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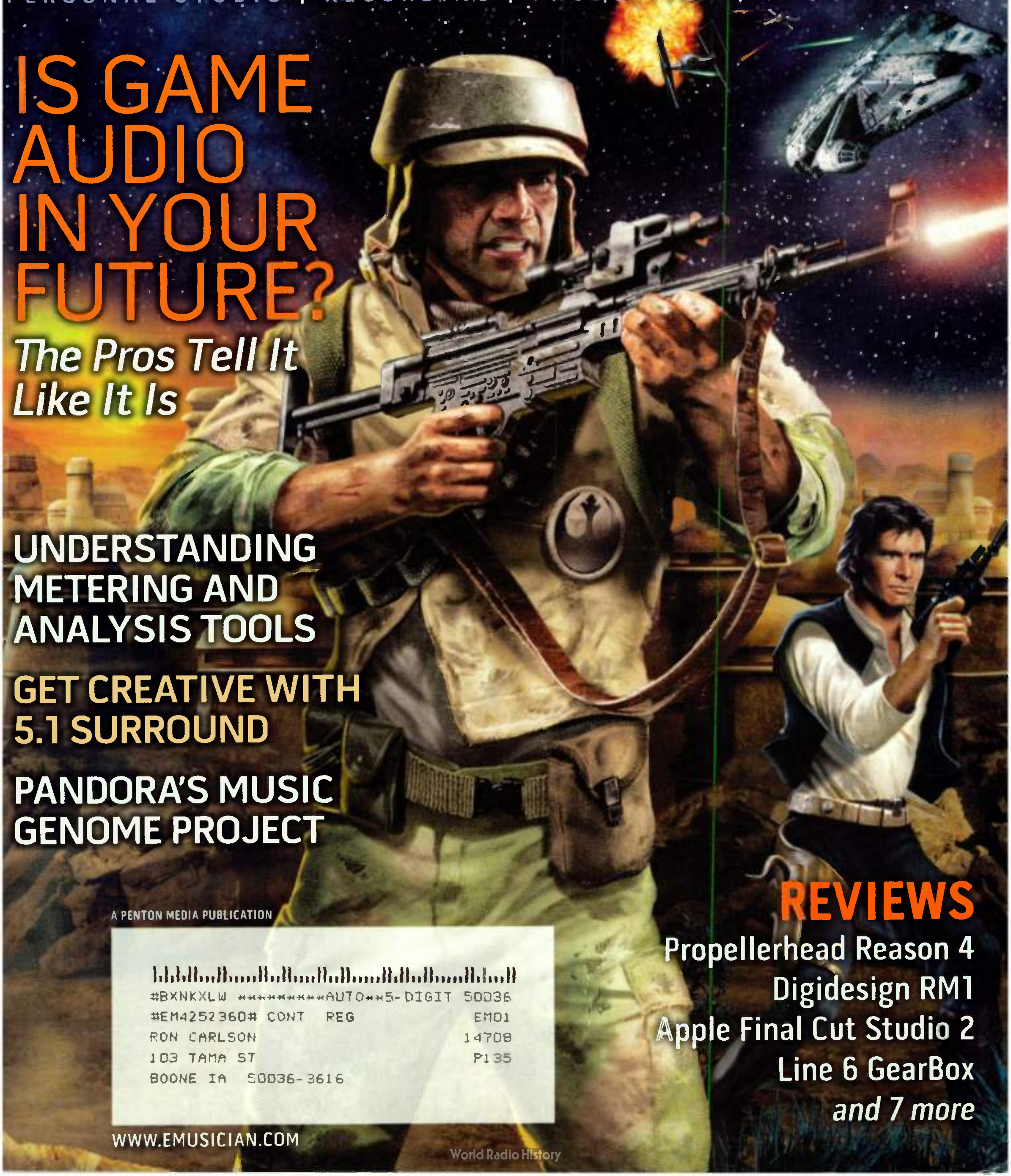

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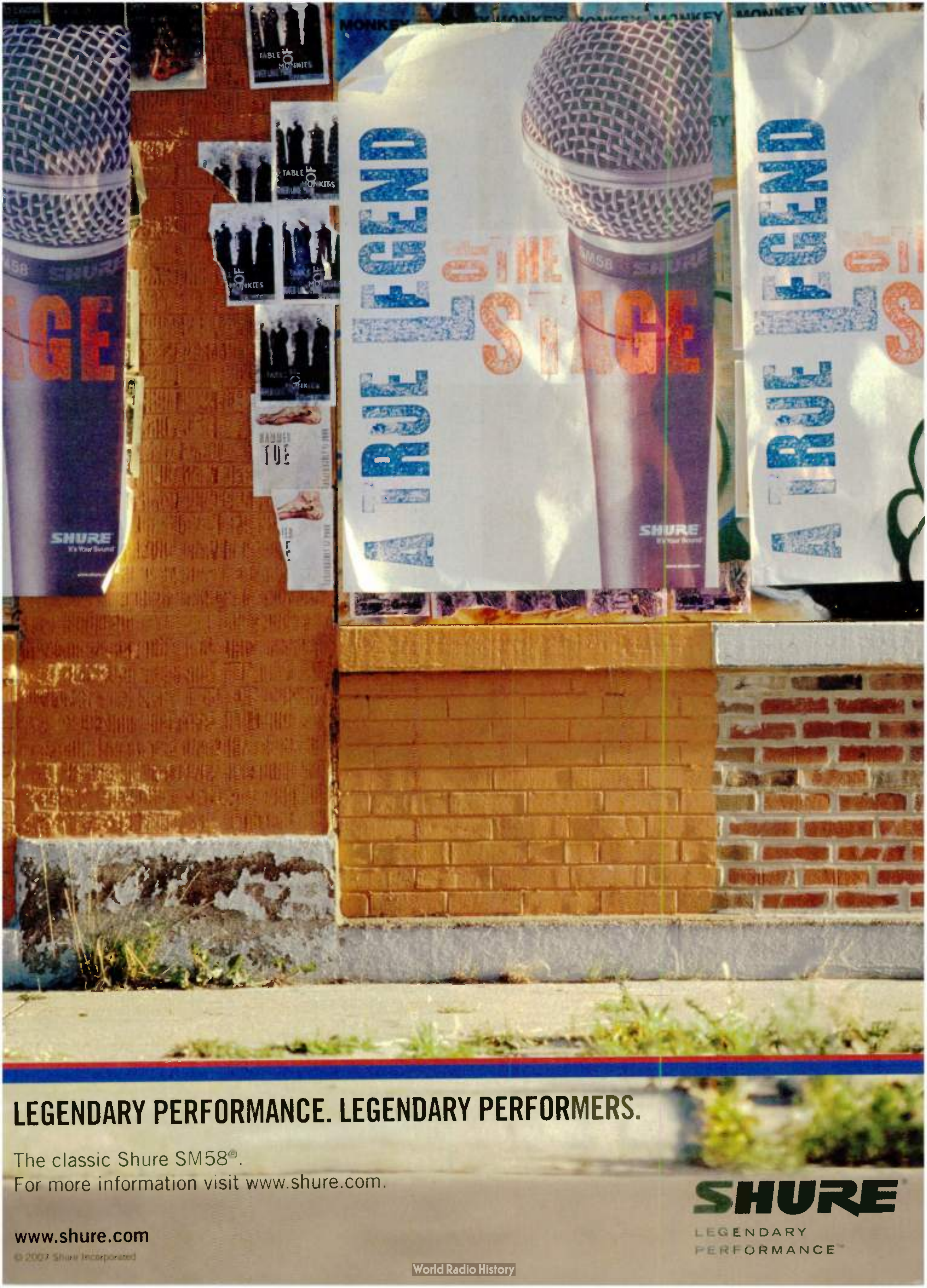

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TABLE OF CONTENTS

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December 2007

138

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CHIEF EXECUTIVE OFFICER
John French John.French@penton.com

CHIEF FINANCIAL OFFICER
Eric Lundberg Eric.Lundberg@penton.com

VICE PRESIDENT, GENERAL COUNSEL
Robert Feinberg Robert.Feinberg@penton.com

EDITORIAL, ADVERTISING, AND BUSINESS OFFICES:
6400 Hollis St., Suite 12, Emeryville, CA 94608, USA
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Electronic Musician®

December 2007

INSIDE

COURTESY LUCASARTS ENTERTAINMENT

FEATURES

COVER STORY

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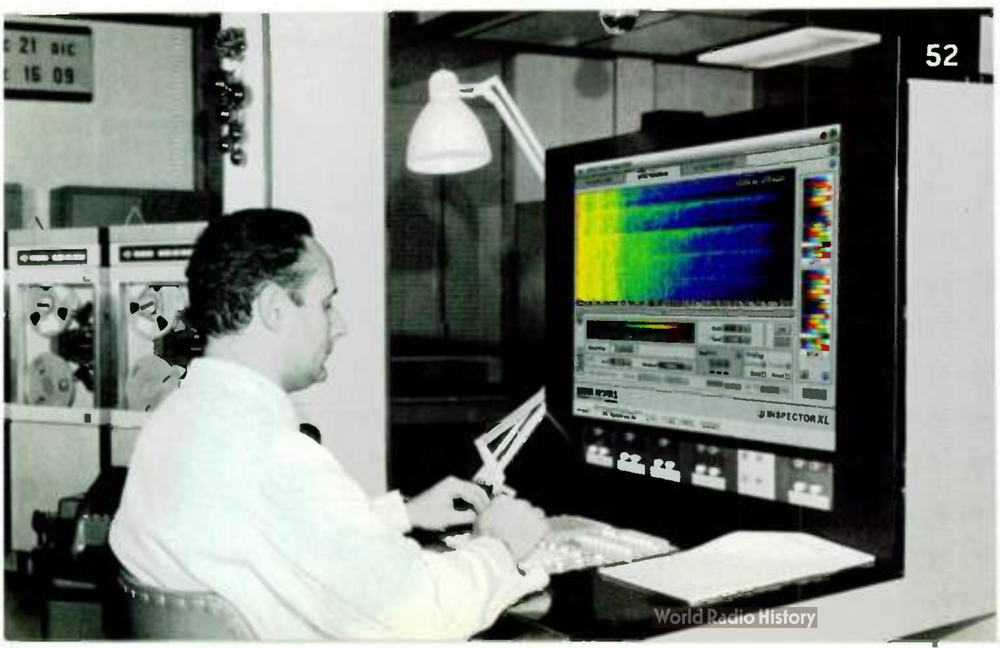

DEPARTMENTS

World Radio History

INSIDE

138

COLUMNS

116

REVIEWS

120

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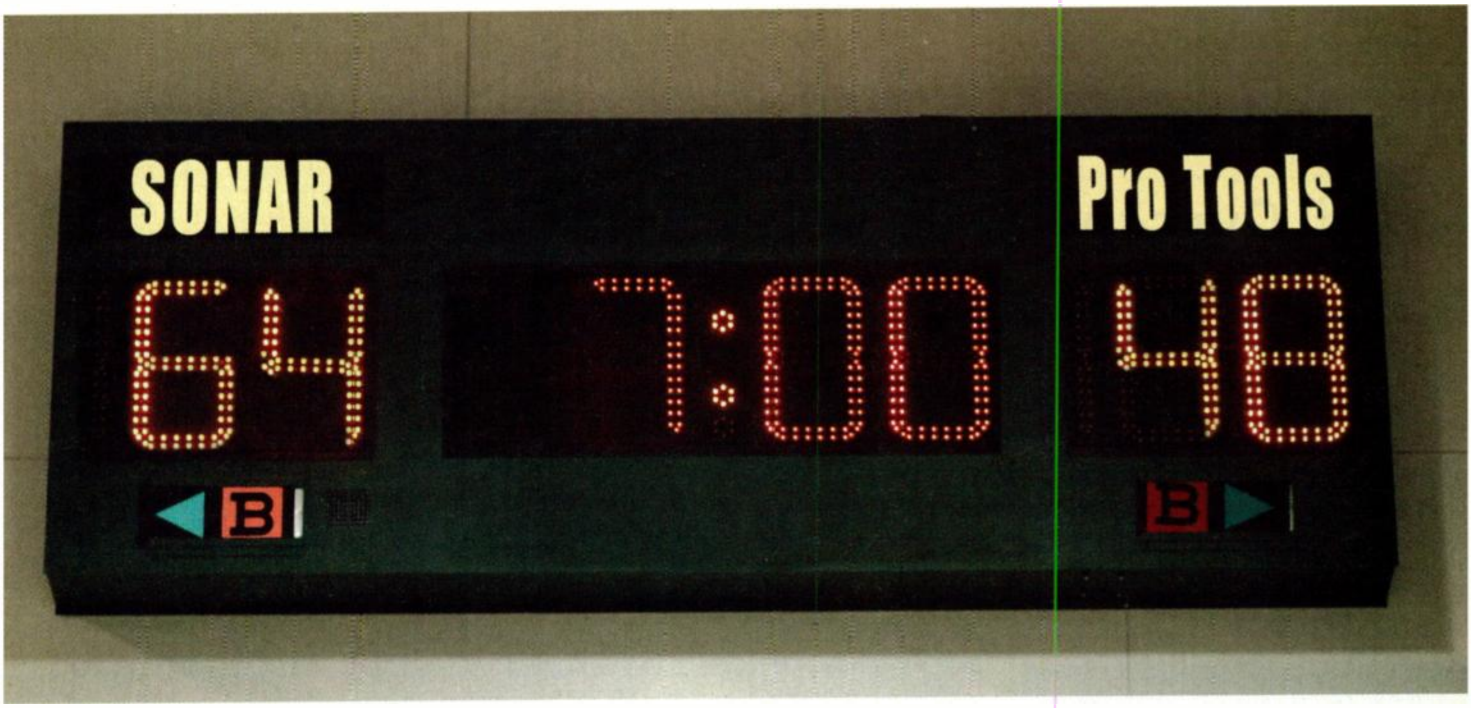

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All Over the MAP

No one likes deceptive advertising. I personally find it infuriating—as an editor and as a consumer—when a manufacturer plays games with pricing for a product that I'm interested in. For example, a microphone might have a manufacturer's suggested retail price (MSRP) of $999.95—a number that is already designed to make you think you're not spending a grand—but it might also have a minimum advertised price (MAP) of $349. Obviously, the company wants you to feel like you're getting the deal of a lifetime. But what's really going on? How can a consumer realistically compare products around a list price of, say, $1,000 when some of them actually sell for less than half that?

Manufacturers often set a minimum advertised price (MAP) to protect their dealers. Such vertical price restraints level the playing field so that retailers can offer the consumer a perceived discount on a product while maintaining enough of a profit margin. This is particularly important where a Web-based retailer, which probably has less overhead and can work on smaller margins, would be inclined to undercut the prices of a brick-and-mortar seller. Of course, there's no way to keep a retailer from selling an item below MAP if it wants to; it just can't advertise the lower price. (No doubt you've seen ads with the line "Call for the Lowest Price"—now you know one reason for it.)

JANE RICHEY

Until now, EM's policy has been to list only the MSRP for products, but this has become increasingly difficult to maintain, because MSRP has become decreasingly believable. More and more, we are asked during fact-check to run a MAP or street price rather than the MSRP. (Because street prices are difficult to pin down, we prefer not to use them, although sometimes street price and MAP are the same.) Manufacturers' requests to run a ridiculously low MAP have sometimes been followed by the admission that "no one in their right mind would pay MSRP for this product." Okay, so why set such a high list price? The replies I've received have never been satisfactory.

Consequently, few companies have an MSRP that is meaningful. In fact, it is common for us to be given a MAP or street price while being told that there isn't an MSRP. Obviously, that makes no sense.

This month EM will begin running minimum advertised prices (except when an MSRP is requested by a manufacturer) in order to better clarify the actual value of the products we cover. That way, when you read that a mic preamp has a MAP of $299, you're seeing what it's actually worth rather than, perhaps, a falsely inflated price. This change will have the biggest impact on our roundups where we set price caps, because it will allow products to be judged on more of an equal footing than ever before.

Sure, everyone likes a bargain. But no one wants to be jived into thinking that an inexpensively produced item is worth more than it is. So before you drop your hard-earned cash on something that lists for more than double the MAP, do your research and be sure you're getting what you pay for.

Gino Robair
Editor

Electronic Musician

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EDITOR
Gino Robair, grobair@emusician.com

EDITOR IN CHIEF/DIRECTOR OF TECHNOLOGY
Steve Oppenheimer, soppenheimer@emusician.com

SENIOR EDITOR
Mike Levine, mlevine@emusician.com

ASSOCIATE EDITORS
Geary Yelton, gyelton@emusician.com
Dennis Miller, emeditorial@emusician.com
Len Sasso, emeditorial@emusician.com

COPY CHIEF
Marla Miyashiro, mmiyashiro@emusician.com

GROUP MANAGING EDITOR
Sarah Benzuly, Sarah.Benzuly@penton.com

GROUP EDITORIAL ASSISTANT
Tracy Katz, tkatz@emusician.com

CONTRIBUTING EDITORS
Michael Cooper, Marty Cutler, Larry the O, George Petersen, Scott Wilkinson

EDITORIAL DIRECTOR
Tom Kenny, Tom.Kenny@penton.com

DIRECTOR OF AUDIENCE AND BUSINESS DEVELOPMENT
Dave Reik, Dave.Reik@penton.com

ONLINE AUDIENCE DEVELOPMENT MANAGER
Tami Needham, Tami.Needham@penton.com

GROUP ART DIRECTOR
Dmitry Panich, Dmitry.Panich@penton.com

ART DIRECTOR
Earl Otsuka, Earl.Otsuka@penton.com

INFORMATIONAL GRAPHICS
Chuck Dahmer, chuckd@chuckdahmer.com

EXECUTIVE VICE PRESIDENT
Darrell Denny, Darrell.Denny@penton.com

VICE PRESIDENT
Jonathan Chalon, Jonathan.Chalon@penton.com

EXECUTIVE ASSISTANT
Natalie Stephens, Natalie.Stephens@penton.com

GROUP PUBLISHER
Joanne Zola, Joanne.Zola@penton.com

ASSOCIATE PUBLISHER
Joe Perry, Joe.Perry@penton.com

EASTERN ADVERTISING DIRECTOR
Michele Kanatous, Michele.Kanatous@penton.com

EAST COAST ADVERTISING MANAGER
Jeff Donnenwerth, Jeff.Donnenwerth@penton.com

SOUTHWEST ADVERTISING MANAGER
Albert Margolis, Albert.Margolis@penton.com

LIST RENTAL
Marie Briganti, (845) 732-7054, marie.briganti@walterkarl.infousa.com

MARKETING DIRECTOR
Kirby Asplund, Kirby.Asplund@penton.com

MARKETING COORDINATOR
Clarina Raydmanov, Clarina.Raydmanov@penton.com

SALES EVENTS COORDINATOR
Jennifer Smith, Jennifer.Smith@penton.com

CLASSIFIEDS/MARKETPLACE ADVERTISING DIRECTOR
Robin Boyce-Trubitt, Robin.Boyce@penton.com

CLASSIFIEDS/SPECIALTY SALES MANAGER
Kevin Blackford, Kevin.Blackford@penton.com

CLASSIFIEDS PRODUCTION COORDINATOR
Jamie Coe, Jamie.Coe@penton.com

GROUP PRODUCTION MANAGER
Melissa Langstaff, Melissa.Langstaff@penton.com

ADVERTISING PRODUCTION COORDINATOR
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CHIEF FINANCIAL OFFICER
Eric Lundberg, Eric.Lundberg@penton.com

VICE PRESIDENT, GENERAL COUNSEL
Robert Feinberg, Robert.Feinberg@penton.com

EDITORIAL, ADVERTISING, AND BUSINESS OFFICES:
6400 Hollis St., Suite 12, Emeryville, CA 94608, USA
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ARTVILLE

Letters

Marching In with Rizzo

I just wanted to thank you for the Carmen Rizzo interview in the August 2007 issue of *Electronic Musician* (see "Production Values: Renaissance Man").

After reading the interview, I was encouraged to contact Carmen and see if he is interested in mixing a few of my tracks. I am very proud to announce that Carmen Rizzo is onboard to mix my debut CD *Marching In*—a tribute to the March King, John Philip Sousa.

Call it either destiny or fate, but I am most grateful for the article, as I now have one of the industry's most talented people helping me create great music. My concept/vision is called "Symphonic Guitars." What I like to do is transpose and transcribe orchestral scores to fit the guitar's tonal range, layering guitar tracks and creating a massive electric-guitar wall of sound. I am currently re-creating a handful of Sousa scores, utilizing only the guitar and lots of DSP.

Thank you again. I'm looking forward to sharing this music with you!

Dan Sindel
via email

We Welcome Your Feedback

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or email us at emeditorial@emusician.com.
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Dip Deeper in DP

Please let Michael Cooper know that rather than dicker with the Spectral Effects window in MOTU Digital Performer (see "Master Class: Pitch Fixers" in the October 2007 issue of EM), he might want to open DP's Sequence Editor instead. Just below the name of one of the audio tracks, he'll see a pop-up menu that is most often set to Soundbites. Click it, scroll down to Pitch, and look at what happens. Pitch correction that rivals—and is in many ways superior to—the plug-ins that Cooper mentions in his article has been in DP for well over a year. I guess you'd call that a major oversight.

James Loomis
via email

EM author Michael Cooper replies: James—You are correct. The editors and I decided not to include Digital Performer's pitch-automation capability in my article, because it was exclusive to DP and not everyone uses that particular DAW. In order to reign in the scope of the article to a manageable size and offer tips that virtually anyone could use, I deliberately focused on the three most popular cross-platform, third-party plug-ins. The discussion of DP's Spectral Effects processing was included to point out to those who don't own any of the featured plug-ins that most DAWs offer rudimentary pitch-transposition tools that can be pressed into service to correct pitch. Please watch for an upcoming "Making Tracks" article that will focus exclusively on how to use Digital Performer's advanced pitch-automation features.

Remembering Dodds

Will you be running any mention of the passing of Philip Dodds? He was the nervous, young ARP Instruments 2500 technician who played the famous 5-note alien-music sequence in the movie *Close Encounters of the Third Kind*.

But he was not an actor—his résumé included engineering stops at ARP and Kurzweil, serving on the committees that established the standards for CDs and DVDs, and creating a basis for the digital interchange of educational information. He died in early October 2007 in Maryland.

His passing may not have the seismic effect that the loss of Bob Moog or John Simonton did, but the guy was the most famous synthesizer player in the galaxy! That has to mean something.

Al Peterson
via email

Al—Thank you for writing. We received your note as we were going to press and were unable to create an obituary in time. However, we do intend to include one about Philip Dodds in an upcoming issue of EM. —Gino Robair

Error Log

October 2007, *Personal Studio Buyer's Guide* (PSBG). There were several errors and omissions in the 2008 edition of PSBG. For the most up-to-date corrections, please visit us online at emusician.com/psbg/psbg_errors. EM

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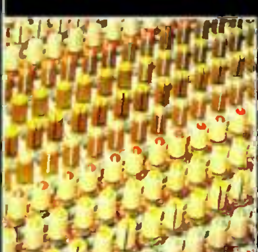

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Tommy Tallarico

Tommy Tallarico is a veteran composer who has written music for more than 250 games. In this interview, he talks about his experiences in the game industry.

Master Class:
Propellerhead Reason 4

Learn the ins and outs of Reason 4 from the ultimate Reason master, Kurt Kurasaki.

COLUMNS

Making Tracks:
Selecting in DP

There are many ways to make selections in MOTU Digital Performer. Our shortcuts will help speed up your work flow so you can make the most of your time.

Square One:
Fading Fast

Crossfades are an essential part of any audio-editing project. Learn what types there are and how to use them.

