stevie Wonder and Vangelis, the CS-80 soon became myth. The Arturia CS-80V offers all the features of the original synth, plus a new generation of innovative features. Does CS-80V really sound like the original? We humbly say a resounding "yes!", thanks to Arturia's TAE®, the proprietary technology developed by Arturia, and already successfully used in the Moog Modular V.

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