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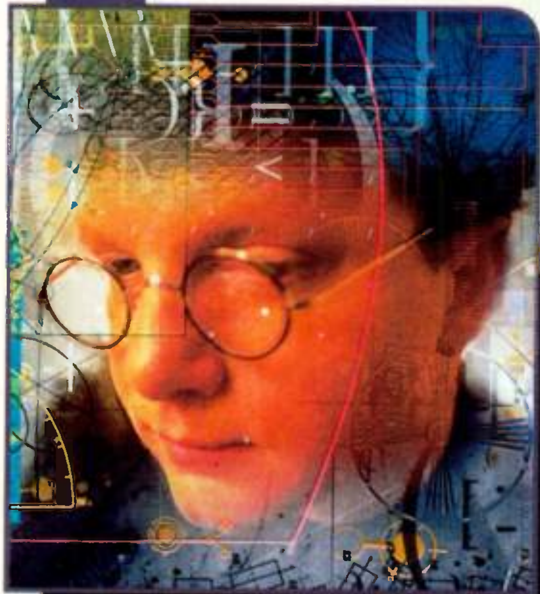

BEN PEARSON

EMspotlight

Creative Multitasking with Charlie Peacock

Keyboardist-songwriter Charlie Peacock's life is like a game of musical chairs: he is constantly changing seats and trading places. In this interview from the archive, Peacock describes how he moves easily between being a songwriter, a musician, and a producer. By Jennifer Conrad Seidel. emusician.com/em_spotlight

MOOGFEST 2007 ON EMTV

Moogfest is an annual festival that honors the influential work of synth pioneer Bob Moog. Visit emusician.com to watch exclusive video from this year's show, featuring Don Preston, Jordan Ruddess, Thomas Dolby, and T. Lavitz. emusician.com/videos/events/moogfest/

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World Radio History

By Len Sasso

Roland SP-555

Roland (www.rolandus.com) announces the newest model in its SP line of samplers. The loop-musician-oriented SP-555 ($595) offers real-time audio-loop capture and will function as a USB audio interface for your computer. Its 30 MB internal memory can record and stream up to 22 minutes of lo-fi, mono audio (11 minutes for standard-mode mono), or you can insert a CompactFlash (CF) card with up to 2 GB of memory for more than 26 hours of real-time lo-fi streaming. The unit comes bundled with Cakewalk Sonar LE (Win) for digital audio sequencing and Wave Converter (Win) software for importing AIFF and WAV files to a CF card (you'll need a CF reader for your computer).

The 4-pound desktop unit has 16 Velocity-sensitive pads for triggering samples, 3 knobs, and Roland's D-Beam 3D controller. Built-in effects range from standard fare to specialties like SuperFilter and DJFX Looper. In addition to audio recording, you get an 8,000-note pattern sequencer organized in banks of 16 patterns (2 banks in internal memory and 8 additional banks in CF memory). You can even sync the SP-555 to V-Link-compatible video equipment. A 9 VAC adapter powers the SP-555, which has a full complement of I/O connectors, including a phantom-powered XLR mic input.

Roger Linn Designs AdrenaLinn III

Roger Linn Designs (www.rogerlinndesigns.com) has announced a major upgrade to its AdrenaLinn guitar processor. The AdrenaLinn III ($375) expands the selection of amp models to 40 and increases memory to 200 user presets and 200 user drumbeats. The unit comes filled with factory presets and drumbeats, of which you'll find a complete listing at the company's Web site. AdrenaLinn II users can buy an upgrade kit for $99. Cross-platform editor software is available for $39.95 from SoundTower (www.soundtower.com).

The AdrenaLinn III's enhancements include adjustable stereo width for modulation and delay effects, adjustable envelope attack and decay times for random filter and tremolo effects, simple guitar-amp distortion for drumbeats, and better drum sounds. You can now trigger the internal drum sounds with MIDI. You get four new modulation effects—Auto Pan, Wah Pedal, Fixed Filter, and Sci Fi—along with reverb, compression, and a built-in tuner. The upgrade lets you assign the right footswitch on a per-preset basis to enable any effect or combination, and you can reassign both footswitches. MIDI pedalboard implementation supports ten MIDI footswitches along with two MIDI expression pedals, and you can use those to control virtually any internal setting. Of course, you still get all the signature beat-synced modulation effects and filter, tremolo, and arpeggio sequences.

World Radio History

JazzMutant Dexter

JazzMutant (www.jazzmutant.com) is shipping the second in its line of graphics-tablet control surfaces. Dexter (Mac/Win, $3,299 [MSRP]) is designed with your digital audio sequencer in mind, if that happens to be Apple Logic Pro, Steinberg Cubase or Nuendo, or Cakewalk Sonar. It gives you instant, graphical access to all aspects of mixing and insert-effects parameters. Its 800 × 600–pixel color display uses proprietary Multitouch Display Technology, so you can simultaneously manipulate multiple faders with your fingers. You can even zoom the fader ranges for more-precise editing. All faders have built-in level meters that are continuously updated from your DAW.

Dexter groups tracks in banks of eight and automatically retrieves track names from your DAW. Paging through banks and creating your own bank groups is quick and easy, and you can limit the display to muted, soloed, or armed tracks, and then toggle their status with a single tap. Four track-edit views—Mixer, Equalizer, Insert, and Surround—let you home in on specific mixing parameters, and a fifth, called Channel Edit view, puts smaller versions of all four views on a single screen. In most edit views, you get a huge fader with zoomable resolution to control level, pan, or send amounts.

Apogee Electronics Duet

Apogee Electronics (www.apogeedigital.com) has just released the smallest in its Ensemble line of audio interfaces. The Duet (Mac, $495 [MSRP]) is a 2-in/2-out, 24-bit, 96 kHz, bus-powered FireWire 400 device requiring Mac OS X 10.4.10 and Core Audio. A breakout cable attaches to the back of the Duet and features two unbalanced ¼-inch TS high-impedance instrument inputs and two 48V phantom-powered balanced XLR inputs. You can configure the XLRs as mic inputs with 75 dB of gain or as line inputs accepting a maximum input level of +20 dBu. The interface also has two ¼-inch TS monitor outputs for powered speakers. A ¼-inch stereo headphone output on the front panel makes it ideal for portable music making.

Sound Advice

Just because Logic Studio now includes everything from every previous GarageBand Jam Pack doesn't mean that **Apple** (www.apple.com) has stopped developing new sound libraries for purchase. The latest Jam Pack, *Voices* ($99), supplies nearly 3 GB comprising more than 1,500 Apple Loops and 28 software instruments constructed from sampled vocals. The variety of sounds is impressive, from divas, rappers, and classical ensembles to vocoder and human beatbox. You'll find all manner of background harmonies, lyrical passages, and solo embellishments in styles ranging from pop, jazz, and gospel to square-dance calls and Bollywood. In addition to several complete choirs, *Voice*'s selection of software instruments furnishes spoken phrases, shouts, whistles, and body sounds.

The hang is a unique handmade acoustic instrument whose popularity has been catapulted by YouTube. It's a sort of hollow steel drum (aka steel pan) played with the fingers. The hang is made only by a tiny Swiss company called PanArt and resulted from years of research on resonating percussion from around the world. **Soniccouture** (www.soniccouture.com), always on the prowl for esoteric sounds, has sampled two models of the hang and created *Hang Drum* ($100 download, $106 disc), a 2.4 GB library for Kontakt 2 and 3. With as many as 21 Velocity layers, each instrument takes advantage of Kontakt's scripting and has a built-in generative pattern sequencer. *Hang Drum* also allows you to use MIDI Control Changes for real-time parameter control.

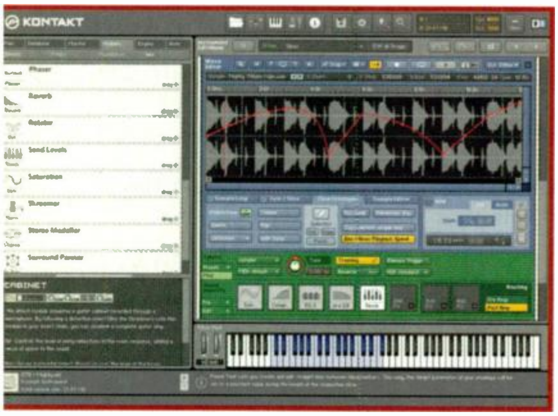
—*Geary Yelton*

World Radio History

Native Instruments Kontakt 3

Native Instruments (www.native-instruments.com) has released a major upgrade to its flagship sampler. Kontakt 3 (Mac/Win, $449 [MSRP], $149 upgrade) offers a passel of new features along with a huge (33 GB) library upgrade. The library is spread across six collections: Band, Orchestral, Synth, Urban Beats, Vintage, and World. Urban Beats contains 50 drum-loop production kits with single-instrument loops for maximum flexibility. All in all, you get 1,000 instruments, each with its own customized control panel (called Performance View) for easy access to relevant parameters.

Kontakt 3 has an integrated sample editor, obviating the need to switch applications for destructive sample editing. The new Zone Envelopes feature enables you to draw modulation envelopes over the sample waveform of any zone. Among the loop-slicing improvements, you can drag-and-drop associated MIDI trigger files directly to a host sequencer. Automapping, which uses file names to determine a zone's Velocity and key range, lets you create your own mapping rules for decoding enigmatic sample names.



You get several new effects, including an amp/cabinet simulator. In addition, the Browser has been improved with automatic updating and an instrument navigator to select the instrument displayed in the rack for editing.

Neyrinck Mix 51

Neyrinck (www.neyrinck.com) has good news for Pro Tools LE and M-Powered users who want to start mixing in surround 5.1. Mix 51 (Mac/Win, $189) brings full-featured surround mixing to anyone who has an audio interface with at least six outputs, such as the Digidesign Mbox 2 Pro, 002, or 003 or the M-Audio Delta 1010 or FireWire 410. The package consists of three RTAS plug-ins—Surround Mixer, Surround Panner, and LFE Send—and supports sampling rates from 44.1 to 192 kHz. The plug-ins give you three independent 5.1 buses and three independent quad effects-send buses. Each bus has automatable volume, panning, mute, and solo with the same volume and panning tapers as Pro Tools HD, so you can transparently migrate projects directly to high-end Pro Tools rigs.

To use the Mix 51 system, you insert the Surround Panner plug-in on any track to route its output to any of the 5.1 buses as well as to any of the quad send buses, pre- or postfader. You then insert the Surround Mixer plug-in on at least one track, and it provides all 30 outputs of the mix and send buses as auxiliary output stems. Those outputs show up as inputs on other aux tracks for routing back into the Pro Tools mixer. Mix 51 requires an iLok Smart Key.

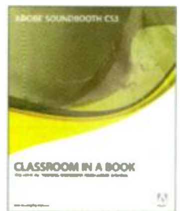
Get Smart

The Classroom in a Book series from **Adobe Press** (www.adobepress.com) is an excellent resource for users of Adobe software. Of particular interest to EM readers is *Adobe Soundbooth CS3* ($54.99), a 211-page text that serves as "the official training workbook from Adobe Systems" and can lead to Adobe certification. It takes you step-by-step through nine lessons that cover the basics and present dozens of tips and productivity techniques. You learn how to find your way around in Soundbooth CS3's Workspace, take advantage of AutoComposer to create custom music, integrate your edits with Adobe After Effects and Premiere Pro, and more. An included CD-ROM contains media files to use with all the lessons.

A new book from **Peachpit Press** (www.peachpit.com), Martin Sitter's *Soundtrack Pro 2* ($54.99), charts a similar course toward Apple Pro certification. Part of the Apple Pro Training Series, the 358-page book is subtitled *Sound for Picture in Final Cut Studio and Logic Studio*. Ten comprehensive lessons cover specific applications such as recording and mixing multitrack audio, integrating work flow with Final Cut Pro, enhancing and restoring audio files, manipulating the frequency spectrum, and so on. A DVD-ROM containing lesson files accompanies the book.

A new book from **Magic Smoke Electronics** (www.magsmoke.com) called *An Analog Synthesizer for the 21st Century* ($20) could help you realize the dream of building your own analog synth. Using easy-to-find parts, legendary DIY electronics author Thomas Henry provides schematics and parts lists for constructing a complete voltage-controlled instrument with two audio oscillators, a multimode filter, a noise generator, ADSR generators, a versatile LFO, sample-and-hold, a phase shifter, and more. Neophytes beware: this 36-page book offers no instruction and is intended for readers who are confident of their DIY chops.

—Geary Yelton



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Apple Logic Studio

The big news from Apple (www.apple.com) is that Logic Pro has gotten a major and very user-friendly overhaul. Logic Studio (Mac, $499 (MSRP)) is a software suite that contains the upgraded Logic Pro 8 and is half the price of Logic Pro 7 alone. It comes with a bundle of other pro-level applications and 40 GB of content.

Among the bundled applications, MainStage turns your MacBook or MacBook Pro into a stageworthy live-performance rig. Soundtrack Pro 2 combines multitrack audio editing with a slew of postproduction tools including video integration. Waveburner is a complete CD-mastering solution, and Compressor 3 is a full-featured surround encoding utility. In addition to Logic's Studio Instruments and Studio Effects collections, you get the Studio Sound Library with over 18,000 Apple Loops and more than 6,000 instrument and effects presets, including 1,300 multisampled EXS24 instruments.

The most dramatic part of Logic Pro 8's redesign is the new Arrange window, which has collapsible panes to consolidate 15 edit and browser areas into a unified work space (multiwindow screen sets are still available when you want them). Multiple recorded takes are now managed in expandable take folders, and Quick Swipe Comping provides a new way to create comps by just swiping over the best take segments. And finally, the XSkey dongle is gone, freeing up a USB port.

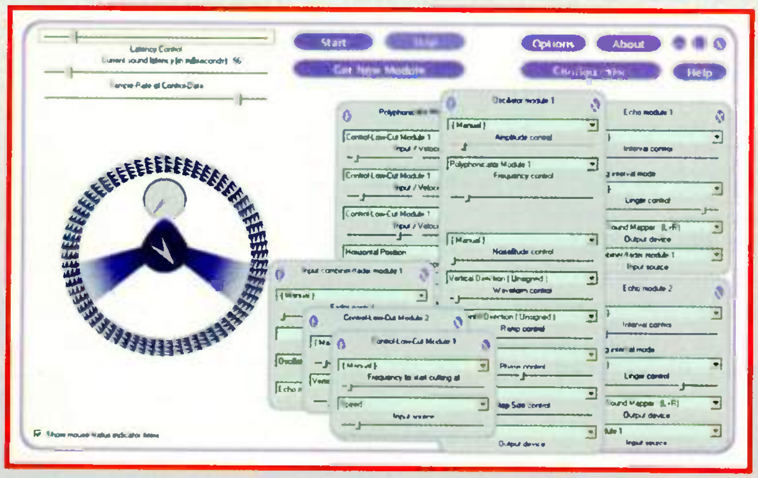
Download of the Month

ROLLOSONIC (WIN)

If you have a PC and you like to make noise, RolloSonic ($29.95) is for you. This sparsely documented and enigmatic modular synth's main claim to fame is that you control its sound by moving the mouse. There are lots of other ways to get sound out of it—MIDI control, internal note sequencer, and so on—but using the mouse is the most unusual and the most creative. It's also the most unpredictable; if you need to be in control of your sonic environment, look elsewhere (see **Web Clip 1**).

RolloSonic is a cordless modular synth; all inputs and outputs are selected from drop-down menus. The sound modules are rudimentary. You get an oscillator with several ways to control its waveform and noise mix as well as the Ding-Dong module, which generates notes with an FM-like sound. You can also process external audio. Audio processors include filters, distortion, delay, and a pitch-shifter. You can have multiple modules of each kind, and aside from a few mysteriously named parameters, hooking them up is straightforward. The secret is in controlling them.

As mentioned, you can use MIDI or the sequencer, but the intended (and more interesting) approach is to use some aspect of mouse motion. Options are variants of position, speed, direction, and inertia. You assign these options to sound-module control inputs to affect the sound. Things get even more interesting when you use Control Source and Effect modules. The purpose of some of these, like Control Offset and Control Echo, is obvious, but with others, like Control Neuron and Polyphonicator, it's anyone's guess. You can also use any audio signal as a control source, and because you can push the control-data rate into the audio range, you can use control sources in the audio signal path. Experimentation is the name of the game, and the absence of documentation makes RolloSonic a challenge, but 22 factory presets help point the way. Buy it or check out the fully functional demo at www.rollosonic.com. EM



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World Radio History

Listen and Learn

By Scott Wilkinson

It's more than just a numbers game.

Audio technology has come a long way since the days of vacuum tubes, though some still believe that tubes produce the best sound. In most applications, however, solid-state electronics have brought high-quality sound to consumers and audio professionals alike.

For the last 30 years, National Semiconductor (www.national.com) has been at the forefront of analog audio electronics, designing and building integrated circuits (ICs, or "chips") that are used in everything from A/V receivers to mixers to cell phones. In the mid-1990s, the company created a division dedicated to producing audio chips of the highest possible quality, and the most recent fruits of this labor blow just about everything else away.

For example, National's new LME line of op-amps achieve a total harmonic distortion plus noise (THD + N) of 0.00003 percent, a record low for this particular spec. Most high-quality op-amps have a THD + N of around 0.002 percent, roughly two orders of magnitude higher than the LME line. A THD + N in this range is fine in and of itself, but after a signal passes through ten such op-amps in a mixer, the final level of distortion and noise is much higher. By contrast, the final THD + N after passing through ten LME op-amps is still in the so-called triple-zero range (0.000x percent), much lower than that of even one typical chip.

Specs such as these are important, but the National team believes that they are only part of the story. Equally important is how the components sound in actual audio equipment. As a result, the team routinely builds preamps, power amps, and other audio devices with the new chips to see just how good they sound.

To help in this effort, National constructed a dedicated sound room, which was originally based on a recording-studio environment with Tannoy close-field monitors mounted in the wall where a mixer would be placed. But the team ultimately decided that a mid- to far-field setup would be more useful, so the space was configured more like an audiophile listening room (see Fig. 1).

The room is fully isolated with a floating concrete floor, airtight door, and acoustic treatments all around. Nothing can be heard outside, even when the volume is cranked, and a special air-conditioning system keeps the ambient noise level within very low. The engineers, many of whom are gifted with golden ears, listen on pairs of Wilson Audio WATT/Puppy 7s and B&W 802s and 801 Matrix 3s, all exceptional consumer speakers that reveal every nuance of any audio device's performance.

National's high-performance chips are made possible by innovations in two areas: design and process. In the design phase, circuit elements are laid out to enhance linearity and reduce distortion.

"Process" refers to the way in which circuits are actually constructed on a wafer. The LME line uses a process called VIP3 (Vertically Integrated PNP), a type of bipolar process that uses both PNP and NPN transistors. (*PNP* and *NPN* refer to the subatomic structure of the transistors and how electrons flow through them.) Unlike a standard bipolar process, in which the NPN transistors have high speed and bandwidth while the PNP transistors do not, VIP3 brings the speed and bandwidth of the PNPs up to the same level as the NPNs, avoiding the compromises that must otherwise be made.

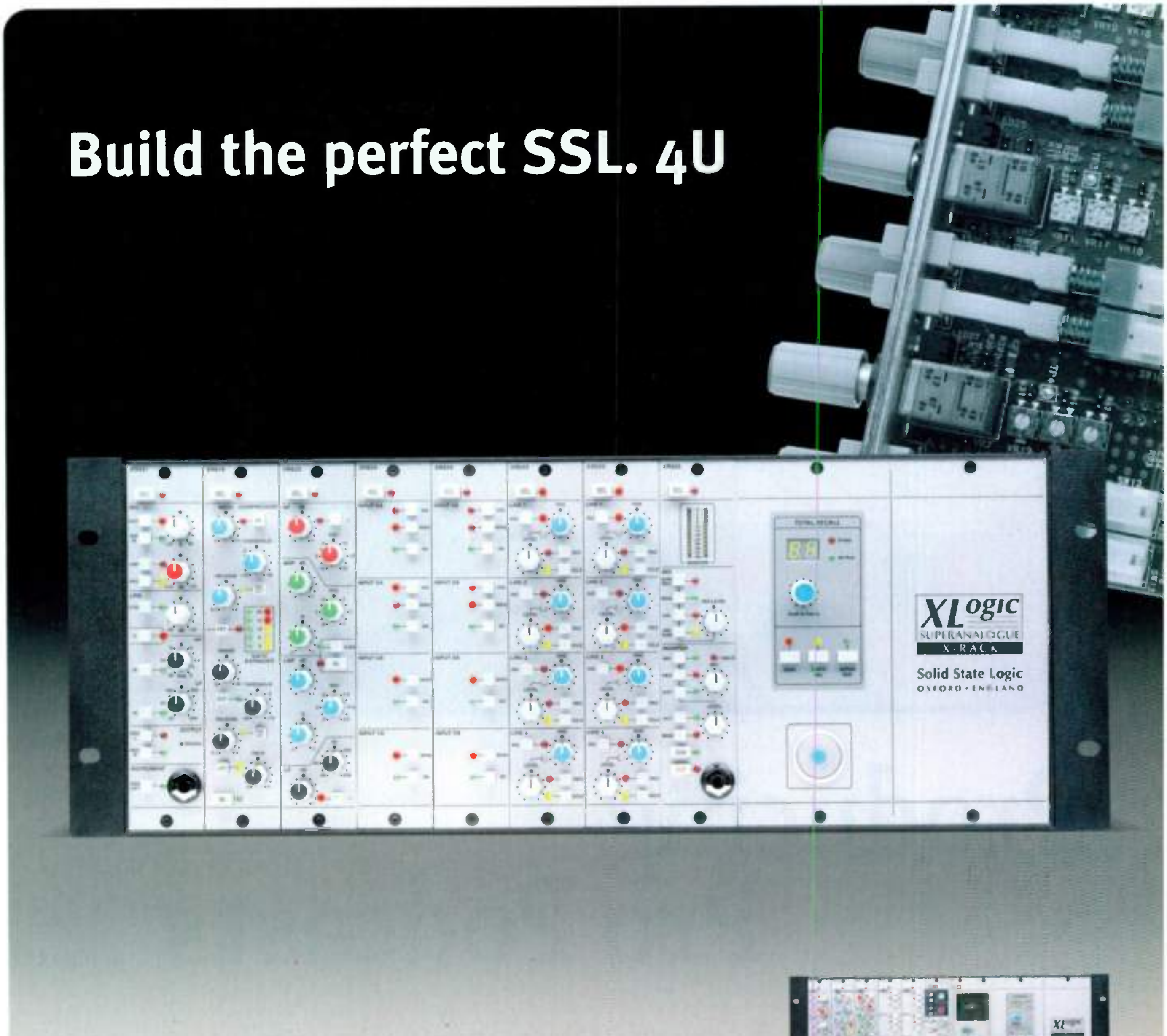
The LME series is just starting to become available to manufacturers of audio products, and the company expects to develop many more super-high-quality audio chips in the near future. Thanks to a combination of design, process, objective measurements, and subjective listening, we can expect wonderful sounds to migrate from National Semiconductor's sound room to our studios. EM

COURTESY NATIONAL SEMICONDUCTOR

FIG. 1: National Semiconductor's sound room is completely isolated and acoustically treated. The finest consumer speakers are used to evaluate the performance of devices built with the company's audio chips.

World Radio History

www.solid-state-logic.com



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DMITRY PANICH

Producing Pro Podcasts

By Mike Levine

Tips and techniques for creating great-sounding online productions.

Anyone with a USB mic and a computer can record a Podcast, but if you want to produce a product that will hold its own with the radio shows and other top-shelf downloadable content available on the Web, you'll need both studio chops and an attention to detail. Sonic problems such as uneven levels, background noise, and distorted audio will all have a negative impact on your listeners no matter how good the content is.

As the producer of *Electronic Musician*'s twice-monthly "EM Cast," I've picked up a lot of useful Podcast production techniques. In this article, I'll try to pass on as much advice as I have space for. For additional information, see David Battino's "The Art of Podcasting" in the December 2005 issue of EM (available online at www.emusician.com). That story covers many general Podcasting issues, such as putting together RSS subscription feeds and promoting your product, that won't be covered here. I will focus on the production side: recording, editing, and mixing.

Your most important tool for Podcast production will be a digital audio sequencer or multitrack audio editor. I recommend fully featured applications like Digidesign Pro Tools, Apple Logic Studio (or Logic Express 8), Steinberg Cubase, MOTU Digital Performer, Cakewalk Sonar, and Ableton Live. You could also do quite well with Apple GarageBand, which has a lot of Podcasting features built in.

World Radio History

Talk to Me

All Podcasts contain at least some voice-overs or other spoken-word elements, and many consist solely of such content. Interviews are one of the best sources of material if you're putting together an informational or topical Podcast.

If you're recording an in-person interview or discussion for your Podcast, I would recommend using a digital 2-track recorder with a stereo mic. There are a number of good, relatively inexpensive models on the market by manufacturers such as M-Audio, Edirol, and Zoom. In a slightly higher price category, you might consider Sony's new PCM-D50, which is a lower-priced alternative to the company's high-end PCM-D1, but which offers similar functionality.

If you have a laptop with recording software and a good USB mic, you could record with that as well. Make sure you can get enough level with the mic you're using; some USB mics do not provide much gain. Another option is to record using a laptop with a portable audio interface and a conventional dynamic or condenser mic.

For both interview and voice-over recording, I recommend using 24-bit resolution whenever possible (at a sampling rate of 44.1 or 48 kHz). Although your final product will end up as an MP3, you want to get the best-quality source recording possible. The increased dynamic range and improved signal-to-noise ratio of 24-bit audio (as compared with 16-bit) gives you the luxury of not having to worry as much about levels that are a little too low. But even at 24-bit, you should always strive to capture levels that are as high as possible without clipping. During interviews, be constantly ready to adjust your recording level down, when necessary, to avoid digital overs. Digital distortion cannot be fixed in the mix, so if you have to err with your level setting, do so to the lower side.

Try to set up your interview in as quiet an environment as possible. Steady or intermittent background noise can wreck your final product. (Beware of air conditioners!) Although there are some methods for removing noise after the fact (more on this later), by far the best route is to get a clean initial recording.

Also pay attention to the acoustics of the space you're recording the interview in. If it's really reverberant, you can minimize that by holding the mic close to the interviewee, and then bringing it close to you for your questions. Try to move it in between questions and answers so that as little handling noise as possible occurs when you or your interviewee is talking.

If you can, do a test recording before you actually start the interview to make sure everything is sounding good. Bring headphones so that you can accurately judge your test recording. Make sure your settings are correct and that the machine is actually recording when your interview starts. There's nothing worse than finishing an interview and discovering that the recorder was in pause mode, not record, for the entire conversation.

Call Me

Telephone interviews are an important part of many Podcasts. They provide the opportunity to bring in the opinions and thoughts of people from anywhere in the world. But recording phone interviews with good fidelity is not easy. While recording phone-based Podcast interviews for EM, I've experimented with a number of different methods. Here are three that I've used, presented in order of preference.

Miking the speakerphone. I've gotten good results close-miking a speakerphone for the caller's voice while separately miking my own voice, and recording the output of both mics onto separate tracks of my recording software. This allows me to use high-quality mics and get pretty good separation. The recording of the close-miked

TAG, YOU'RE IT

FIG. A: The Info page in iTunes, where you can edit your ID3 tags.

Once you have the converted MP3 file of your Podcast, you have one more step left, which is to add ID3 tags. These embed important metadata into your file, such as the name of the Podcast, the artist, and a description of the contents. If you're setting up your Podcast so that listeners can subscribe to it in a program like Apple iTunes, the ID3 tag is a critical item.

I usually drag the MP3 file of my Podcast into iTunes to edit the ID3 tag. I've tried editing in other audio applications, but sometimes the results aren't satisfactory (entries can get truncated). Standalone ID3 tag editors are also available.

Select your Podcast in iTunes' main window and hit Command-I, and you'll get the Info page (see Fig. A). Choose a standard naming convention that you'll use across all your Podcasts, and which is descriptive and will serve as your file name. The reason for standardization is that if you have a number of your Podcasts together in, say, an RSS subscription feed, or even in a list on your Web site, standardized file names will make your list look a lot cleaner and more professional.

Also, make sure that you put a concise description in the Comments field. When your Podcast is subscribed to, the comments are important because they help potential listeners choose which episode they want to download.

World Radio History

CHUCK DAHMER

FIG. 1: This drawing shows the basic setup for the miking-the-speakerphone method of phone-interview recording.

speakerphone sounds surprisingly good, as long as I get good levels without turning the speaker up to the point of distortion. Overall, it's the best method I've found.

To do this, position yourself facing the speakerphone so that your mic is about 1.5 feet from it (see **Fig. 1**). If you get too close, you'll have too much bleed between your mic and the speakerphone mic, but if you get too far, the person on the other end of the line won't be able to hear you talk. (The only way he or she will hear you is through the built-in mic in the speakerphone.)

Point your vocal mic away from the speakerphone to minimize leakage. Use cardioid or supercardioid mics to reduce both bleed and room noise. As when you record voice-overs, a pop screen on your mic is a must for reducing plosives ("popped" *p*, *b*, and other consonant sounds). But even with such a device, some exaggerated plosives will inevitably end up on your recording, and you'll have to lessen or remove them when you're editing (more about this later).

Because there's a degree of leakage in this method, it's best to edit out or mute the audio on your track when you're not talking, and do the same for the other party. This will clean up the sound immensely, and the only places where you may run into trouble due to bleed are those moments when both parties talk at once, the result of which is that you can't mute one of the tracks. I've never found this to be a deal breaker, but it's one of the downsides, along with the aforementioned editing, of this method.

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You want to get the best-quality source recording possible.

More Phone Solutions

Beyond those three methods, there are other phone-recording options. For example, Parliant Phone Valet

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Talk to Me

All Podcasts contain at least some voice-overs or other spoken-word elements, and many consist solely of such content. Interviews are one of the best sources of material if you're putting together an informational or topical Podcast.

If you're recording an in-person interview or discussion for your Podcast, I would recommend using a digital 2-track recorder with a stereo mic. There are a number of good, relatively inexpensive models on the market by manufacturers such as M-Audio, Edirol, and Zoom. In a slightly higher price category, you might consider Sony's new PCM-D50, which is a lower-priced alternative to the company's high-end PCM-D1, but which offers similar functionality.

If you have a laptop with recording software and a good USB mic, you could record with that as well. Make sure you can get enough level with the mic you're using; some USB mics do not provide much gain. Another option is to record using a laptop with a portable audio interface and a conventional dynamic or condenser mic.

For both interview and voice-over recording, I recommend using 24-bit resolution whenever possible (at a sampling rate of 44.1 or 48 kHz). Although your final product will end up as an MP3, you want to get the best-quality source recording possible. The increased dynamic range and improved signal-to-noise ratio of 24-bit audio (as compared with 16-bit) gives you the luxury of not having to worry as much about levels that are a little too low. But even at 24-bit, you should always strive to capture levels that are as high as possible without clipping. During interviews, be constantly ready to adjust your recording level down, when necessary, to avoid digital overs. Digital distortion cannot be fixed in the mix, so if you have to err with your level setting, do so to the lower side.

Try to set up your interview in as quiet an environment as possible. Steady or intermittent background noise can wreck your final product. (Beware of air conditioners!) Although there are some methods for removing noise after the fact (more on this later), by far the best route is to get a clean initial recording.

Also pay attention to the acoustics of the space you're recording the interview in. If it's really reverberant, you can minimize that by holding the mic close to the interviewee, and then bringing it close to you for your questions. Try to move it in between questions and answers so that as little handling noise as possible occurs when you or your interviewee is talking.

If you can, do a test recording before you actually start the interview to make sure everything is sounding good. Bring headphones so that you can accurately judge your test recording. Make sure your settings are correct and that the machine is actually recording when your interview starts. There's nothing worse than finishing an interview and discovering that the recorder was in pause mode, not record, for the entire conversation.

Call Me

Telephone interviews are an important part of many Podcasts. They provide the opportunity to bring in the opinions and thoughts of people from anywhere in the world. But recording phone interviews with good fidelity is not easy. While recording phone-based Podcast interviews for EM, I've experimented with a number of different methods. Here are three that I've used, presented in order of preference.

Miking the speakerphone. I've gotten good results close-miking a speakerphone for the caller's voice while separately miking my own voice, and recording the output of both mics onto separate tracks of my recording software. This allows me to use high-quality mics and get pretty good separation. The recording of the close-miked

TAG, YOU'RE IT

FIG. A: The Info page in iTunes, where you can edit your ID3 tags.

Once you have the converted MP3 file of your Podcast, you have one more step left, which is to add ID3 tags. These embed important metadata into your file, such as the name of the Podcast, the artist, and a description of the contents. If you're setting up your Podcast so that listeners can subscribe to it in a program like Apple iTunes, the ID3 tag is a critical item.

I usually drag the MP3 file of my Podcast into iTunes to edit the ID3 tag. I've tried editing in other audio applications, but sometimes the results aren't satisfactory (entries can get truncated). Standalone ID3 tag editors are also available.

Select your Podcast in iTunes' main window and hit Command-I, and you'll get the Info page (see Fig. A). Choose a standard naming convention that you'll use across all your Podcasts, and which is descriptive and will serve as your file name. The reason for standardization is that if you have a number of your Podcasts together in, say, an RSS subscription feed, or even in a list on your Web site, standardized file names will make your list look a lot cleaner and more professional.

Also, make sure that you put a concise description in the Comments field. When your Podcast is subscribed to, the comments are important because they help potential listeners choose which episode they want to download.

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CHUCK DAHMER

FIG. 1: This drawing shows the basic setup for the miking-the-speakerphone method of phone-interview recording.

speakerphone sounds surprisingly good, as long as I get good levels without turning the speaker up to the point of distortion. Overall, it's the best method I've found.

To do this, position yourself facing the speakerphone so that your mic is about 1.5 feet from it (see **Fig. 1**). If you get too close, you'll have too much bleed between your mic and the speakerphone mic, but if you get too far, the person on the other end of the line won't be able to hear you talk. (The only way he or she will hear you is through the built-in mic in the speakerphone.)

Point your vocal mic away from the speakerphone to minimize leakage. Use cardioid or supercardioid mics to reduce both bleed and room noise. As when you record voice-overs, a pop screen on your mic is a must for reducing plosives ("popped" *p*, *b*, and other consonant sounds). But even with such a device, some exaggerated plosives will inevitably end up on your recording, and you'll have to lessen or remove them when you're editing (more about this later).

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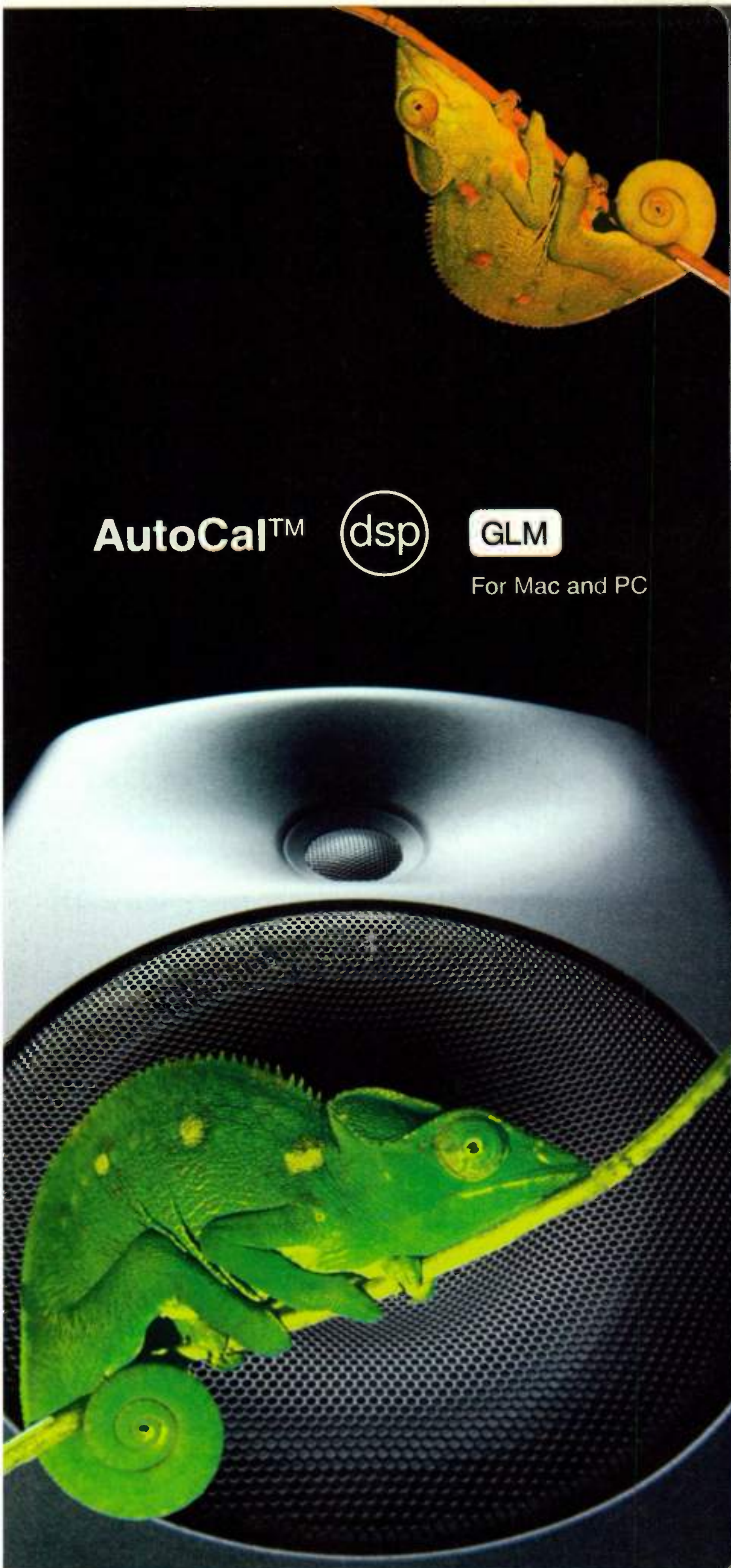

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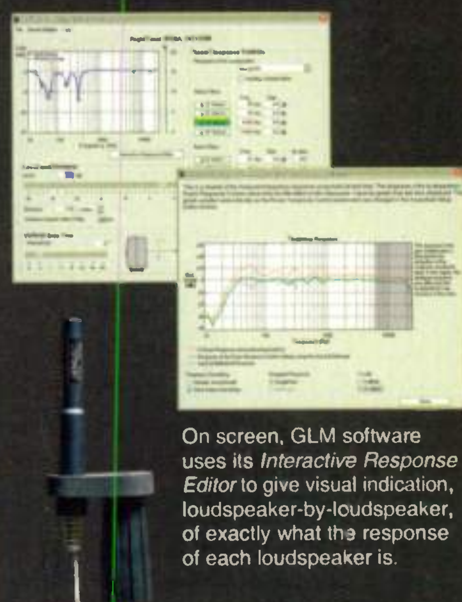

On screen, GLM software uses its *Interactive Response Editor* to give visual indication, loudspeaker-by-loudspeaker, of exactly what the response of each loudspeaker is.

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(Mac) is a system that includes a phone interface and software. It digitally records from the phone (and is also compatible with most VOIP services) and has a built-in limiter that allows it to produce recordings in which both parties on the phone call are at approximately equal volume. The system costs well under $200, and for an additional $50, the company offers a Podcast package that also includes BIAS Soundsoap 2 noise-reduction software and Peak Express, a 2-track editor. I haven't used Phone Valet, so I can't offer opinions on its performance.

If you have a larger budget, the best possible solution for phone recording is probably a broadcast phone interface such as the JK Audio Broadcast Host. It plugs between the phone line and the phone itself and has a number of I/O options (XLR and ⅛-inch). The ⅛-inch jack splits your voice onto one side and the other person's onto the other. You have separate volume controls for each.

Finding Your Voice

Besides interviews, many Podcasts include introductory and transitional voice-overs (VOs) as well as bumpers, which are transitions (usually musical) that go before and after segments.

VOs can be recorded in a multitrack audio program and then placed in their proper spot for the final mix of the Podcast. Record VOs using a good-quality cardioid mic (condenser or dynamic). Use a pop screen to minimize plosives, and place yourself within a few inches of the mic (a little closer than if you were recording yourself singing). Getting close to the mic lets you take advantage of the proximity effect to help give you that larger-than-life "radio voice."

As with in-person interviews, record your VO in as quiet an environment as possible. Do not use figure-8 or omni polar patterns, because they will pick up a lot of room sound. Set your mic pre so that you have plenty of gain, and record at a healthy level without clipping. You don't want to have to boost the volume too much when mixing, because this can increase noise.

FIG. 2: The arrows point to the breaths in this waveform display from an interview track.

Bumper music can come from a number of sources: original music that you already have on hand, music you compose specifically for the Podcast (bumper music can be very short—under 15 seconds), or royalty-free stock music, for example the "jingle tracks" that come with Apple Logic. Do not use another composer's copyrighted material unless you have express permission, or you'll be violating copyright law.

To give you an example of the construction of a Podcast, here's the sequence of events for a hypothetical two-interview Podcast:

- Intro music with VO. Music is "ducked" (temporarily reduced in level) when VO starts.
- Music fades out, then introductory VO tells what will be in the Podcast.
- Music and VO to introduce interview 1.
- Interview 1.
- Bumper music coming out of interview 1.
- VO to introduce interview 2.
- Interview 2.
- Ending music with VO. Music is ducked when VO starts.
- Ending music fades out.

Of course, your Podcast could be a lot simpler. It's totally up to you.

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Hum Bug

If any of your interviews or VOs end up with significant background noise, consider using a noise-reduction plug-in. One affordable solution is BIAS Soundsoap 2 (Mac/Win), which lets you remove various noise types using automatic or manual settings. There are a number of noise-reduction software products on the market, including plug-ins like Soundsoap and Waves X-Noise and the standalone RX from iZotope. Some host programs, like Apple Logic Pro 8 and Sony Creative Software Sound Forge, come with noise-reduction

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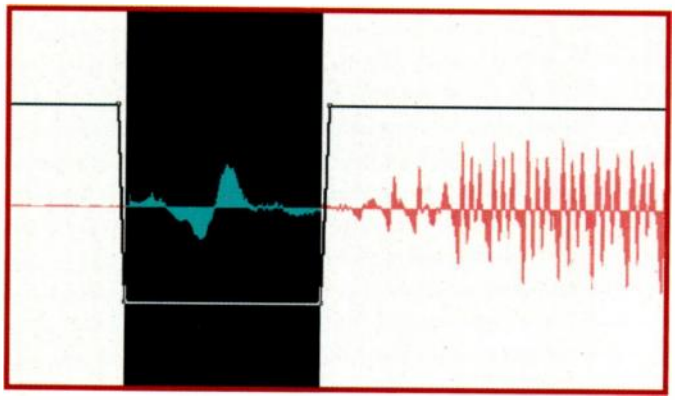


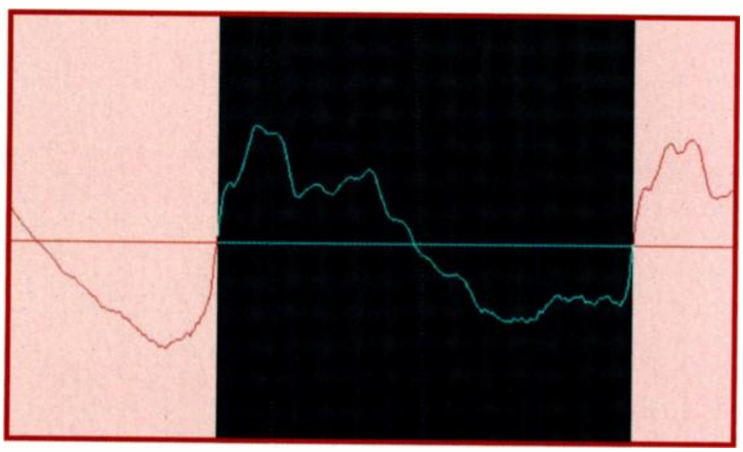
FIG. 3: The shaded area shows the plosive part of the waveform, which is being reduced using Pro Tools' volume automation.

plug-ins built in. These products do an excellent job of detecting and removing noise, but use them judiciously—if turned up too high, software-based noise reduction can make a voice sound very unnatural.

If you don't have noise-reduction software, there are often situations where you can remove problem noise using EQ. If, for instance, the offensive noise is steady at a fixed frequency—say, a 60-cycle hum—you can use an EQ plug-in to notch it out. Set your EQ for as narrow a bandwidth as possible (use the notch filter setting if there is one), set it for 60 Hz, and cut it to the maximum amount allowed. If you're not sure what the frequency of the offensive noise is, set your notch and sweep through the frequencies until you hear the noise lessen. You may not be able to remove all of it this way, but you can probably lessen it.

Another way to go, if the noise is of the low-frequency variety, is to filter out frequencies below about 100 Hz using an EQ plug-in. Your settings will depend on the frequency of the noise and the effect of the filter on your program material. You may lose a bit of bottom end from the voices in your Podcast, but if it gets rid of annoying noise, it's well worth it. Just don't overdo it and make the voices sound tinny.

FIG. 4: By cutting out the center (shaded) portion, you'll be joining the left and right segments at a zero-crossing.



Sweating the Details

When you listen back to interviews, you'll likely notice a lot of glitches including loud plosives, clicks, or just inarticulate phrasing (such as "um's," "uh's," and excessive "you know's") and overly loud breaths. Unless you're going for an audio vérité, warts-and-all production, you'll want to edit out most, if not all, such anomalies.

I'll tackle breaths first. There are a number of ways to deal with them, including deleting them completely, or reducing their volume with automation or the Change Gain command (or equivalent). The key is to do it so that the result sounds natural. After a while, you'll be able to quickly spot breaths in the waveform display just by looking at them (see **Fig. 2**).

One effective method for getting rid of breaths in a natural-sounding way is to copy a little bit of room tone (ambient noise during a pause when nobody is speaking), and then paste it in to replace the breath. That way, the breath is gone, but there's still a pause between the words. If you simply cut the breath out and delete the space where it was, the speaker's phrasing may seem rushed and it may sound like you made an edit, which you want to avoid whenever possible.

If you completely reduce the volume of a breath, you'll hear an unnatural dropout at that point because there won't be any ambient noise. That is why using room tone is so useful (see **Web Clip 1**). MOTU Digital Performer has a feature called Smooth Audio Edits that places room tone (which you can designate or it can find for you) in between edit points automatically, with crossfades added.

If you want breath removal done automatically, Waves makes a plug-in called DeBreath, which is part of the company's Vocal Bundle. DeBreath is designed to automatically detect breaths and allow you to separate them from the program material.

Pop Goes the P

The noises made by loud plosives are distracting and should be reduced. Typically, you can't just cut them out, because the word will not sound right with a consonant sound removed completely. Reducing them is the best strategy. You'll find that it's pretty easy to spot plosives due to their distinctive, dense waveshape (see **Fig. 3**).

Here are the steps for reducing plosives:

1. Find the offensive plosive and zoom in on it.

2. Select the plosive part only (check by auditioning your selection).

3. Reduce the level of the plosive using either volume automation or the Change Gain command in your recording or editing software. Either way, reducing the level of the plosive by about 6 to 8 dB will usually do the trick, though sometimes you have to reduce it quite a bit more.

No matter what you're editing, always apply short crossfades at the edit point. Often crossfades will smooth out your edits. But if you have a click or pop at your edit point and crossfading doesn't help, you can often clean it

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up by changing your edit point slightly so that it happens on a zero-crossing, which is the point where the signal crosses from negative into positive or vice versa. Here's what you do: zoom in on the waveform until you find the zero-crossings for both sides of the edit. Then make your cuts so that the two newly joined sections meet at a zero-crossing (see Fig. 4). That will usually take care of the pop.

Level Playing Field

Another area that you're likely to have to pay a lot of attention to is levels. Human beings don't talk at perfectly even volume over the course of a conversation, so spoken-word recordings are going to have level variations you need to address. Whether it's voices trailing off at the end of sentences, loud interjections, or other inconsistencies related to volume, you're sure to find plenty of points during the course of an interview or voice-over where you need to make adjustments. You don't want the listeners to have to adjust their volume controls while listening to your Podcast.

A digital audio sequencer's volume automation is a great way to even out levels. You can also use compression for controlling dynamics, but be careful: too much compression can bring up the level of noise on spoken-word tracks. I generally lightly compress my voice-over and interview tracks using a plug-in that doesn't add a lot of color, such as Waves' Renaissance Compressor.

Mix It Down

Rather than dealing with the aforementioned editing issues during the final mix of my Podcasts, I find it easier to edit and mix my interviews separately and then import the mixed stereo files into the final Podcast mix and place them where I want in the timeline along with any transitional music elements (see Fig. 5).

I generally wait to do any introductory, ending, or transitional VO segments until I've got all these other elements in place. You have better context for writing or improvising your voice-over once you have the Podcast's structure fully fleshed out.

If you've premixed your interviews, the biggest issues you'll be facing in your final mix will be evening out levels between the various elements. Be particularly careful when you're mixing a voice-over that's going over a piece of music. Use volume automation to duck the music down significantly for the duration of the voice-over, and then bring the level back up if the music continues after the VO. Typically, you'll want to smoothly fade out music elements to end them.

When you think your mix is ready to go, double-check the consistency of the levels between the beginning, middle, and end of your Podcast. I do this by setting the volume of my monitors to a comfortable level for the intro section, and playing back random snippets from various parts of the Podcast (without touching the volume control) to make sure the levels are even.

Overall, you want your Podcast to be at a healthy level without clipping. Pay attention to gain-staging protocol: don't raise your

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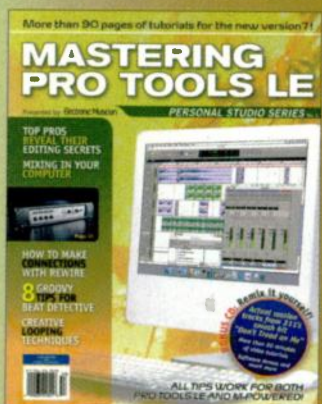

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master fader over 0 dB; boost your individual channel faders instead. When you've finally got your mix sounding the way you want, bounce it to disk (still at 24-bit).

One Last Pass

After you've mixed down to stereo from your multitrack, I would suggest taking a break from the material for a couple of hours or, even better, overnight. As with a music mix, that time away from it will allow you to regain your perspective. After you've given it a break, load your file into a 2-track editor application. Check once more for any offensive breaths, pops, or vocal stumbles and mismatched levels. All of those glitches can be addressed in a good 2-track editor.

When you're satisfied, convert it to a 16-bit file and then into MP3 format. (Many applications will let you do this in one step.) MP3 is the universal format for Podcasts and is compatible with both Macs and PCs.

If your Podcast is music centered, I would recommend encoding your MP3 file in stereo at a minimum of 128 Kbps (kilobits per second). If talk is the focus and music is secondary (or nonexistent), you can probably get away with a mono, 64 Kbps or 48 Kbps file. Remember that the higher the bit rate, the better the sound quality, but the longer it will take for users to download or stream the Podcast. Also, make sure to add ID3 tags, which are metadata embedded in MP3 files to identify them (see the sidebar "Tag, You're It").

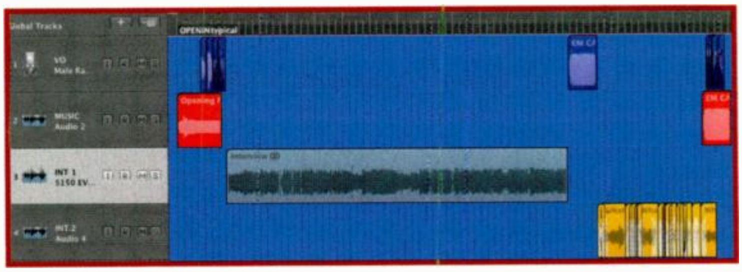

FIG. 5: Place the various elements into your sequencer's timeline and arrange them as you wish. Then record transitional VOs before mixing.

On the Web

Taking the time to make the audio in your Podcasts sound as good as possible will pay off in improved enjoyment for your listeners. Of course, content is the most important aspect of a Podcast, but if the sound quality is lousy, with stray noises, jumpy volume levels, and bad-quality interview recordings, people will be less likely to want to download, stream, or subscribe to it again. I try to make my Podcast production as radio-like as possible. If it sounds like it could be on the air, then it's ready for the Web. **EM**

EMWEB CLIPS

Mike Levine is an EM senior editor and the producer of "EM Cast," the twice-monthly, interview-based Podcast available at www.emusician.com/podcasts.

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FUN and GAMES

By Nick Peck

Before seeking work in any industry, you naturally will want a clear idea of what you're getting into. You are likely to inquire about the hours, the balance between creative and technical tasks, your employer's expectations, and what a typical workday is like. You might also debate whether you're better off as an employee or as a freelancer.

It's no different if you are seriously considering contracting or employment in game audio. Whether your focus is on music composition and production or sound design, working in game production can be challenging, exhausting, and rewarding. Tales of nightmare hours and incredible stress abound—as do success stories. But what's the game-audio world really like on a day-to-day basis, and what are the differences between being a game-company employee and being a freelancer?

To help you get a handle on what it's like to work in the game-audio business and what characteristics make up a successful game-audio professional, I'll relate some of my experiences in the field, as well as offer perspectives from several other successful and highly skilled industry veterans. Together, we'll explain what to expect—and what will be expected of you—once you land that killer game-audio gig.

The Way In

My path into the world of game audio was typical. I grew up at the dawn of the information age, dividing my time between keyboards on the piano and the Tandy Radio Shack TRS-80 personal computer. I studied computer science in college before earning degrees in electronic music. Through a succession of multimedia computer-programming and audio-production jobs, I built up my project studio, while constantly networking with people in the game and music industries. Freelance gigs in multimedia became plentiful and were close to what I wanted to do, but not right on target.

Then the LucasArts Entertainment Company called, in search of another sound designer for the new game *Grim Fandango*. (For the inside story of scoring *Grim Fandango*, see "Dance of the Dead" in the September 1999 issue of EM, available online at www.emusician.com.) I went in, passed the tests, and was soon off to the races at my dream job. My work at LucasArts led to a continuous stream of positions and freelance assignments in film and video-game production.

The game industry is small and incestuous. You hear the same names over and over as industry veterans migrate from one company to another. Everyone in the industry ends up working on some great games and some not-so-great ones. It's good to remember that no matter how fantastic the game might be, it probably has a shelf life of only a few months—far less time than it

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took to develop the game. With that in mind, I have found that the most rewarding part of the game-sound process is the day-to-day work flow and the relationships you forge in the crucible of hard work. After all, you will spend more of your waking hours with your colleagues than you will with your family.

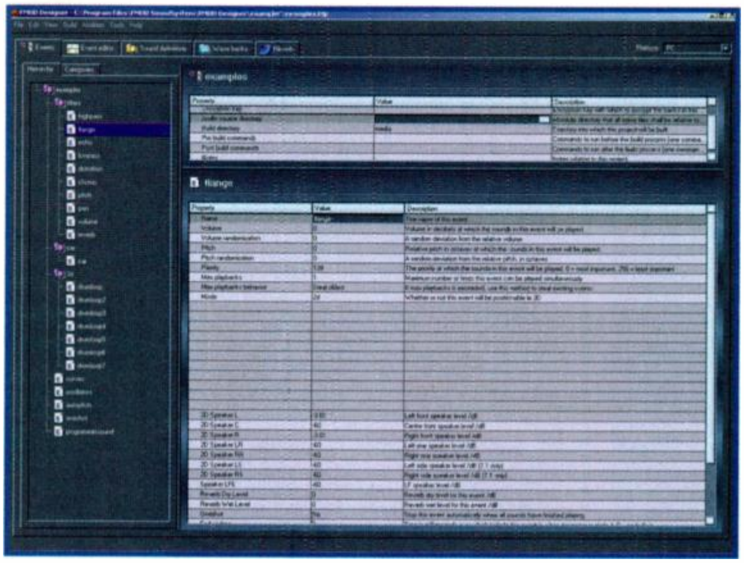
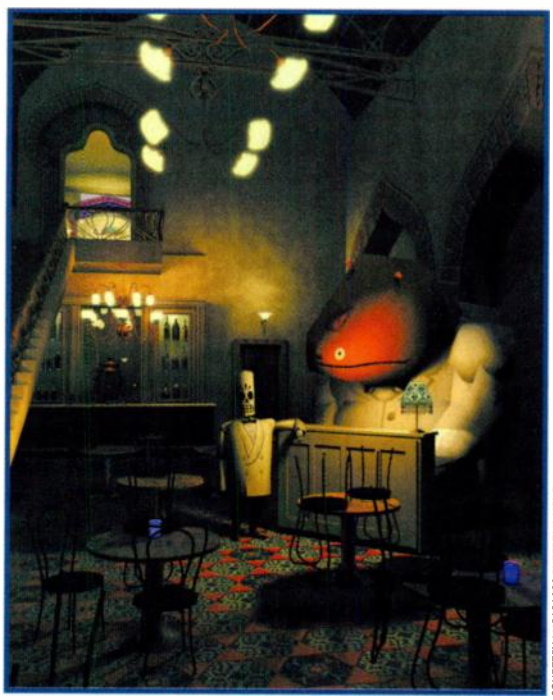


FIG. 2: Audio-integration middleware solutions like FMOD are terrific tools that the fledgling game-audio pro should learn.

Getting the Gig

Before you can build those relationships and create sound for those great and not-so-great games, you have to garner your first gig. Every game-audio artisan had to cross that threshold at the beginning of their career. So what can you do to get that job? What skills do the audio directors of the world look for in a potential freelance or employee candidate? In general, successful candidates have skills in three areas: content creation, content integration, and interpersonal dynamics.

If you aspire to be a sound designer or composer, your first task is to demonstrate that you have the chops to design or compose—and that takes lots of practice. If sound design is your passion, then grab a field recorder, go into the world, and grab a load of sounds. Then manipulate the daylights out of them back in your studio and experiment with layering and mixing until you start to gain an intuitive sense for what works.



COURTESY LUCASARTS ENTERTAINMENT

FIG. 1: Author Nick Peck's first big break in game audio came when he was hired as a sound designer for LucasArts Entertainment's colorful *Grim Fandango*.

Electronic Arts Redwood Shores (EARS) senior audio artist J. White looks for candidates who go the extra mile by creating original sounds. In his opinion, "nothing is a greater turnoff when listening to a reel than hearing the same library sounds being used again and again."

Aspiring composers must spend hours in the woodshed composing. Fortunately, a clear, well-trodden path helps pave the way to success: music school. Without exception, every successful game composer I know (and many sound designers) has a degree in music. I have no doubt that there are exceptions, but I don't know of one. There is nothing like the distraction-free atmosphere of a practice room to help lay the bedrock of one's musical craft.

It all starts with your instrument, a pencil, and paper. No reverb algorithm or cool Reason patch will substitute for that time spent writing musical notes. "You will need tech chops and business skills," observes freelance composer Mark Griskey, "but your ability as a composer will ultimately be judged based on the music you deliver. Look for any opportunity to compose music that you can find. Study music from as many sources as you can. Ask questions of all of your teachers, friends, and colleagues."

The Challenge of Integration

Once you've created the content, you need to get it into the game. Audio integration is a huge part of the job, typically requiring as much or more time as it took to make the sounds in the first place. Though you don't have to be a hard-core C++ computer programmer, knowing a programming language such as Lua (www.lua.org) can serve you well, as it did for me on games such as *Grim Fandango* (see Fig. 1) and *Escape from Monkey Island*.

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White is a fan of graphical programming environments such as Cycling '74's Max/MSP (www.cycling74.com), while LucasArts Entertainment audio lead David Collins suggests buying a game that ships with a game editor (*Unreal* and *Star Wars: Republic Commando* are examples) and then going in and creating your own audio cues within that environment. Several audio-integration middleware solutions, such as Firelight Technologies' FMOD (www.fmod.org; see **Fig. 2**) and Audiokinetic's Wwise (www.audiokinetic.com; see **Fig. 3**), offer free demo versions for download. Learning one or more of these systems will pay dividends regardless of what system you use for audio integration, as many of the core concepts will transfer from tool to tool.

Attitude Adjustment

As with many jobs, interpersonal skills are enormously important. Years ago, during a casual conversation, a highly successful video-game executive tossed off a saying that has stayed with me: "Your attitude determines your altitude." Everything about being a successful game-audio pro is encapsulated in those five words.

Talented, knowledgeable contributors are so plentiful as to seemingly grow on trees. A good attitude differentiates the successful candidate. Life is too short to deal with arrogance, with prima donnas, or with people who backstab or belittle. I love to hear "Can do," but I am just as happy hearing "I don't know if I can do it, but I'll look into it and get back to you quickly." My golden rule for game-audio success is, "Do what you say you are going to do, when you say you are going to do it, with a minimum of drama."

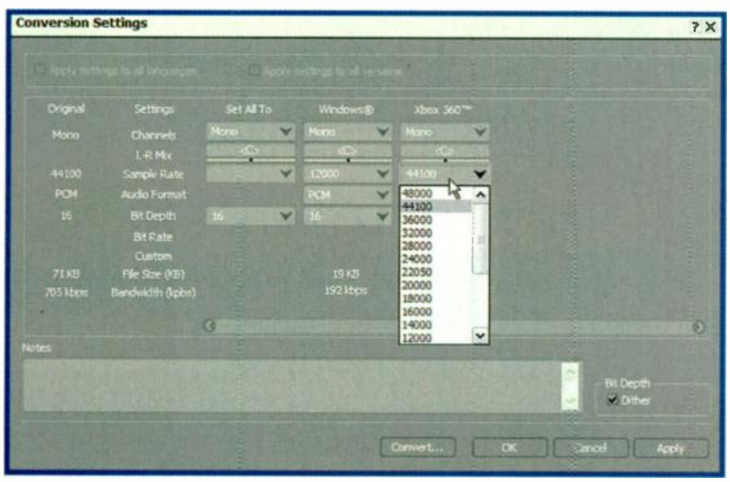

FIG. 3: Like FMOD, Wwise is an audio-integration middleware tool for games. Shown here is Wwise's audio conversion settings screen.

FIG. 4: Pro Tools is the DAW most frequently used by game-audio pros.

Sony Computer Entertainment of America (SCEA) sound-design manager Ken Felton looks for the basics in a potential hire. "Have good communication skills," he advises. "Use a spell-checker. Wear clean clothes. Be polite. Show up on time. It's pretty simple."

Tools of the Trade

Whether your goal is to be an employee or a freelancer, you should have and thoroughly understand several basic tools. One required tool is a computer, of course. You can use a Macintosh or a Windows PC to create content, but PCs are usually required for audio-integration tasks. EARS audio director Paul Gorman and I use Macs for production and PCs for integration. On the other hand, Jim Diaz, a senior sound designer at Activision's Underground Development studio, uses Windows PCs for both asset creation and integration.

Digidesign Pro Tools (see **Fig. 4**) is the DAW of choice for virtually every game- and film-audio professional I have spoken with. Simply put, you must learn how to use it.

If your focus is on composition, you will probably need some specialized music-production software as well. Griskey employs Apple Logic Pro for composition and Sibelius Software's Sibelius scoring software for notation. He also uses most of the major orchestral sound libraries.

Similarly, sound designers require additional tools of the trade, notably a field recorder, microphones, and a couple of general-purpose sound-effects libraries to help fill the gaps in your personal, custom library. (For a comparative roundup of field recorders, see "Playing the Field" in the October 2005 issue of EM.) Gorman also points out the importance of having "a neutral, balanced listening environment where you design assets and listen to them in the game."

Let's assume that you've worked on your content-creation, integration, and interpersonal skills and that you've put together a studio where you've mastered the

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technology. You are ready to rock. Now it's time to get your demo reel together, network with everyone you know, watch the job-board postings (www.gamasutra.com is a good place to start), and be ready for that crucial interview. If you follow the suggestions I've made and you have the talent, that gig is as good as yours.

Of course, you have to find the gigs first. Fortunately, several of the major game companies, such as Electronic Arts and Sony Computer Entertainment of America, post job information on their Web sites (see **Fig. 5**).

A Day in the Life

Now that you've settled on a career in game audio, what can you expect? What is a typical day in the life? The answer is that there is rarely a "typical" day; a variety of challenges crop up, and which issues you deal with depends on your position and the phase of project development.

Content creation and integration are certainly part of your day, and the lower you are on the totem pole, the larger the proportion of your day you spend on these creative tasks. As you rise through the ranks of responsibility, more of your time is spent managing people and resources, going to meetings with people in other disciplines who work on their aspects of the game, and evaluating the content created by your team as it makes its way into the game.

For example, as of this writing, David Collins and Paul Gorman each supervise the work of ten content creators for their current games (*Star Wars: The Force Unleashed* and *The Simpsons*, respectively). That is the same-size crew that you would see handling audio on a feature film. Different types of tasks often take precedence at various stages of development. As a sound designer, I always set aside time for field recording at the beginning of a project so that we have a new library of raw content to work with.

Jim Diaz likes to break a game project's development cycle into three phases: spotting (or preproduction), production, and postproduction. The spotting phase is filled with meetings where the audio team learns about the project. Lists of audio assets are created and stored in databases. Members of the team work out preliminary development schedules and approach potential contractors.

FIG. 5: Employment opportunities are often posted on game companies' Web sites. Here are Sony Computer Entertainment of America's game-audio job openings as of late September 2007.

Found 7 Employment Opportunities
Keywords: N/A Location: All Locations
Job Category: Audio / Music / Video
Req Number: N/A

Click on each title to view details.

Position Title	Requisition Number	Location	Job Type	Date
INTERNSHIP (Fall 07'): Audio Intern	SF11162	CA, San Diego	Part-Time	08/23/2007
INTERNSHIP (Fall 07'): Audio Intern	SF11163	CA, San Diego	Part-Time	08/23/2007
INTERNSHIP: Music Editor Intern	SS11081	CA, Foster City	Full-Time	08/17/2007
INTERNSHIP: Music Editor Intern	SS11082	CA, San Diego	Full-Time	08/17/2007
Sr Sound Designer (In-Game)	GR10861	CA, San Diego	Full-Time	08/09/2007
Sr Sound Designer (In-Game)	GR10862	CA, Santa Monica	Full-Time	08/09/2007
Sr Sound Designer (Post Production)	RR11003	CA, San Diego	Full-Time	07/16/2007

Click here to submit a general application.

The primary sound-effects design and music composition take place during the production phase. The team creates and integrates assets, slowly filling the new virtual world with sound as the game levels are created.

Postproduction is filled with volume tweaks, sound-effects revisions, bug fixes, and playing the game over and over while polishing the soundtrack to a high shine. Eventually, the game is ripped from our clutches, packaged, and placed on store shelves.

The Freelance Perspective

Freelancers spend a good portion of their day creating content, but again, that is only the first of their tasks. Mark Griskey and Julian Kwasneski (founder of independent game-audio firm Bay Area Sound, Inc.) work remotely, so they exchange a lot of information via email. As contractors, they must send invoices for work completed, create quotes for prospective projects, and iron out technical implementation details. The all-important schedule must be continually managed and updated.

Once the development team starts hearing the audio in the game, the feedback process begins, with inevitable revision requests that must be managed. You have to maintain a delicate balance here: feedback is critically important, and you have to allocate time for revisions within the development schedule or you will quickly become double booked, trying to revisit old content while creating new materials simultaneously.

As an audio director, I believe there is an art to delivering feedback along these lines. I recommend that audio directors maintain a single line of communication with the freelancer, always keep their eye on the schedule, and know when something is good enough. Of course, this process is made much easier when working with high-caliber freelancers, such as Kwasneski and Griskey, who routinely hit the ball out of the park the first time around.

In the recent past, out-of-house freelancers were relegated to content-creation duties only, because the asset-integration tasks often relied on custom solutions that required being on-site. These days, though, freelancers are being used for integration as well.

Ken Felton prefers all-in-one sound-design contracts, where the freelancer provides the assets prebundled into sound banks that drop right into the game. Collins has a current off-site contractor who does nothing but asset integration. This development style can be excellent for freelancers who have highly developed studios and prefer to work off-site. Griskey, for example, lives in a tiny, remote beach town in Northern California and handles the vast majority of his business dealings via the Internet.

Money (That's What I Want)

As I'll discuss later, you have to love this business to have the dedication to do it. Nevertheless, the goal for a working professional is to make a fair living. While you are

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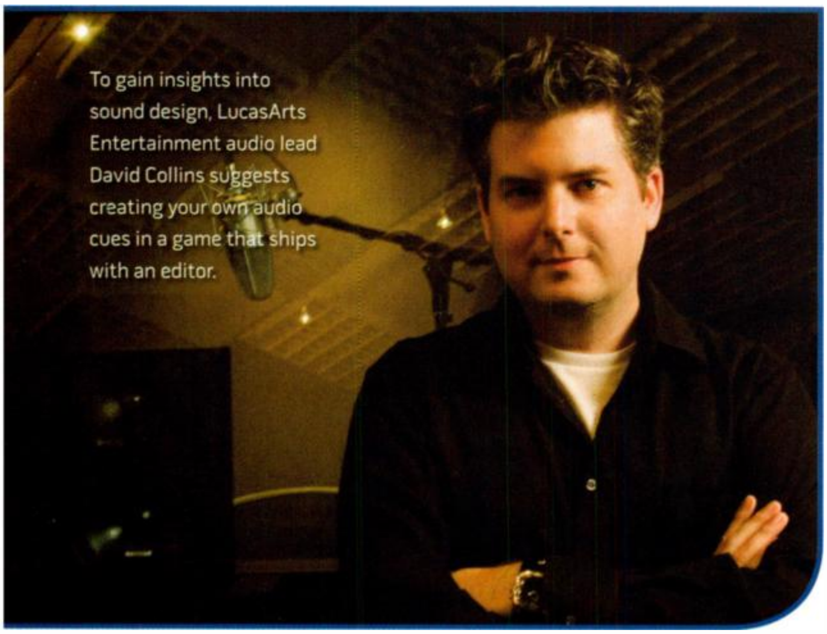
unlikely to get rich in game-audio production, a decent middle-class living is certainly attainable as a freelancer or as an employee.

You'll find a wide variance in salary ranges, depending on a company's geographic location and financial health and an employee's experience. A small startup in a region with a low cost of living might offer as little as $30,000 per year for a junior sound designer, whereas an audio director with some major titles under their belt, working for a major company in an expensive area, can earn a comfortable six-figure salary. Typical salaries for sound designers range between $50,000 and $90,000. Some big companies also offer paid summer internships for students. Expect to make between $10 and $20 per hour if you are lucky enough to land one.

In the big leagues, benefits such as stock options, employee stock-purchase plans, and bonuses can sweeten that pot. Of course, the more common employment benefits also have value: paid vacation time and medical, dental, and vision packages are expenses that freelancers have to pay for on their own.

Freelancers have a different set of challenges, though financially they can end up in about the same place. The average rate for an experienced individual freelance sound designer is between $50 and $65 per hour. If you are a well-established freelancer, working with your own gear at home, and with a variety of big games to your name, you may be able to negotiate your rate upward to between $75 and $100 per hour. Your ability to command such rates frequently depends on the size of the project and how desirable you are to the company. If you have a proven track record with the company, you have more bargaining power.

Not surprisingly, lower-budget projects often have less money set aside for freelancers. If you are just starting out and are trying to woo a small company, try aiming for $25 per hour. Next time, when you and the company have some experience and revenue under your belts, you can try to renegotiate for a higher hourly rate.



To gain insights into sound design, LucasArts Entertainment audio lead David Collins suggests creating your own audio cues in a game that ships with an editor.

JANE RICHEY

Freelance composers usually charge by the minute of finished music rather than by an hourly rate. That number can range from $800 to $1,500 per minute, depending on the experience and reputation of the composer. Budgets for orchestras, recording studios, and additional specialists are negotiated on top of the composer's fee.

Striking a Balance

It is impossible to consider a career in the game industry without giving some thought to life/work balance. Rightly or wrongly, the industry has gained a reputation for promoting excessive work hours, setting punishing schedules, and expecting very high output from everyone involved. When a game enters its "crunch period," which often begins several months before the project's completion, you may face grueling hours, seven-day workweeks, and even an occasional all-nighter. During the past 15 years, I've slept on the floor of my office a half-dozen times and have worked past midnight at least 200 times.

As an in-demand freelance composer, Griskey has had his share of taxing schedules and insane deadlines. But, he observes, "it's the same in the film and TV worlds. It's hard to keep the balance between work and personal time, but luckily I get to do something I really enjoy for a living, so it makes the craziness easier to handle."

Peer pressure plays an important role in working long hours. Successful professionals take pride in their sense of responsibility, never wanting to let down the game team. Audio production is a customer-service discipline, and it is supremely important to do what you say you are going to do, when you say you are going to do it.

However, there comes a point in the lives of many game-audio pros when a new priority trumps the overarching desire to please the game team at all costs: parenthood. The moment my son Julian was born, I knew that I had a new responsibility that was more important than any other: to give him the love, guidance, and personal time necessary for him to grow straight and true. Ironically, part of this responsibility includes having a stable income stream and medical benefits to provide for his needs, so working has become more important than ever before.

Interestingly, I think I've become a more effective audio director as a result. The desire to spend quality time as a parent compels me to find more-effective time-management strategies, whether it be through creating realistic schedules, hiring more freelance help when necessary, or simply getting it right the first time more often. There are still unavoidable crunches, but setting appropriate work boundaries can result in a better product, because a fresh, rested group is more creative.

Kwasneski, who is a single parent of two, agrees.

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"Sometimes there is no avoiding a crunch," he admits, "but I find that if you plan your resources well and are not overextended, you can avoid some of the more nightmarish times. As with anything in life, you need to know when to say, 'Enough.' Remember that it's easy to burn out if you don't allow yourself downtime."

As the game industry matures, more professionals across all disciplines face similar family needs. According to Collins, LucasArts Entertainment's management realizes that they can't just burn out their staff because, he says, "we would lose all of our talent that way. We are very careful to make sure that we have the staff that we need and that we work reasonable hours and keep crunch times down to a minimum." Felton feels that Sony shares that approach. "Some companies do a better job than others supporting a healthy work/life balance," he says. "I'm happy to say that SCEA sees value in personal time off." Such a philosophy represents a significant improvement over the situation found in much of the game industry just a few years ago.

You Gotta Love It

I recently saw a drummer packing up after a solo street performance. His setup was enormous, including a huge kit, a gong, ancillary percussion, a large P.A., merchandise tables, and a portable riser. The effort to simply put everything away after giving his all onstage was impressive. He mentioned that he did about 20 performances per month. I marveled over the sustained effort that he put in. His response was simple: "It beats flipping burgers."

You have to feel driven to work in game audio. It can be enormously tedious and stressful and filled with disappointments and difficulties. However, it can also be supremely satisfying when you realize that you have just contributed materially to an excellent game that millions of people will enjoy. That will put a spring in your step for weeks—which is how long you will need to recover from the crunch period that was required to complete the project. But you will be making your living doing what you love—and it certainly beats flipping burgers! **EM**

Nick Peck is a sound designer, composer, audio engineer, and keyboardist. He is currently serving as audio director of Underground Development, an Activision studio, and has been involved in interactive audio and film for over 15 years. Peck's new album of jazz/funk Hammond organ, Fire Trucks I Have Known, *is available on CD Baby.*

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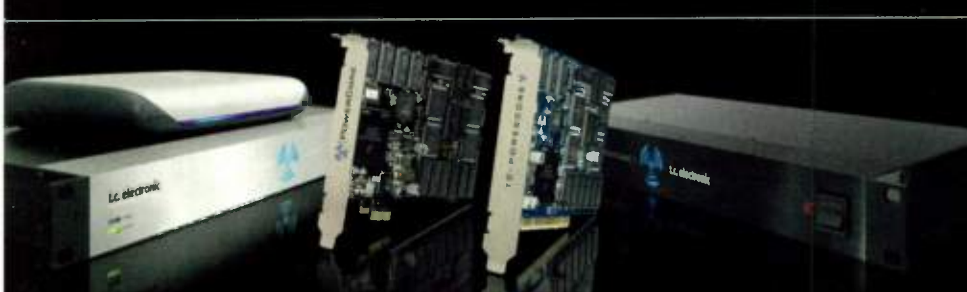

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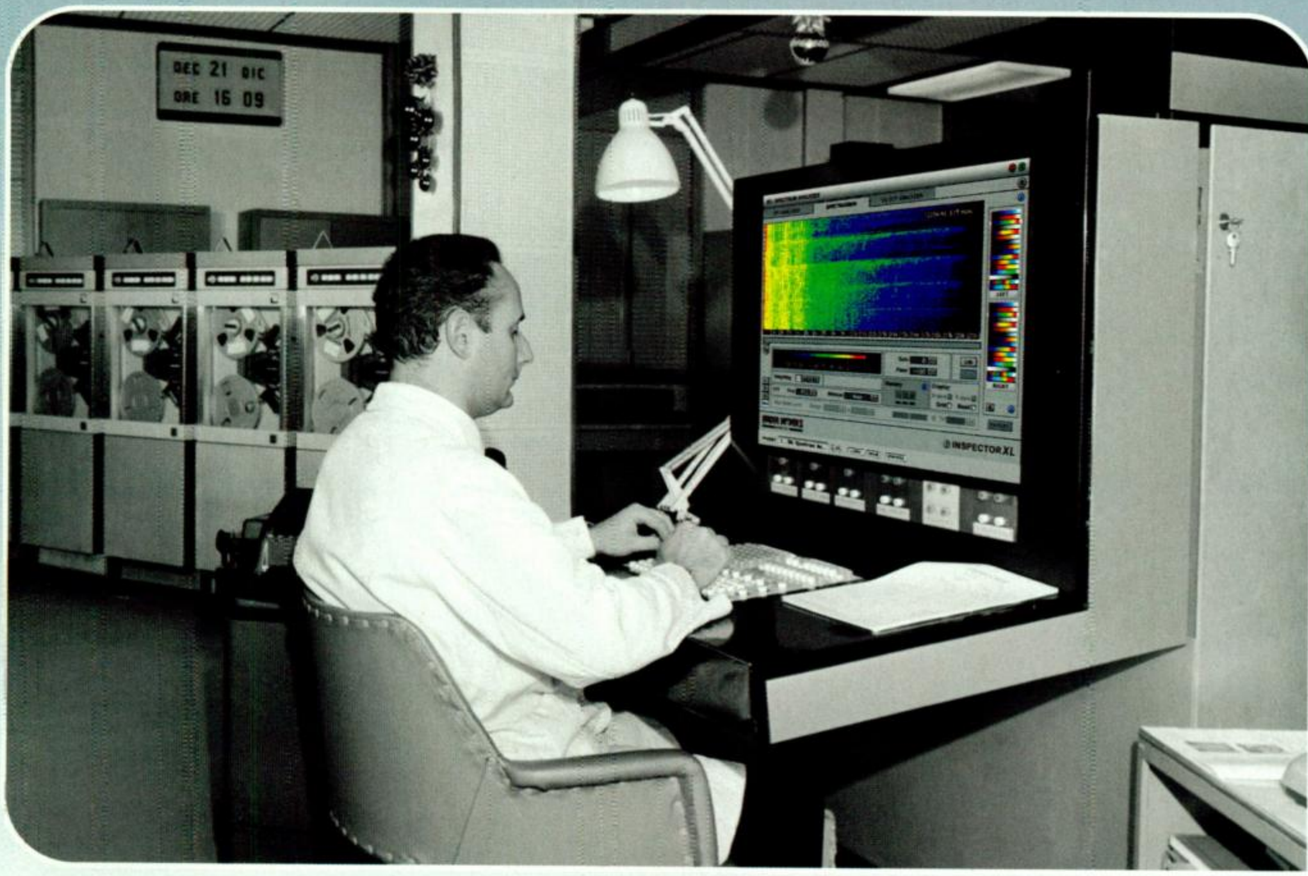

JUPITER IMAGES

Analyze This

By Larry the O

Metering and analysis programs help identify and avoid sonic problems.

We rightly rely on our ears as the final arbiters of recording and mixing issues. Sometimes, however, identifying the cause of an audible production problem by ear can be difficult. Metering and analysis tools provide quantitative information that allows us to observe an audio signal from various perspectives, which can be the key to avoiding problems and identifying pesky artifacts.

Today, analysis tools have emerged from research laboratories to become affordable and readily available in plug-in suites such as BIAS Reveal (part of the company's Master Perfection Suite) and Roger Nichols Digital Inspector XL. The tools are embedded in editors and processing software like Steinberg WaveLab, iZotope Ozone, and Sony Creative Software Sound Forge. You also can find them in standalone form in the godfather of modern desktop audio analysis, Metric Halo SpectraFoo, as well as in newer entries such as Audiofile Engineering Spectre and SuperMegaUltraGroovy FuzzMeasure Pro. You can even find shareware and freeware analysis tools, including a scaled-down Inspector from RND.

Although the tools are available, how many people really know how to use them? Most users understand level meters and know the difference between peak and RMS (averaged) levels, but far fewer can read a spectrogram or a correlation meter, or even know what these tools are.

To help you understand computer-based metering and analysis tools, I'll discuss what they are, when you would use them, and how to make sense of what they say. I can only scratch the surface here, but hopefully that will be enough to embolden you to try out and learn more about some analysis tools. I'll look primarily at affordable software tools for signal analysis, rather than

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at tools for system or component measurement, although the line between the two types can be fuzzy. Hardware analyzers and high-end, specialized software such as acoustical-analysis programs have been excluded.

A Few Distinctions

Mathematically speaking, an audio phenomenon can be seen as a function of time or of frequency, and there are ways to convert between these two domains. Most audio-analysis tools exploit this concept to let you view the same data in multiple ways, each of which yields its own insights. For our purposes, we can break the available audio-analysis tools into the following categories: level, spectral analysis, phase- and stereo-image meters, transfer functions, and code tools and statistics.

Analyzers are often distinguished by the ways in which they handle time. Analysis tools fall into one of two presentation approaches; I will refer to the first as "now," and the second as "then." "Now" tools are real-time, and they display an analysis of a signal as it happens. "Then" tools show analysis over time, which provides a historical context for spotting trends or episodes in the audio (see **Fig. 1**).

Since histories and averaged values represent analysis over time, some time must elapse before a history or an average can be created. In real-time applications, such as live sound, the latency incurred by these processes is annoying at best, and unacceptable at worst. However, in many cases, such as RMS measurement, the time over which the signal is averaged can be short enough to be almost real-time.

Another distinction occurs between those analysis tools that are available as plug-ins and those that run standalone. Plug-ins are usable within DAW and audio-editor hosts, which makes them well suited for production tasks such as monitoring signals during recording, overdubbing, or mixing. Standalone programs can be used to observe a live signal—sometimes even multiple signals. But here the basic paradigm is different: only a single instance of an analyzer program is expected to be running, and you generally use the program as a bench-test instrument rather than as a production-monitoring tool.

Analysis tools provide broad capabilities, but people's needs for them are often specific. To accommodate such needs, most analysis tools are highly configurable, offering graphic preferences, variable parameters for processes such as Fast Fourier Transforms (FFTs), and parameters for defining quantities (such as the number of clipped samples that constitute a "clip" condition). The more comfortable you become with these tools, the more you will probably customize.

PFFT!

A good place to start working with analysis tools is by looking at the most important analysis technique for audio, which underlies a number of tools: the FFT. FFTs convert data from the time domain into the frequency domain. (An inverse FFT converts in the other direction.) FFTs allow you to look at the spectrum of a signal.

The SpectraFoo manual says, "The FFT algorithm is an efficient means of computing a Fourier transform on a computer. The Fourier transform, developed between 1804 and 1807 by the mathematician Joseph Fourier as part of a study of heat transfer, converts a continuous record of amplitude versus time into a record of amplitude versus frequency. A modification of the Fourier Transform called the Discrete Fourier Transform (DFT) was developed to deal with sampled, rather than continuous, waveforms. The FFT algorithm was developed as an efficient way of computing the DFT on digital computers."

The FFT is the basis for everything from spectrum analyzers to transfer functions. The two key parameters

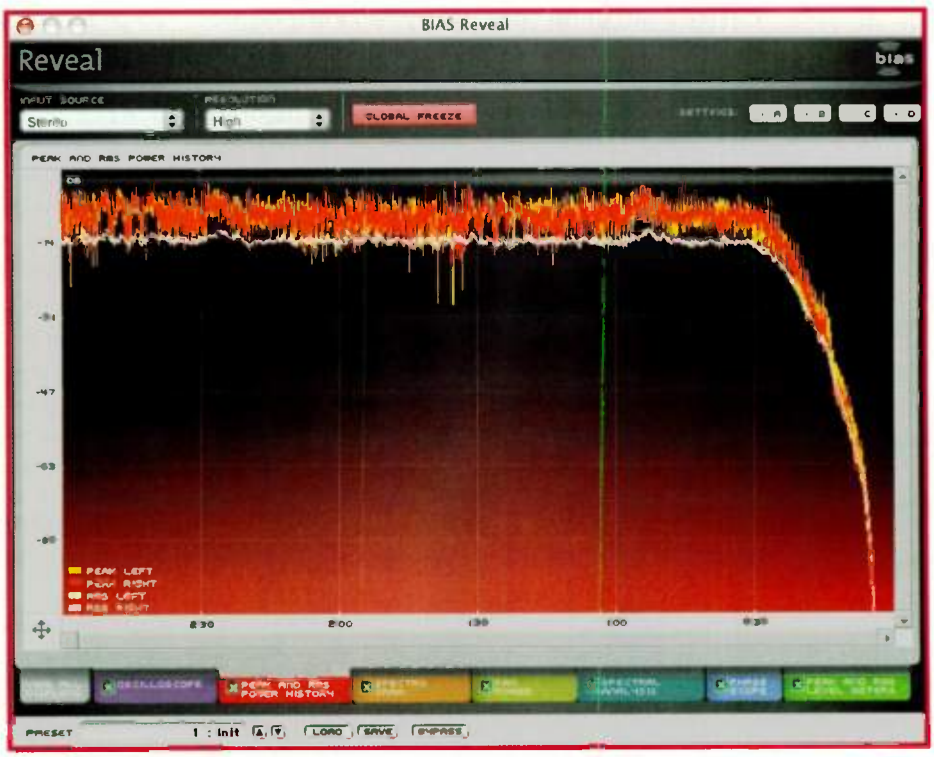

FIG. 1: This BIAS Reveal display shows peak and RMS power histories for each channel. Notice the song fade-out at the right of the display and the roughly 12 dB crest factor.

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that determine FFT performance are block size (the number of samples on which the FFT is performed) and window type (a preconditioning function applied to reduce error in the FFT). For more on FFTs and the impact of these parameters, see the **online bonus material** at www.emusician.com.

ONLINE BONUS MATERIAL

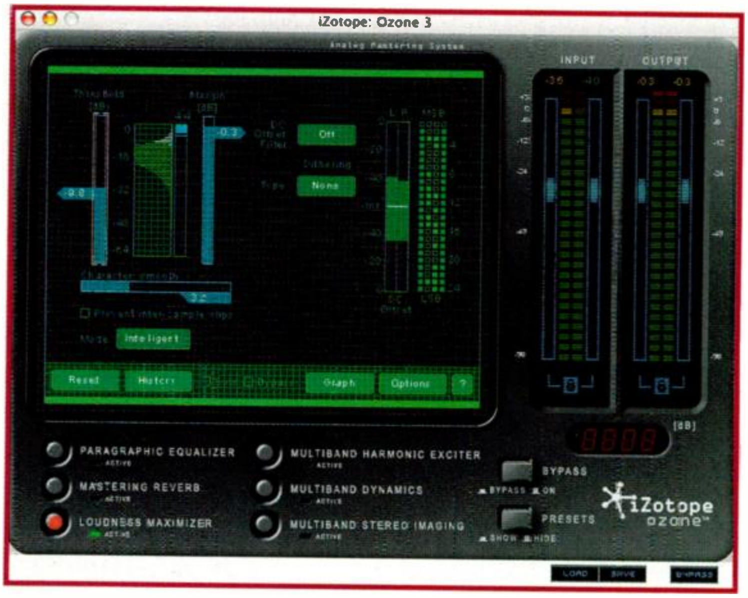


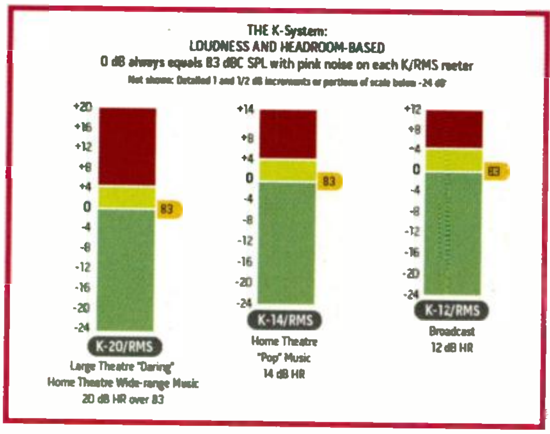
FIG. 3: This screen from iZotope Ozone shows six different metering functions: a histogram to the right of the threshold control, a gain-reduction meter, a DC offset meter, a bit scope, and input and output level meters with clip indicators.

How Loud Is It?

Measuring level is the most familiar metering function. While people frequently say that a level meter shows how loud a sound is, loudness is actually a perceptual attribute that is almost never directly measured. Level meters show signal amplitude or power in decibels and vary in their response characteristics and demarcation.

The ear responds primarily to the average level of sound. Therefore, averaged metering techniques such as RMS metering are useful when you are concerned with how level is being perceived. Having peak-level metering is more critical for equipment that can overload.

With music and most other signals (assuming that they have not already been subject to dynamic compression), a substantial difference typically exists between the peak and average levels, often as much as 20 dB. The ratio between the peak and average (RMS) levels of a signal is called the *crest factor*. Crest factor is important because it determines the amount of headroom that is required. In audio, crest factor is usually expressed in decibels, making crest factor calculations as easy as subtracting a signal's RMS value from its peak value.



COURTESY BOB KATZ

FIG. 2: Mastering engineer Bob Katz devised the K-System scales for level metering and monitoring. This graphic has been reprinted from Katz's book, *Mastering Audio: The Art and the Science* (Focal Press, 2007).

Mastering engineer Bob Katz's K-System for level metering has been gaining acceptance among engineers and metering-software manufacturers. The K-System takes the idea of crest factor to the next level, defining three meter calibrations with different zero levels that indicate varying amounts of headroom (see Fig. 2). The three calibrations are used because recordings in different genres vary in crest factor. Pop-music recordings, for example, are typically highly compressed, producing a lower crest factor and higher average level, so that less headroom is needed on the meter; classical or audiophile recordings typically have a higher crest factor. Therefore, commercial pop recordings might use the K-12 scale, which defines only 12 dB of headroom, while classical recordings might use K-20, which sets zero at –20 dBfs. Whichever scale is used, according to the K-System, your monitors should be calibrated so that zero produces 83 dB SPL in the room. For a more complete explanation of the K-System, see part 2 of the paper "Level Practices" at www.digido.com/bob-katz/level-practices-part-2-includes-the-k-system.html, on Bob Katz's Digital Domain Web site.

Viewing with VUs

Volume unit (VU) meters and RMS-based meters are intended to indicate average levels, but they use different methods to do so. Mechanical VU meters have long integration (rise and fall) times of 300 ms and use mechanical smoothing to achieve an approximation of averaging. RMS meters, in contrast, perform a root-mean-square calculation to derive an average (mean) power level over a period of time that is called an RMS window. Smaller window sizes make the measurement

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more responsive to short-duration events and low-level peaks, while larger sizes apply more smoothing but add latency.

In the VU's heyday, mechanical VU meters were useful because of how difficult it was to create a more accurate averaging meter. Modern software meters are rarely true VU meters, even when they are marked as such. RMS meters provide a more meaningful indication of average level.

Peeking at Peaks

Signal peaks can be much higher than average levels, and they are faster and shorter in duration. The peak program meter (PPM) was created to show signal peaks. The PPM-meter standard (European Broadcast Union technical document 3205-E) mandates an integration time of about 10 ms, which is fast enough to catch most peaks, but just slow enough to ignore spurious artifacts. Digital peak meters, which are the most common in DAWs, have no integration time—that is, they show the instantaneous peak value of the signal. While that method is truthful and can help to avoid clipping problems, it can lead you to focus disproportionately on peak values rather than average values, which give a better indication of what you are hearing. It is unfortunate that RMS meters aren't as prevalent as peak meters in DAWs and audio editors.

Loudness is a perceptual attribute that is almost never directly measured.

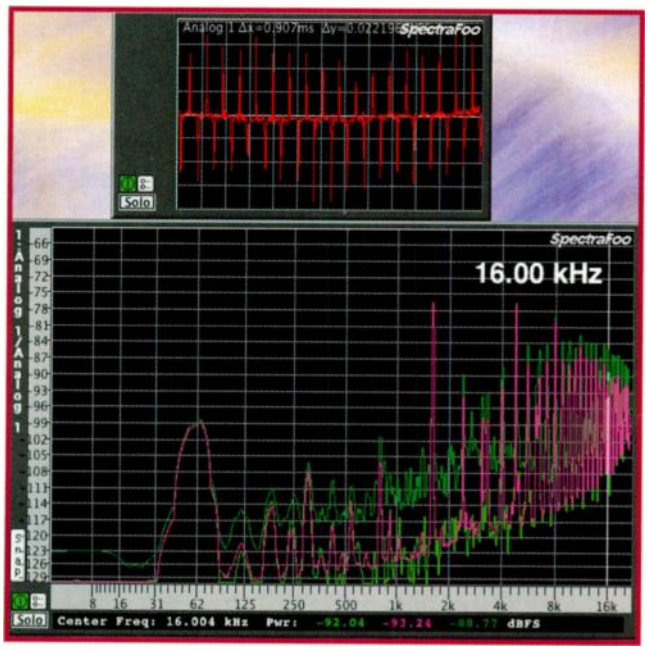

FIG. 4: Metric Halo SpectraFoo's FFT spectrum analyzer (called the Spectragraph) and its oscilloscope are both highly configurable.

Hold functions highlight the hottest points in a signal by retaining the last high point that the level meter hit for a configurable amount of time (the hold time). Assuming that a higher peak has not already come along and reset the indicator, it will then fall to the current peak and remain there. Many engineers rely on the peak-hold indicator even more than they rely on the peak meter to monitor the maximum levels in the signal.

The Historical Record

Level history can be shown in several ways. A peak and/or RMS power-level history gives a continuous record of how the level evolves. That information is useful for, among other things, checking level in different parts of a song to make sure that the level of one verse or chorus is not too much higher or lower than that of other verses and choruses.

Level histograms give a different perspective on history. Histograms show frequency distribution, and so a level histogram indicates how often a given level occurs over time (see **Fig. 3**). When setting dynamics processors such as compressors and gain maximizers, a level histogram helps by letting you see where most of a song's energy falls.

The time-domain waveform displays found in DAWs and audio editors are level histories, too. Oscilloscopes, on the other hand, provide real-time waveform monitoring (see **Fig. 4**).

Implementing real-time analysis tools for multichannel surround can be a daunting task in several respects. As a result, it's rare to find surround monitoring other than within a DAW host. Programmable Analysis Software's (PAS) shareware Surround Meter is one of the few such tools that shows up in a Google search.

Spectral Analysis

After level, the most common analysis task is spectral analysis. Spectral-analysis tools show the amplitude of each frequency component of a signal. The analysis parameters are key to determining frequency resolution and accuracy across the spectrum. FFT spectrum analyzers, the most common type, are *x-y* displays that show frequency on the *x*-axis and amplitude on the *y*-axis (see Fig. 4). FFTs can be read intuitively: if you have a lot of energy at or around one frequency, that fact is self-evident from the higher readout shown at that frequency.

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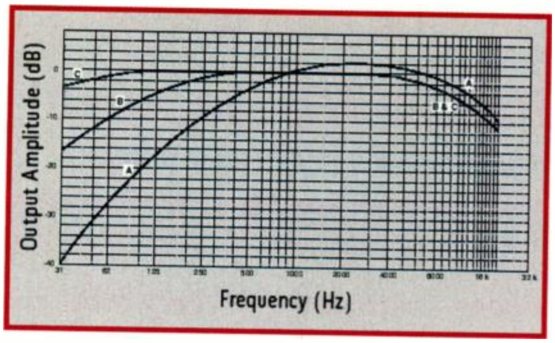

FIG. 5: This graphic shows the A-, B-, and C-weighting curves, derived from Fletcher-Munson equal-loudness curves.

That is not to say, however, that spectrum analyzers are only narrowly useful. You can look at both RMS and peak spectral analyses with them and use hold functions in the way that they are used with level metering. Using hold functions can be very helpful when you are trying to pinpoint problem frequencies that aren't easily identified by ear. Since spectrum analyzers usually give a readout of the frequency and level corresponding to the current position of the cursor, you can easily home in on where a problem lies by looking at the hold display and moving the cursor where there are anomalies.

Weighty Matters

Many spectrum analysis tools offer A-, B-, and C-weighting curves, which make the analyzer read more in the way that sound is perceived. Human hearing response is not linear across frequency, so when a spectrum analyzer shows equal levels of high and low frequencies in a signal, it is likely to sound as though there is more treble than bass. Worse, the frequency response of this filtering effect in human perception changes with level.

Weighting curves apply filters to the signal that is routed to the analyzer in order to bring the readings more in line with how sound is perceived. A-weighting initially approximated the 40 dB Fletcher-Munson equal-loudness curve; B-weighting (rarely used), the 70 dB curve; and C-weighting, the 100 dB curve. In that method, you used the appropriate curve for the source level (see **Fig. 5**). That idea mutated, though, and there was a movement to standardize level measurement around A-weighting only. Today, weighting curves are often chosen for their appropriateness to the application rather than to the listening level. For instance, A-weighting has long been used in outdoor measurements of ambient sound, in which people easily tune out continuous, low-frequency background sounds.

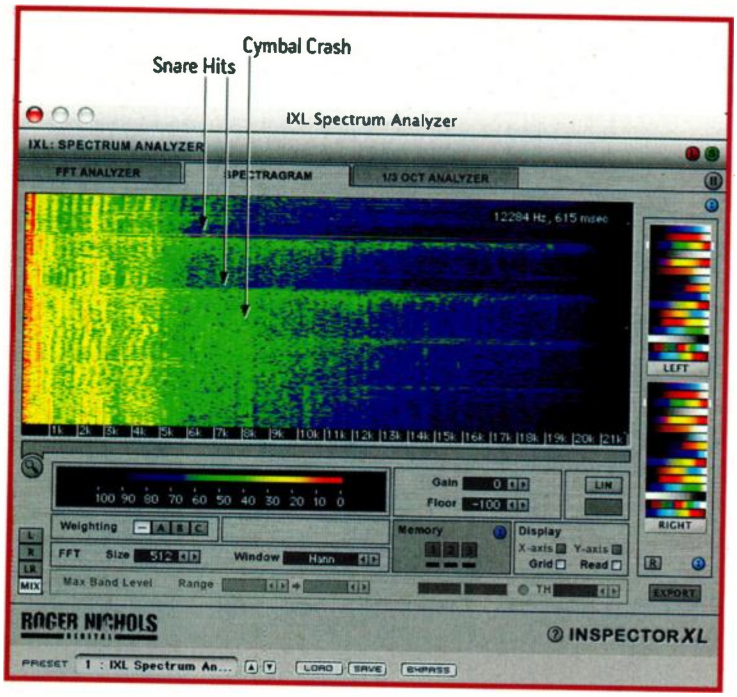

FIG. 6: In RND Inspector XL's spectrogram, note the visibility of cymbal crashes and beats from snare hits.

Third-Octave Analysis

Third-octave analyzers show the spectrum broken down into ISO third-octave bands, a display familiar to those who have used hardware third-octave RTAs (real-time analyzers). Third-octave analyzers are useful for getting a feel for the overall shape of the spectrum. Third-octave bands, however, are not fine enough to pinpoint many problems, and their center frequencies do not closely relate to the harmonic relationships that dominate musical and acoustical signals.

There is an important distinction between FFT spectrum analyzers and third-octave equalizers. FFT analysis produces a linearly spaced response; that is, it breaks the spectrum into bands with spacing that has a fixed number of hertz. White noise, which has equal energy per frequency, shows a flat response when viewed with an FFT analyzer.

In contrast, third-octave analysis produces a logarithmically spaced representation, based on a division of an octave. Pink noise, which has equal energy per octave, shows a flat response when viewed with a third-octave analyzer. White noise shows much more energy in the higher octaves than the lower ones, looking like a low-frequency rolloff.

The Colorful Spectrogram

A spectrogram (sometimes called "sonogram") shows a spectral history, a continuous 3-axis record of FFTs performed on an incoming signal. The spectrogram shows time along one axis and frequency along the second axis and uses color (the third axis) to show level.

Spectrograms are used heavily in speech research but are useful also for studio work, and they're easy to read once you are accustomed to them. For example, it's not hard to see where the beat is in a typical pop song: sharp, regularly spaced lines along the spectrum

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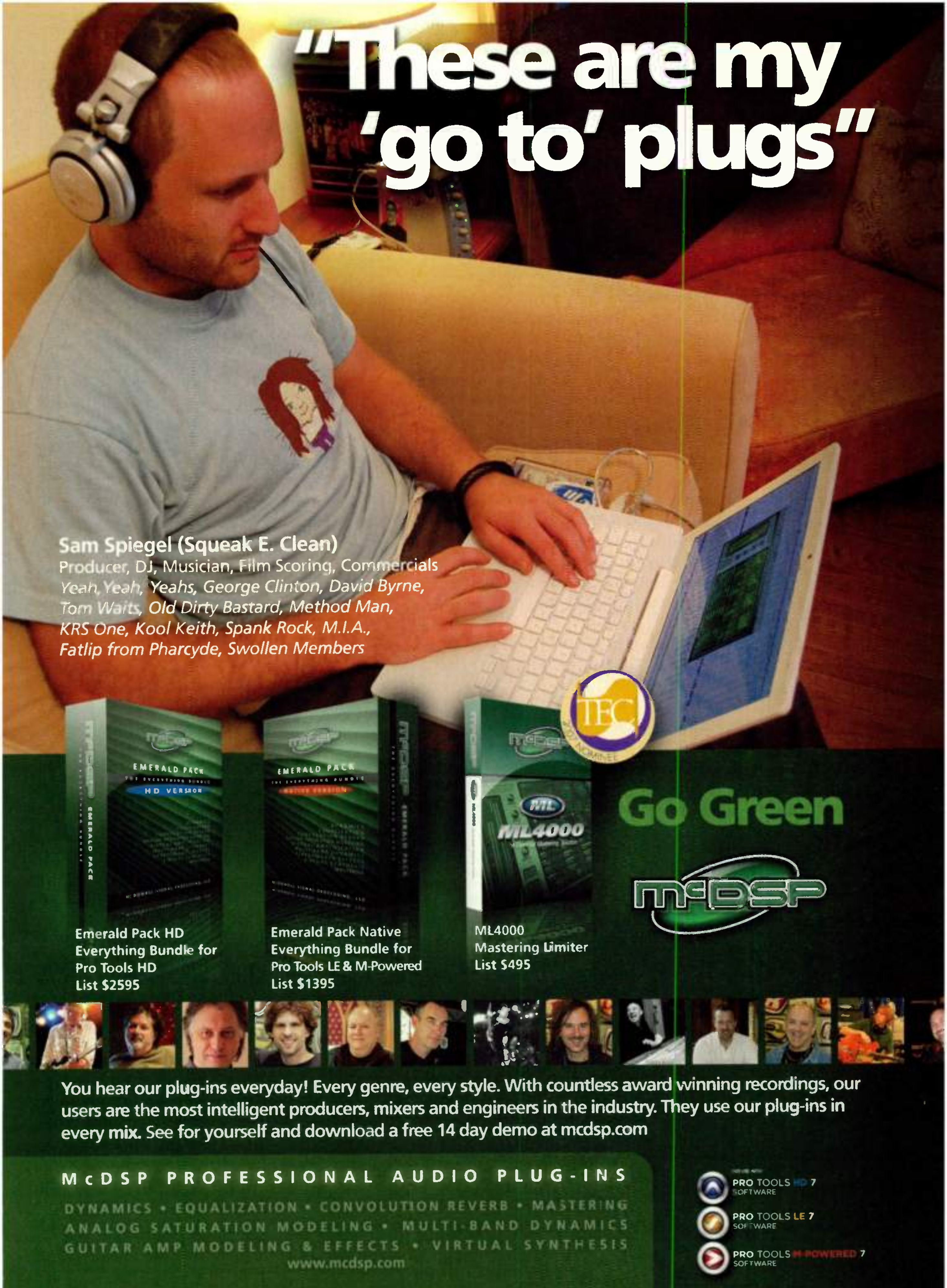

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indicate transients that are probably the snare, and cymbal crashes can be seen by the smear in the high frequencies that follow some snare hits. Other transient events, such as a cough or a door-close during a live recording, can also be spotted (see **Fig. 6**). Spectrograms are helpful in comparing the spectra of different songs. Note that the larger the FFT size, the more history is shown in a spectrogram.

Phase and Stereo-Image Meters

Phase and stereo-image meters illustrate the relative time relationships between the left and right channels of the stereo signal. The simplest of these tools is the stereo-balance meter, a horizontal strip showing power distribution between the two channels. The stereo-balance meter can be useful for balancing stereo tracks of sections, such as background vocals and strings, and for checking stereo-miking techniques.

Phase monitoring comes in several forms, the most familiar being the Lissajous display—a simple *x-y* display in which each axis shows the instantaneous level of one channel. If the display shows a line (or, more often, in practice, a narrow oval) pointing from lower left to upper right, the material in both channels is very much in phase, meaning that the signal should be highly mono compatible. If the display shows a straight line pointing from upper left to lower right, then that means there are identical signals in each channel, but with opposite polarities, which usually is not good. A somewhat fatter oval is the most common, and a circle would indicate that you have as much out-of-phase material as you would want in a mix.

A vector scope works similarly to a Lissajous, except that the display has been rotated 45 degrees counterclockwise, so that in-phase behavior is shown by a line going straight up and down, and out-of-phase material is shown by a horizontal line.

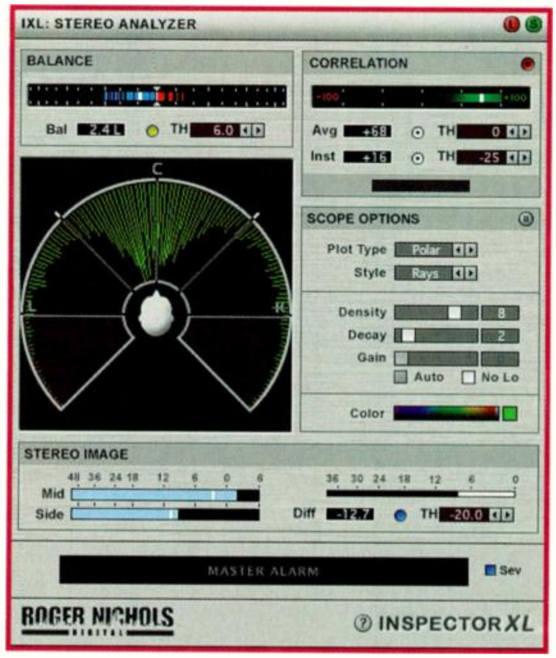

FIG. 7: This phase scope shows a polar plot of amplitude versus perceived location. This material doesn't have much stereo image; note the 13 dB difference between mid and side signals.

Can You Correlate?

Correlation meters measure the similarity between two signals (see **Fig. 7**). A fully correlated signal is the same in each channel—that is, the signal is mono. But *correlation* is not the same as *phase:* if two channels are extremely dissimilar, they may have a lot of out-of-phase information, or they may have a number of hard-panned mono instruments. However, uncorrelated material is usually perceived as stereo material, and highly correlated material as

SOME METERING AND ANALYSIS TOOLS

Many metering and analysis tools are available. Below is a list of the major ones, along with their Web addresses, which will lead you to more information about them.

Audiofile Engineering Spectre www.audiofile-engineering.com/spectre.php

BIAS Reveal (Master Perfection Suite) www.bias-inc.com/products/masterPerfectionSuite

Blue Cat Audio Analysis Pack www.bluecataudio.com/Products/Category_Analysis

iZotope Ozone www.izotope.com/products/audio/ozone

Metric Halo SpectraFoo www.mhlabs.com/metric_halo/products/foo

Programmable Analysis Software (PAS) Surround Meter (shareware) www.audio-software.com

Roger Nichols Digital Inspector XL and Inspector (freeware) www.rogernicholsdigital.com/inspectorXL.html, www.rogernicholsdigital.com/inspector.html

Sony Creative Software Sound Forge www.sonycreativesoftware.com/products/product.asp?pid=431

Steinberg WaveLab www.steinberg.net/128_1.html

SuperMegaUltraGroovy FuzzMeasure Pro www.supermegaultragroovy.com

Troodon Technologies TrooTrace www.troodontechnologies.com/products.htm

Waves PAZ Analyzer www.waves.com/Content.aspx?id=233

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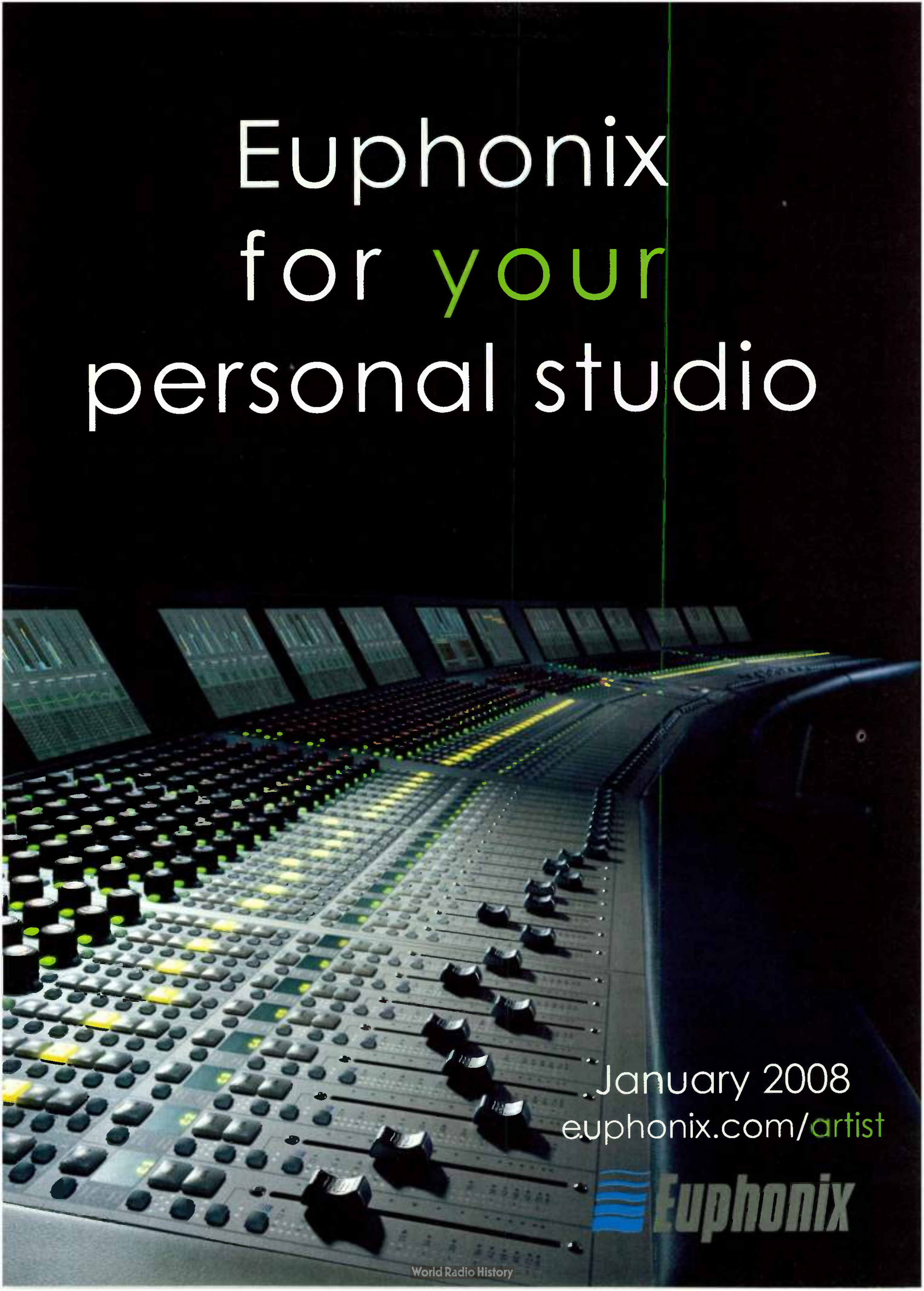

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mono, so a correlation meter can give some idea of the overall width of the signal. It is important to look at trends in the meter more than its instantaneous behavior, because moments with a lot of uncorrelated material between channels are common.

Using mid-side meters is another way of contrasting mono and stereo information in a signal. A mid-side meter typically has two displays: one showing material that appears equally in both channels (mid), the other showing material that is different between them (side). A healthy mix generally has near-equal amounts of both.

Interchannel phase can also be displayed on a polar plot, but two polar-phase plots might represent very different information. For instance, SpectraFoo's Phase Torch shows a vector whose length represents frequency and whose angle represents phase angle. Inspector XL's Phase Scope, on the other hand, plots amplitude as the distance from the circumference of the scope and perceived direction of the sound as the angle.

Transfer Functions

A transfer function is generated with a differential FFT analyzer: that is, FFTs are computed for two signals, and then one FFT is divided by the other, leaving only the difference between them. If one signal is the input to a system and the other its output, the display shows how the signal was altered going through the system. Put a vocal track into one input and an EQ'd version of the same vocal on the output, and you will see the actual EQ being applied (regardless of what the EQ display says).

Transfer functions have several powerful properties. First, since they are a comparison, the source material could be anything, including music. Second, it is possible to show phase differences as well as spectral and level differences, and even to calculate coherence, which is a good indicator of signal-to-noise ratio.

Transfer functions can be used for any kind of comparison, but they are most commonly used for acoustical measurements, especially in tuning sound systems. You can also check the performance of pieces of equipment or software and compare the outputs of two microphones, among other things. As powerful as transfer functions are, they are rarely used in studio applications. One reason for that is they are CPU intensive, making them difficult to release in plug-in format. Another is that test applications often call for comparison, but music recording and mixing production deal more with what is happening in real time and, in some cases, the recent past.

> Analysis tools fall into one of two presentation approaches: "now" and "then."

Code Tools and Statistics

There is a class of information in digital recording that relates to the behavior of the digital audio medium rather than of the source signal, and these are code-related tools. Test packages such as SpectraFoo have a more extensive set of code tools than do most production-oriented analysis packages.

Clipping counters let you know the frequency with which digital clipping has occurred. Some analysis tools refer to "overs" rather than "clipping," and Inspector XL distinguishes the two, defining *clipping* as levels exceeding a user-defined clip threshold, and *overs* as signals that exceed full-scale. In many cases, one or two clipped samples won't be audible, so if a clip counter shows only a few clips for an entire song, there may not be reason for concern. On the other hand, having dozens of clips might be a concern. Clip counters can generally be programmed to respond to a specified number of consecutive clips.

Bit scopes are real-time indicators of the instantaneous use of each bit in a digital word. If, for example, there were a plug-in or device that operated at 16-bit resolution but claimed to use 24-bit, it would be obvious on a bit scope, because bits 17 through 24 would never light up to indicate use. Another example: a bit in the middle of the word that never lights up to show that it is used could indicate a problem in a digital audio converter.

Usage Challenges

The strength of analysis tools is that they give visualizations of quantitative data, which makes understanding the data more intuitive. One limitation of analysis tools is that using more than two or three at a time creates visual clutter. Small analysis windows are not very useful, so with four or five of them open at the same time, even a dual-monitor system must devote substantial space to meters. Constantly updating multiple, real-time graphic readouts is also taxing on a CPU.

There is also the human problem of paying attention. With too many dancing displays going at the same time, the analytical focus that was the impetus for using a meter in the first place becomes dissipated, and you can end up glancing from one meter to the next, trying to catch events that require attention. One way around this is to let the program watch the meters for you, alerting you when specified conditions are met. However, the only package I have found that implements alarms is Inspector XL.

Analysis tools give us useful data, but it is only data, not knowledge about the audio. So, in the final analysis, the ears remain the best and most important source of knowledge about whether something sounds good. But data can be seductive, and people sometimes come to rely more on what they think meters are telling them than on what their ears tell them. That becomes a problem if someone is not metering the appropriate

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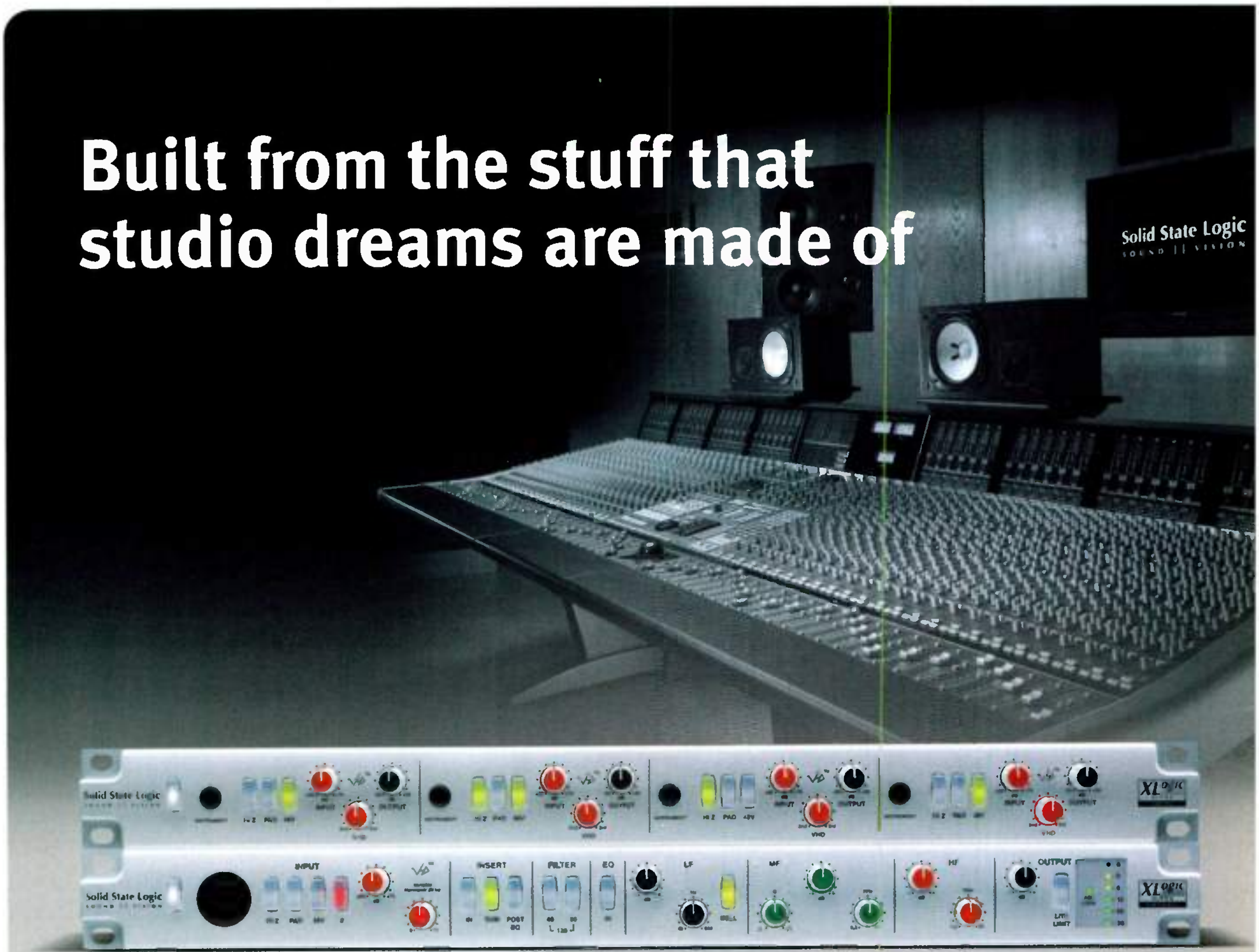

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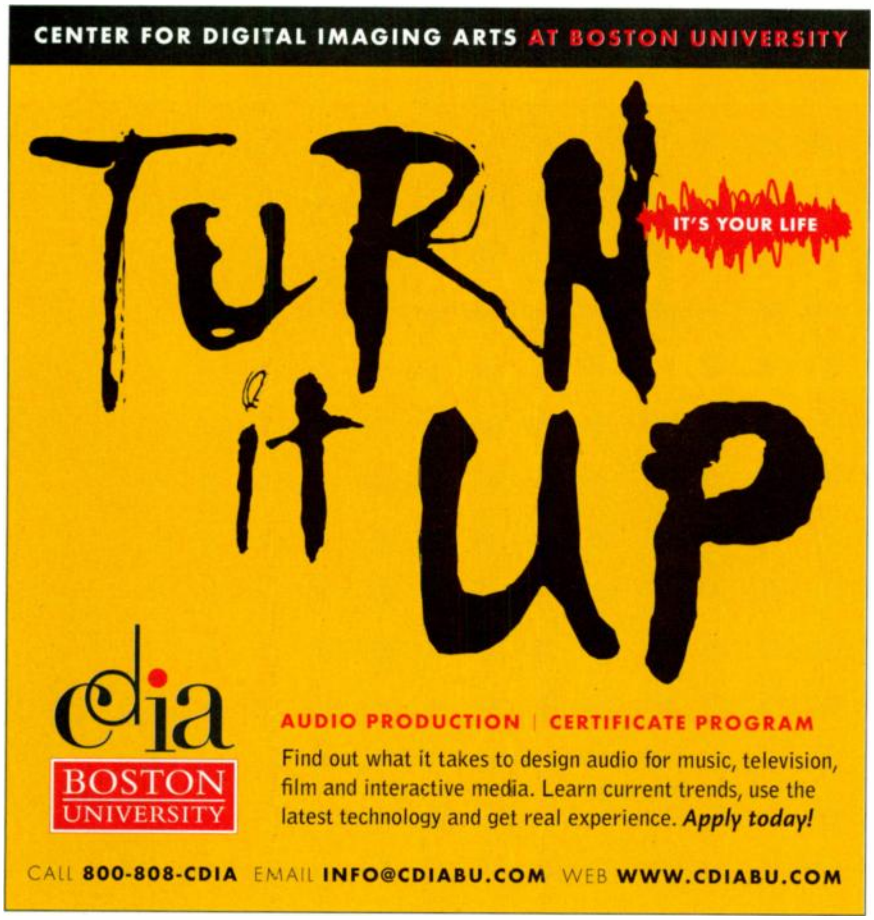

information or if the quantitative data supplied by a meter does not map well to the most closely related perceptual attribute. For instance, level meters mostly give representations that are of the power in a signal. But loudness is a perceptual attribute that does not map directly to signal power as shown on a typical meter. Or the problem may be as simple as not metering the right parameter.

In any event, meters are best treated as supplements to what we hear. If there is a discrepancy between the two, further inquiry may be in order. But it is foolish to assume that the meters must be "right" and that you aren't hearing correctly.

Be a Meter Reader

Meters and analysis tools present various ways of looking at our audio. Getting good results from them depends on three things: understanding the working principles of each tool well enough to know when to use it, being able to read each tool's display, and having the context to interpret what the display is showing and to extrapolate useful knowledge from it.

> Finding surround monitoring other than within a DAW host is rare.

A large and growing number of analysis packages are priced affordably and are available for all of the major operating systems and plug-in formats (see the sidebar "Some Metering and Analysis Tools"). For a creative artist, making informed decisions isn't always necessary, but for an engineer at any level, making informed decisions can help lead to good results. **EM**

Larry the O is a musician, producer, engineer, sound designer, writer, and master soup stirrer in the San Francisco Bay Area. Special thanks to Bob Katz, Joe and B. J. Buchalter of Metric Halo, and Roger Nichols Digital. Tips of the hat to BIAS, Steinberg, iZotope, and Audiofile Engineering.

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INSTRUMENTS

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The 5.1 Mix

By Brian Smithers

Getting creative with surround sound.

Surround production is not as mysterious as you might think. Everything you know about stereo (or mono) still holds true, but with more possibilities. There are new places to localize sounds, new ways to create space for an element, and new corners to paint yourself into. In this month's column, I'll look at both sides of the 5.1 coin (see "Step-by-Step Instructions" on p. 68).

Off-Center

Turn the center channel to your advantage by using it for sounds that you want to stand out or be tightly localized. A part that comes primarily or exclusively from the center channel is essentially mono: its timbre and position are the same from any seat in the house. Beware, however, that if you place your lead vocal only in the center, a listener could mute the center channel and create instant karaoke.

Phantom center is not dead, though. In film, sounds that occur center stage behind the actors exist only in phantom center—not in the center channel. In Digidesign Pro Tools 7, a control called Center Percentage moves center-panned sounds between only the center channel at one extreme and the left and right channels at the other. You can use that to achieve any blend of discrete and phantom center. In Cakewalk Sonar 6, phantom center in a surround project requires assigning the track output to a stereo bus and then assigning that bus to the surround output.

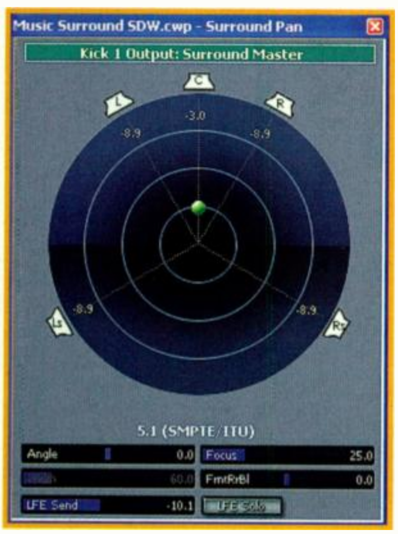

FIG. 1: The LFE Send parameter copies the full-range signal to the LFE channel, which should then be lowpass filtered.

Your Other Left

Theoretically, you can create a stereo field between any pair of channels in a surround setup. In practice, having ears on the sides of your head limits (but does not eliminate) your ability to localize between front-to-back or diagonal pairs. That still leaves plenty of room for experimentation with L/C, C/R, and Ls/Rs stereo, not to mention good old L/R.

Web Clip 1 demonstrates this technique, spreading a drum kit across the L/C stereo field, bass and keyboards across C/R, and strings across Ls/Rs. This is a natural-sounding way to take advantage of the extra real estate that surround sound affords. It builds on tried-and-true stereo-panning techniques while removing the need for all the stereo kludges invented to deal with a crowded soundstage.

On the Down Low

The LFE channel should be used sparingly, because in too many circumstances—Dolby Digital down-mixing, for example—it is compromised or eliminated. In music production, the LFE is usually used to add thump and punch to kick drums and low bass notes. Keep in mind that bass management in the listener's playback system will take care of shifting the low end of each surround channel to the subwoofer, so you needn't use the LFE for that. (For more on bass management, see "Square One: How Low Can You Go?" in the February 2006 issue of EM, available at www.emusician.com.)

Sonar's surround panner includes an LFE Send parameter (see **Fig. 1**), which copies the signal (the way that any send does) and sends it to the LFE channel. Note that this is a full-range copy, whereas the LFE is designed for only the bottom couple of octaves. If you were to connect your LFE output directly to a subwoofer, it would try to play back the entire signal, with bad results. Although some surround sound monitor controllers filter the LFE, you should do it yourself to be sure that you bounce a filtered LFE.

Insert a Sonitus:fx EQ on the main surround output bus. Because it's not a surround effect, it will load using SurroundBridge. Unlink the LFE processor and bypass the rest. On the LFE EQ, disable four bands and set the other two to Lowpass, with a 90 Hz cutoff and a 1.5 Q. The serial filters increase the effective slope without creating a resonant peak. I set the cutoff at 90 Hz, making the 3 dB down point approximately 80 Hz, which is more or less the standard.

Check This

Because most listeners will hear your mix on a bass-managed system, it's important that you check your mix with bass management to listen for any phasing problems it may introduce. Sonar makes this simple, with a checkbox in the Surround tab of Project Options.

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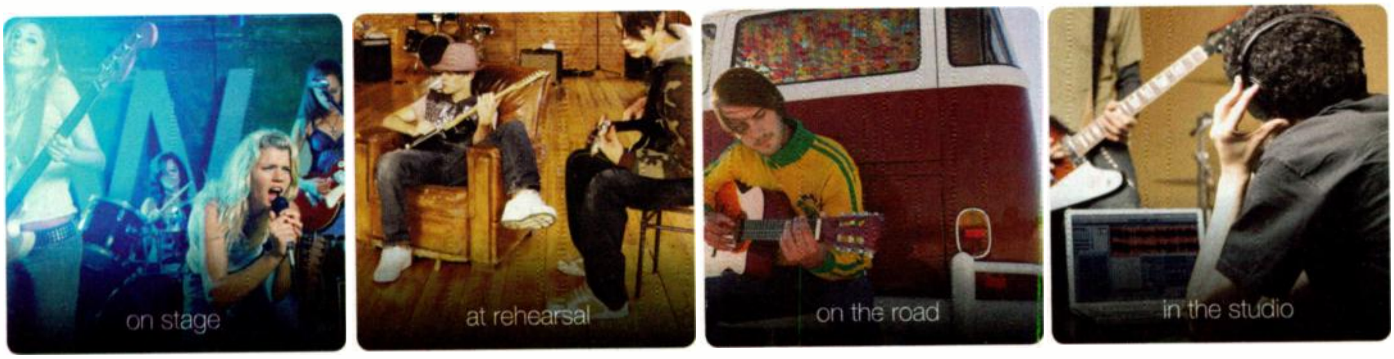
Professional recording — straight to your iPod



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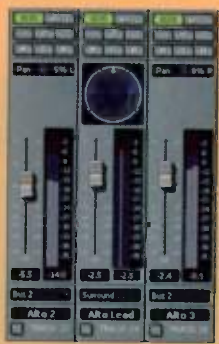
STEP-BY-STEP INSTRUCTIONS

1

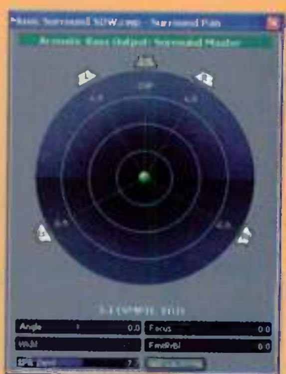


Use discrete center-channel signals to emphasize important parts and L/R phantom images for secondary parts.

2

Use L/C, C/R, and Ls/Rs stereo fields (shown) to set the stage.

3



Use LFE sparingly. Your mix should be able to survive without it.

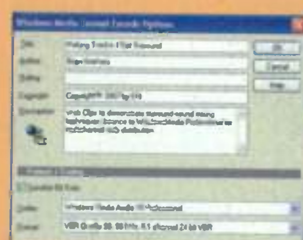
4

Lowpass-filter the LFE channel steeply; the signal should be 3 dB down at 80 Hz.

5

Check your mix with bass management.

6



Know the proper delivery format, such as Windows Media Professional for Web delivery or multimono PCM to be encoded to Dolby Digital.

Check out **Web Clip** 2 for instructions and an example for doing the same thing in Pro Tools.

Now all that's left is to deliver your mix. Know which surround sound format your client needs. For a DVD, which requires Dolby Digital and supports DTS, bounce multimono (not interleaved multichannel) Broadcast Wave files and bring them into an encoder such as Apple's QuickTime Pro or one of Minnetonka's SurCode products. For Web distribution, Sonar can export directly to Windows Media Professional. EM

EM WEB CLIPS

Brian Smithers is course director of audio workstations at Full Sail Real World Education and the author of Mixing in Pro Tools: Skill Pack *(Thomson Learning, 2006).*

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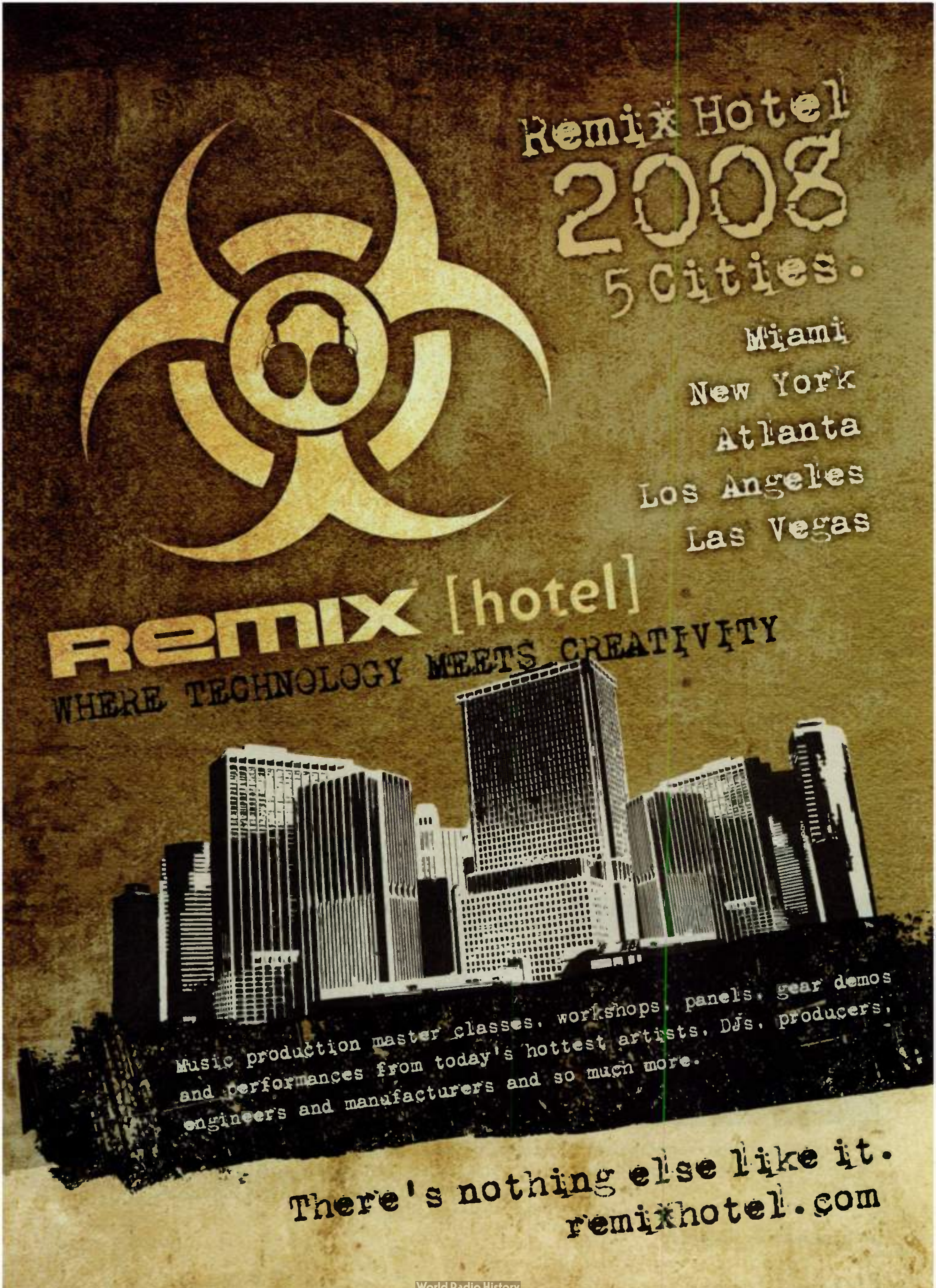

World Radio History

Imaginative Processing

By Brian Smithers

Make the most of multimono surround plug-ins.

Once you start creating in surround, working in stereo feels like painting with half your palette missing. With an immersive sound field, you have many more ways to craft a sound or to frame a musical thought. Cakewalk Sonar Producer's surround capabilities offer plenty of creative options. I'll assume you know the basics—how to create and use a surround bus, how to assign audio channels to outputs, and so forth. Here I'll focus on the sound.

Although my examples use Sonar Producer 6, you can accomplish the same thing in the other major digital audio workstations that support surround mixing. (Digidesign Pro Tools LE is a notable exception; surround is an HD-only feature.) Each DAW has its own implementation and terminology, but the basic principles are universal.

Multiple Choice

Effects in surround can be *multichannel,* in which case they work across all channels in a unified way, or *multimono,* which is like having a separate mono processor on each channel. Although the multichannel method might seem more appropriate, especially for natural-sounding effects, each approach has its place.

I like to put a short multimono delay on lead instrumental or vocal sounds to give them a sense of space before adding reverb. In **Web Clip 1,** you'll hear short single-tap delays (10 to 15 ms) on the left and right channels, slightly longer delays (20 to 30 ms) on Ls and Rs, and no delay on center. In Sonar, you use SurroundBridge for this (see **Fig. 1**), which makes any mono or stereo effect available as linked processors on a surround bus. When the effect is bypassed, the center-panned sound just sits there in the middle of the room, but then the delays move it out of the center by simulating reflections from the walls.

FIG. 1: Sonar's SurroundBridge makes any mono or stereo plug-in available as a selectively linked set of processors on a surround bus.

The sound appears to be everywhere without being anywhere in particular, because the delays are decorrelated (no two delay times are the same). To allow this, go to the SurroundBridge tab and uncheck Controls Linked To Groups for all channels.

Once the sound sits in the surround field the way I like, I can apply reverb. I often prefer multichannel reverbs because they behave like a real space, allowing the sound to bounce from channel to channel as nature intended. Sometimes, however, decorrelating a multimono reverb is good for distancing the reverb from the dry signal.

Near or Far

In stereo, you control proximity of a sound by using EQ and time-based effects. In surround, it's possible to move a signal from the perimeter of the room to the center merely by panning. In surround parlance, this is called *divergence,* although the use of that term is not as tightly standardized as one would hope. High divergence values move a sound closer to the speakers, whereas lower values do the opposite and make the sound converge in the center of the room, right in the listener's lap. Divergence allows the signal from one speaker to bleed into adjacent speakers. In Sonar, this is called Focus.

In **Web Clip 2,** the sound walks around the perimeter of the surround field, first at maximum distance (Focus = 100), then in a smaller circle (Focus = 60), and finally in a small circle around the listener (Focus = 30). You can lock the radius of the circle by holding Alt as you drag the cursor.

To create a spiraling-in effect, automate several rotations at the perimeter, then go back and automate the Focus control from 100 down to 0. For a more even effect, draw the envelopes. Create an Angle envelope, and using the Saw envelope shape, drag from the top of the track to the bottom to create a clockwise spin (or vice versa), then drag to the right for the duration of the desired effect. The sound will pan one complete circle for every cycle of the Snap to Grid value. Halve the frequency by holding the Ctrl key as you drag; double it by holding Alt. To spiral in, create a Focus envelope, add a node at the top of the track for the beginning and another at the bottom for the end of the effect, and choose a linear curve from the envelope's right-click menu. **EM**

Brian Smithers is course director of audio workstations at Full Sail Real World Education and the author of Mixing in Pro Tools: Skill Pack *(Thomson Learning, 2006).*

World Radio History

World Radio History

MUSIC BUSINESS INSIDER

What percentage of the popular recorded-music base that's available in the United States do you have cataloged at this point?

It depends a little bit on how you define "popular." We have everything that's ever been on *Billboard* since it started back in the '50s. But that's not that much music. In sheer number terms, in the North American catalog for a really complete collection, it's probably about 4 million songs. And we have about 500,000, which is about 10 percent.

Do you have classical music in your database?

We don't have classical, but we're going to be launching that soon.

So listeners go to your Web site, sign up, type in a song or artist, and the site starts playing songs?

It will start playing a radio station (see **Fig. 2**). It's really like a radio station. Currently, [the site's] mostly being used on a computer, but it's now also available on a [mobile] phone. So the idea for the long run is for [it] to be a radio.

In the mobile phone space, which carriers are you on?

So far, just Sprint. But there are more to come.

What's the business model? Do you have advertising on the site?

It's advertising supported.

What about the Internet-radio royalty issue that has been in the news? How is that affecting your business?

It's very directly affecting us. The new rates, if they go into effect, will put us out of business.

How much more per song is it than it used to be?

It's a tripling of our rates. That's pretty disastrous. And the same goes for every Internet radio company. A few months ago, when this decision came down, everybody in the Webcasting community, including us, thought it was Armageddon. And it looked like we were all going to go out of business. But what happened was that there was this massive public uprising. The Webcasters appealed to their listeners and got well over a million people to call their congressperson. It was such an overwhelming public response that Congress intervened and basically said, "You've got to negotiate a better number."

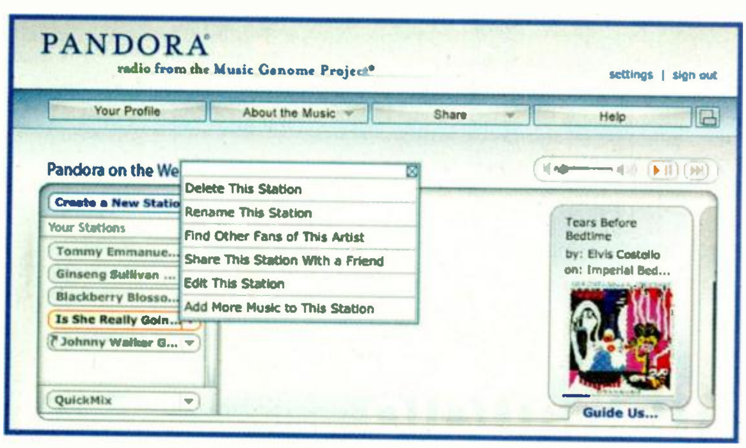

FIG. 2: The Pandora user interface lets you set up "stations" based on artists or songs that you like.

Are you confident that something you can live with will get worked out?

Let's say I'm cautiously optimistic.

When do you expect a decision to come down?

It's really hard to say. The timing of it depends on how the RIAA responds to it.

So the record industry feels like it's not getting enough compensation for the use of its music?

Right.

But they are getting compensation, just at a lower level than broadcast radio?

No. You pay two forms of licensing—a publishing and a performance royalty. There are three forms of radio: terrestrial, satellite, and Internet. Each of us pays the same publishing rate, about 3 or 4 percent of revenue. But for the performance fee, we pay ten times the amount that satellite pays, and broadcast radio doesn't pay it at all. It's really, really uneven—unequal. It's discriminatory.

Do you have a lot of indie musicians along with established stars in your database?

In our collection of half a million songs are 40,000 artists. Seventy percent of our artists are not affiliated with a major record label.

How does one go about getting music submitted?

There are instructions on our Web site. Artists can just mail their stuff in. And there's no prerequisite for getting included. You don't need to have had any kind of commercial success at all—you could be a hobbyist. The only criteria is that it's got to be good.

So there is a gatekeeper that they must get past?

That's the one point in the process where we actually have an editorial voice, when we decide who we think is good. And that's what the trained musicians are partly for, since they play that role. The real power of this is that if you're an unknown, this is the only radio where you'll get played alongside artists you sound just like. It's a great promotional tool, and that's why I started this. I used to be a musician myself; I used to play in rock bands. I did the whole "live-in-a-van" thing. It's why I'm doing it. EM

Mike Levine is an EM senior editor.

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World Radio History

The Sculptor's Tool

By Brian Smithers

How to use EQ to shape your music.

At the most basic level, a musician controls three elements: pitch, time, and timbre. An engineer's job is to help (or sometimes second-guess) musicians in this quest. Although minute control over pitch and time has been within our grasp only in recent years, controlling timbre is a well-established part of engineering. Each time you choose a microphone or tweak a preamp, you're affecting the recorded timbre. The tool we most often think of for timbral processing, however, is the equalizer.

An equalizer, or EQ, is a hardware or software signal processor that allows you to manipulate a sound's frequency spectrum. If you've ever used the tone controls on your home receiver or made a smiley face out of the sliders on the front of your car stereo, you know what EQ does—just enough to get you in trouble. In this column, we'll look at the various components of an equalizer, explore how they work, and describe how they can be applied to improve your sound.

EQ Pluribus Unum

To paraphrase the 18th-century French mathematician Joseph Fourier, any complex sound is made up of a number of different frequency components. Because this is true, we can change a sound's basic timbre by manipulating those components independently. That is the function of an equalizer: to amplify or decrease portions of the frequency spectrum. This is accomplished with filters (for a discussion of filter basics, see "Square One: Just Passing Through" in the February 2005 issue of EM, available online at www.emusician.com). As you may know, a filter is also an essential part of a synthesizer. Although a synthesizer filter is designed to impart a unique character to the sound, and an EQ filter is more often designed to do its job inconspicuously, they are in fact the same thing.

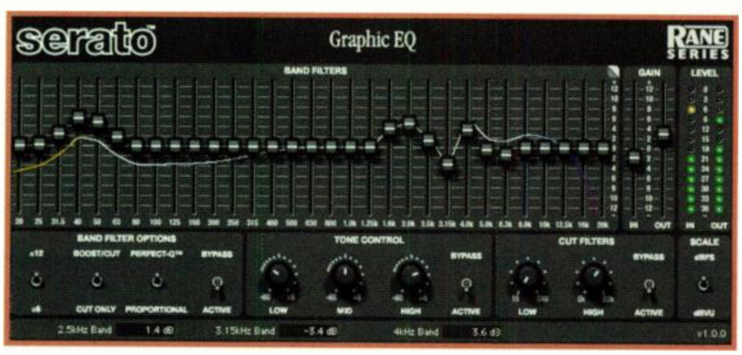

FIG. 1: A graphic equalizer divides the audio spectrum into 30 or 31 bands and provides clear, concise visual feedback of what has been cut or boosted.

Like synthesizer filters, EQ filters come in various types. An equalizer can combine several regions or *bands* of filters within a single processor, and most do. That smiley face on your car stereo is really seven or more bandpass filters with fixed bandwidths and center frequencies. Because the sliders give obvious visual feedback, this is known as a *graphic* equalizer (see **Fig. 1**). Professional graphic EQs offer 30 or 31 bands, each affecting a range of one-third of an octave. Such devices are standard equipment in sound reinforcement, as they give the engineer easy and direct control over any frequencies that start to feed back during performance. A skilled front-of-house engineer can listen to a room and visualize the graphic EQ's curve before touching a control.

Although graphic EQs are also found in studios, a more common design includes four to seven bands of different types. Software EQs often allow the user to disable unused bands to conserve resources or to change filter types for each band. Although each band is technically a filter, when an engineer refers simply to "the filters," she's probably talking about the first and last bands, which are often fixed or defaulted as high- and lowpass filters, respectively (see **Fig. 2**). In addition to variable cutoff frequency, filters normally offer variable slope, which refers to the steepness of the filter. The slope could be continuously variable, or it might be adjustable in increments of 6 dB up to 24 or even 32 dB per octave. In the latter case, the term *pole* is typically used for each 6 dB increment—a 24 dB-per-octave slope would be a 4-pole filter.

Filters are most often used simply to restrict the bandwidth of a track. After all, if you're recording piccolo, anything below 500 Hz is by definition not part of the piccolo's sound. Filtering the low end reduces bleed from other instruments, helps mitigate any mechanical noises around the mic, and helps prevent environmental sounds such as air-conditioning and electrical hum from accumulating on multiple tracks. Filtering the high end of a kick drum reduces bleed from the snare and cymbals, keeping the kick track as clean and isolated as possible (see **Web Clip 1**).

Engaging both high- and lowpass filters creates a bandpass filter. Although on some EQs the filters have limited range, many EQs allow the high- and lowpass filters to cover most if not all of the audio range. Such an

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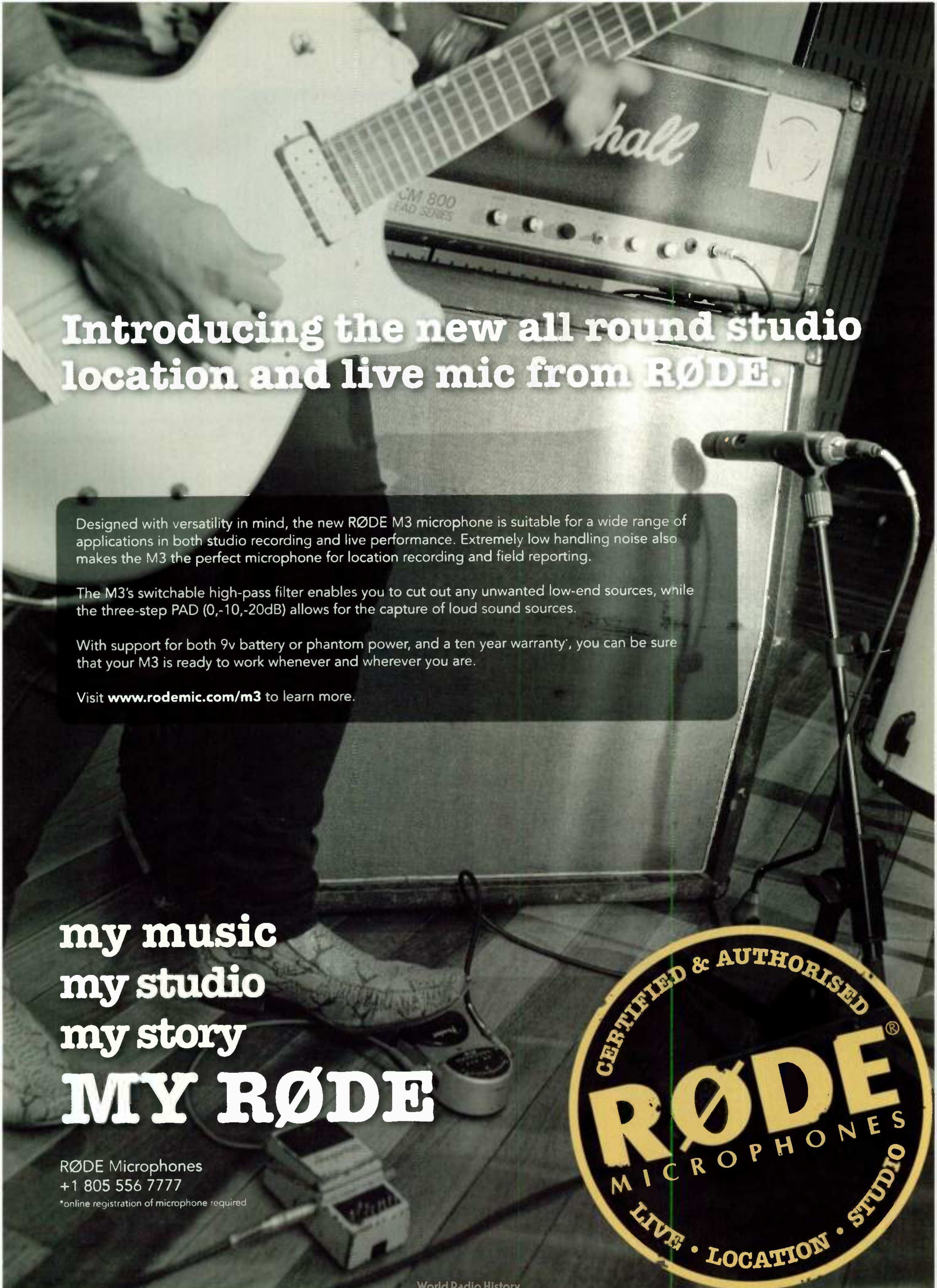

World Radio History

EQ would enable you to place the two filters very close to each other, creating the standard "telephone" filter that's used on so many lead vocals and drum loops (see **Web Clip 2**).

Often the top and bottom bands of an equalizer can be switched from high- and lowpass mode to a shelving mode. A shelving filter boosts everything above (or below) the cutoff frequency by an equal amount—its effect does not increase with frequency (see **Fig. 3**). The treble and bass controls on your home stereo are most likely shelving filters with a fixed corner frequency. On an EQ, shelving filters ordinarily offer a variable corner frequency, but their use is essentially the same—to boost or cut "the bass" or "the treble" of a track or mix. For example, a high shelving filter would be just the thing to add sizzle to a synth pad or string ensemble (see **Web Clip 3**). Shelving filters may offer control over only the corner frequency and amount of boost or attenuation, or they may also allow the user to adjust the slope of the transition band.

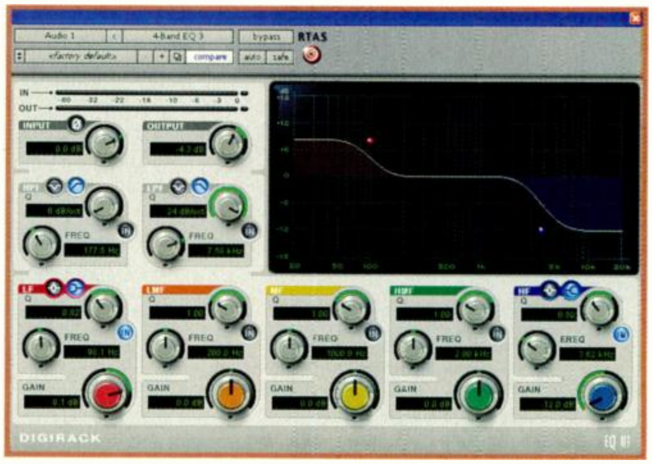

FIG. 3: A shelving filter's effect is not progressive like that of a lowpass or highpass filter. Instead, it boosts all frequencies above (or below) its corner frequency equally.

The remaining bands of equalization are ordinarily made up of bandpass/band-reject filters, which do exactly as their names suggest, affecting a range of frequencies. The affected range is defined by a center frequency and bandwidth. Bandpass filters are used for a great variety of purposes, and some sounds may require the use of several bands. Unlike those found in a synthesizer, however, an EQ's bandpass filters do not block all frequencies outside the band, but rather provide a boost or cut within the band without affecting other frequencies. The alternative name *peak/notch filter* is arguably more precise.

The most flexible bandpass filters are called *parametric*, meaning simply that each of their parameters (frequency, bandwidth, and gain) is independently adjustable. A band whose controls are not fully independent—for instance, one whose bandwidth is linked to its center frequency—may be called *quasi-* or *semiparametric*.

FIG. 2: A multiband equalizer's lowest and highest bands often default to high- and lowpass filters, respectively.

In principle, the use of a bandpass filter is quite simple: boost what you like, cut what you don't. In practice, this takes a skilled ear and refined judgment. A good way to develop your ear is to "sweep" the EQ. Boost a band as much as you can, and narrow its bandwidth as far as you can. If the EQ provides a graphic display, you will have created a very sharp spike. Now slowly raise and lower the band's frequency, listening carefully. At some point, you will hear either an attractive or an unpleasant portion of the track jump out at you. Note that in a host-based DAW, latency may cause you to overshoot the target frequency, so take your time zeroing in.

Once you've found a key frequency band, adjust the amount of boost or cut and the bandwidth to a more appropriate level—no matter how attractive the frequency, you probably don't want the band maxed out. Repeat this for as many bands as needed.

It's common practice to rely on cutting bands when possible, for reasons of gain staging and filter character. When boosting a band, make sure the increased gain does not distort the subsequent circuit. If the EQ provides output meters, pay attention to them. Also listen for the edgy sound of phase distortion, and consider whether a complementary cut in a similar sound would serve your purpose as well. For example, when a kick drum and bass part are getting in each other's way, a wise engineer looks first to trim away part of each to let the other shine through.

The term Q is sometimes substituted for bandwidth or slope. You may remember this as the term for resonance in a synthesizer filter, and in an EQ filter, it means essentially the same thing. However, EQ filters are more typically (though not always) crafted without resonant peaks even at very steep slopes and narrow bandwidths. Remember that a high Q value counterintuitively means a narrow bandwidth or steep slope.

Whatever EQs you have at your disposal, whether software or hardware, no matter the design, these essential principles hold true. Get to know the sound of different band types, slopes, and bandwidths, and tune your ear to pick out the good and bad frequency components. Listen, experiment, and listen some more, and you'll be a master of timbre in no time. EM

EM WEB CLIPS

Brian Smithers is course director of audio workstations at Full Sail Real World Education in Winter Park, Florida.

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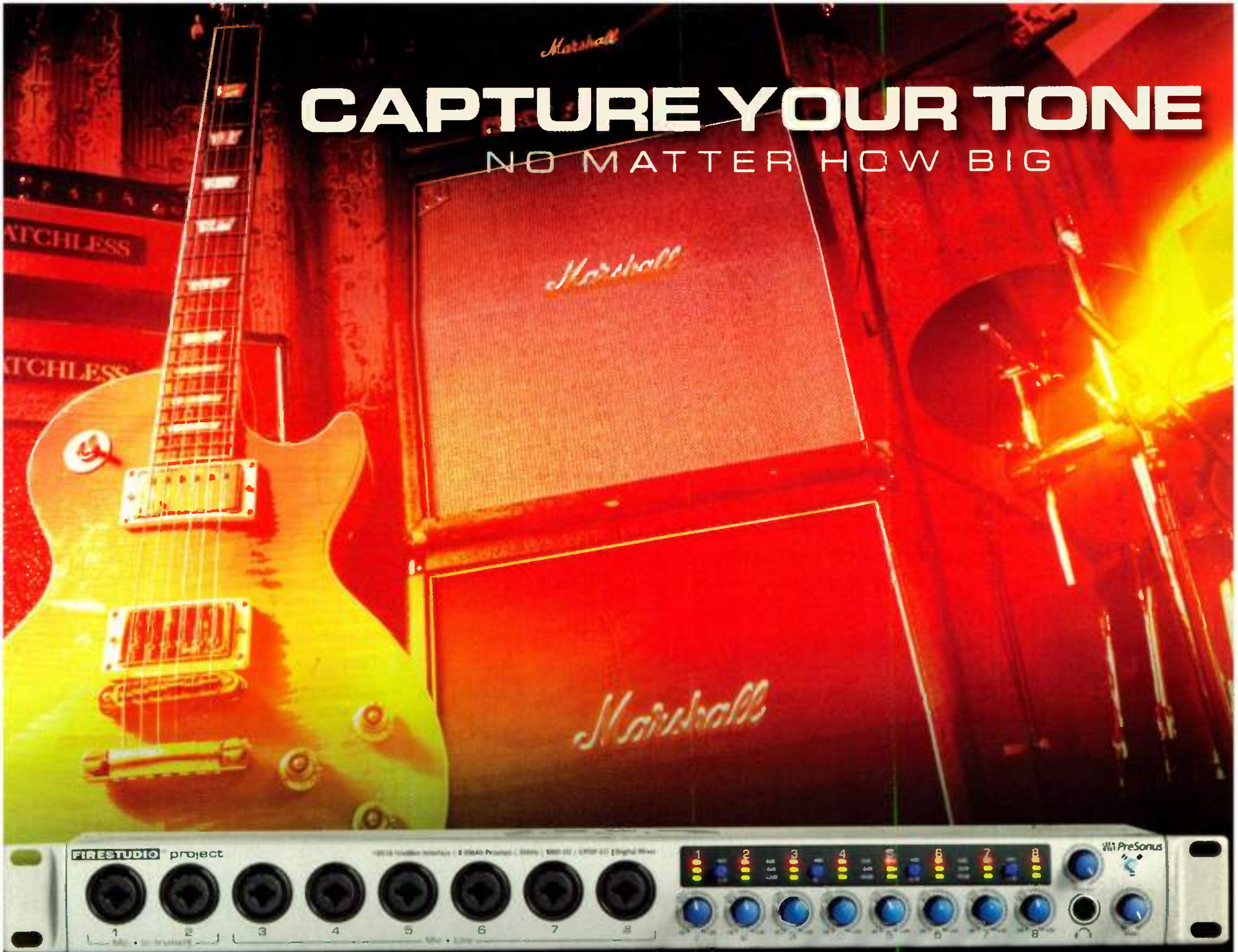

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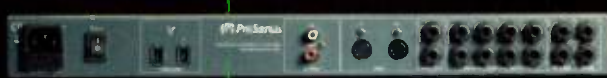

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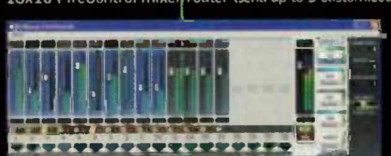

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World Radio History

Q&A: Tim Westergren

By Mike Levine

Pandora maps the musical genome.

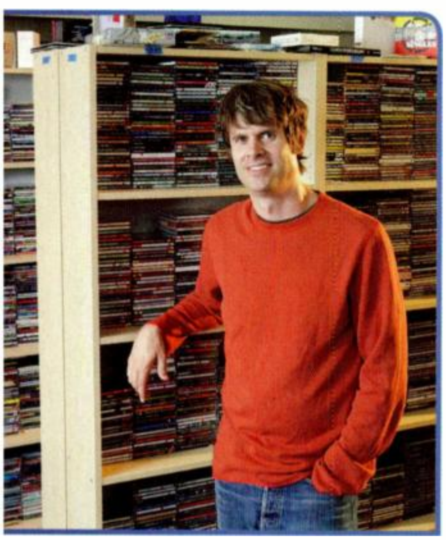
If you haven't yet been to Pandora.com, get ready to be impressed. Go there, join (for free), type in the name of an artist or song, and the site will automatically create a personalized radio "station" for you that plays similar music. The site will play music continuously and allow you to rate the individual songs to further refine your preferences. It does this using the Music Genome Project, a rating system for songs involving a huge number of variables that are entered into a database by trained musicians enlisted to analyze music.

Although Pandora is a consumer Web site, it does offer musicians the chance to get their songs into its system. You can submit your music to the site and, if accepted, it will be categorized alongside all the others. When somebody initiates a "station" with an artist whose music is similar to yours, or with a song similar to one of your songs, your music will be placed in the pool of songs that are recommended and played by Pandora's engine. I recently spoke with Pandora's founder, Tim Westergren, about how it all works (see **Fig. 1**).

Describe the Music Genome Project.

It's an enormous musical taxonomy. Hundreds of thousands of songs have been analyzed by a trained musician for close to 400 musical attributes per song. So it's kind of like musical DNA, and that powers the way Pandora creates its radio stations. When you're a listener, what you do on Pandora is type in the name of an artist or a song that you like, and the Music Genome Project will look [in its database] at that song's sort of musical DNA, and then go look for musical neighbors and start sequencing songs.

FIG. 1: Tim Westergren is the founder of Pandora.



RAFAEL FUCHS, COURTESY PANDORA

How do the songs get classified?

We have a team of about 50 trained musicians. These are folks who all have a very long background in music theory. They spend each day with headphones on, listening to music, and capturing and describing every minute detail of every song. Everything from melody, harmony, rhythm, and form to instrumentation, the sound of the voice, and the vocal harmony—every aspect of the song. They score [the songs] piece by piece, so the voice, as an example, is scored on about 30 attributes, which include things like the use of vibrato, the use of falsetto, the range, and the different timbral elements. They look at all the kind of primary colors in a way that gives that vocalist his or her sound.

All the other instruments are analyzed as well?

Right. We basically have on this taxonomy, this Genome Project, a feel for just about everything you could imagine.

If, for example, they were breaking down the drum part on a funk song, they would be notating the syncopated rhythms in the drum parts?

Sure. The rhythm has a whole set of attributes that describe it—the use of syncopation, the meter, the rhythm, the level of swing, all the different elements. The idea, in a way, is that you could literally just look at the page where these scores are and hear the song in your head.

How much do you take genre into account when you're analyzing a song?

We don't really deal with genre. So for us, a genre is a construction of these many attributes. What makes something alternative rock, what makes something country, classic country, folk country, blues country? We try not to deal with the genre; we try to deal with the elements that determine that, in a way.

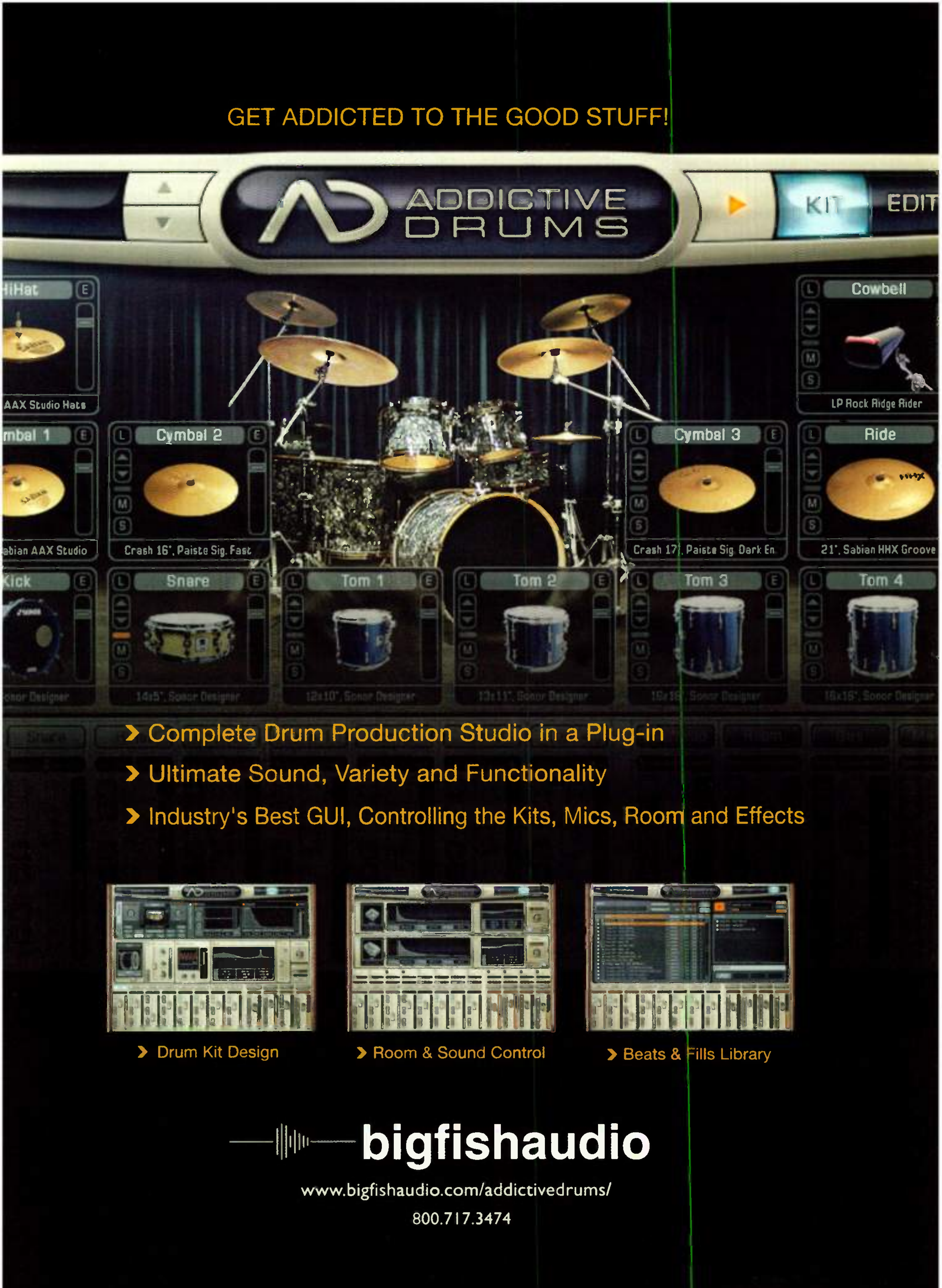
In other words, if you analyze the component parts of a song, you'll know what genre it is based on the results?

As you know, genres change over time. The definitions are very fluid. The nice thing about this kind of methodology is that we can accommodate new genres as they come in. Like, electronica sprouts a new genre every week. And we can handle it because we're not dealing in preset notions of genre.

There are some different attributes in your analysis of jazz, right?

A wider palette of instruments and more detail on the soloing, which is such a key part of jazz.

World Radio History



World Radio History

What percentage of the popular recorded-music base that's available in the United States do you have cataloged at this point?

It depends a little bit on how you define "popular." We have everything that's ever been on *Billboard* since it started back in the '50s. But that's not that much music. In sheer number terms, in the North American catalog for a really complete collection, it's probably about 4 million songs. And we have about 500,000, which is about 10 percent.

Do you have classical music in your database?

We don't have classical, but we're going to be launching that soon.

So listeners go to your Web site, sign up, type in a song or artist, and the site starts playing songs?

It will start playing a radio station (see **Fig. 2**). It's really like a radio station. Currently, [the site's] mostly being used on a computer, but it's now also available on a [mobile] phone. So the idea for the long run is for [it] to be a radio.

In the mobile phone space, which carriers are you on?

So far, just Sprint. But there are more to come.

What's the business model? Do you have advertising on the site?

It's advertising supported.

What about the Internet-radio royalty issue that has been in the news? How is that affecting your business?

It's very directly affecting us. The new rates, if they go into effect, will put us out of business.

How much more per song is it than it used to be?

It's a tripling of our rates. That's pretty disastrous. And the same goes for every Internet radio company. A few months ago, when this decision came down, everybody in the Webcasting community, including us, thought it was Armageddon. And it looked like we were all going to go out of business. But what happened was that there was this massive public uprising. The Webcasters appealed to their listeners and got well over a million people to call their congressperson. It was such an overwhelming public response that Congress intervened and basically said, "You've got to negotiate a better number."

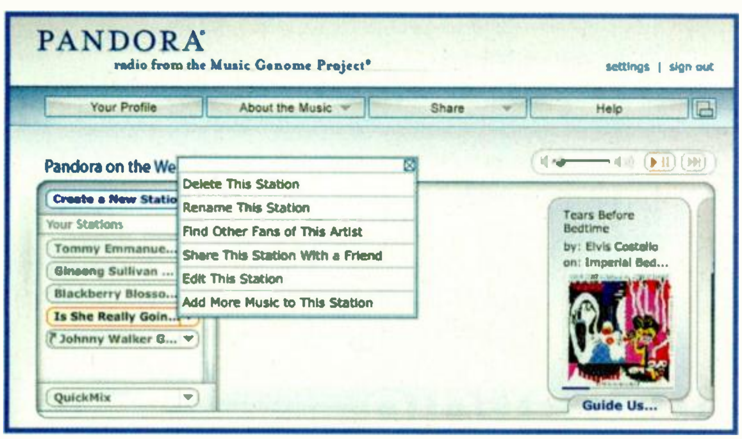


FIG. 2: The Pandora user interface lets you set up "stations" based on artists or songs that you like.

Are you confident that something you can live with will get worked out?

Let's say I'm cautiously optimistic.

When do you expect a decision to come down?

It's really hard to say. The timing of it depends on how the RIAA responds to it.

So the record industry feels like it's not getting enough compensation for the use of its music?

Right.

But they are getting compensation, just at a lower level than broadcast radio?

No. You pay two forms of licensing—a publishing and a performance royalty. There are three forms of radio: terrestrial, satellite, and Internet. Each of us pays the same publishing rate, about 3 or 4 percent of revenue. But for the performance fee, we pay ten times the amount that satellite pays, and broadcast radio doesn't pay it at all. It's really, really uneven—unequal. It's discriminatory.

Do you have a lot of indie musicians along with established stars in your database?

In our collection of half a million songs are 40,000 artists. Seventy percent of our artists are not affiliated with a major record label.

How does one go about getting music submitted?

There are instructions on our Web site. Artists can just mail their stuff in. And there's no prerequisite for getting included. You don't need to have had any kind of commercial success at all—you could be a hobbyist. The only criteria is that it's got to be good.

So there is a gatekeeper that they must get past?

That's the one point in the process where we actually have an editorial voice, when we decide who we think is good. And that's what the trained musicians are partly for, since they play that role. The real power of this is that if you're an unknown, this is the only radio where you'll get played alongside artists you sound just like. It's a great promotional tool, and that's why I started this. I used to be a musician myself; I used to play in rock bands. I did the whole "live-in-a-van" thing. It's why I'm doing it. EM

Mike Levine is an EM senior editor.

World Radio History

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

Jenna Drey – TAXI Member – www.jennadrey.com

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

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

Who Hears Your Music?

I'll bet you've also noticed that no matter how much preparation you've done, it doesn't mean anything if you can't get your music heard by people who can sign on the dotted line.

I found out about TAXI a few years ago, and have kept an eye on it ever since. The longer I watched, the more I became convinced it was the vehicle I needed for my music. When my demos were done, I joined. And guess what – it worked!

A Record Deal With Lots of Zeros!

Seven months after joining, TAXI connected me with a great Indie label that's distributed by Universal. The president of the label heard my song, "Just Like That," and just *like* that, I was offered a record deal, and that song became my first single.

Madonna, Bowie, Jagger, and me!

The icing on the cake? The label hired legendary producer, Nile Rodgers (Madonna, David Bowie, Mick Jagger, and the B-52s) to produce it! All these amazing things happened to me because I saw an ad like this and joined TAXI.

1,200 Chances to Pitch Your Music

It seems like all the serious artists and writers are hooking up with TAXI. Where else could you find more than 1,200 high-level opportunities for your music every year?

You'd hire an accountant to do your taxes. Doesn't it make sense to hire the world's leading independent A&R company to make all the connections you need? Do you have the time to do all the leg work yourself?

It Worked for Me

TAXI doesn't take a percentage of anything, and it will probably cost you a lot less than the last guitar or keyboard you bought. Think of TAXI as the most important piece of gear you'll ever need. It's the one that can get you signed.

If you're a songwriter, artist, or composer who wants to succeed in the music business, then do what I did and make the toll-free call to TAXI right now.

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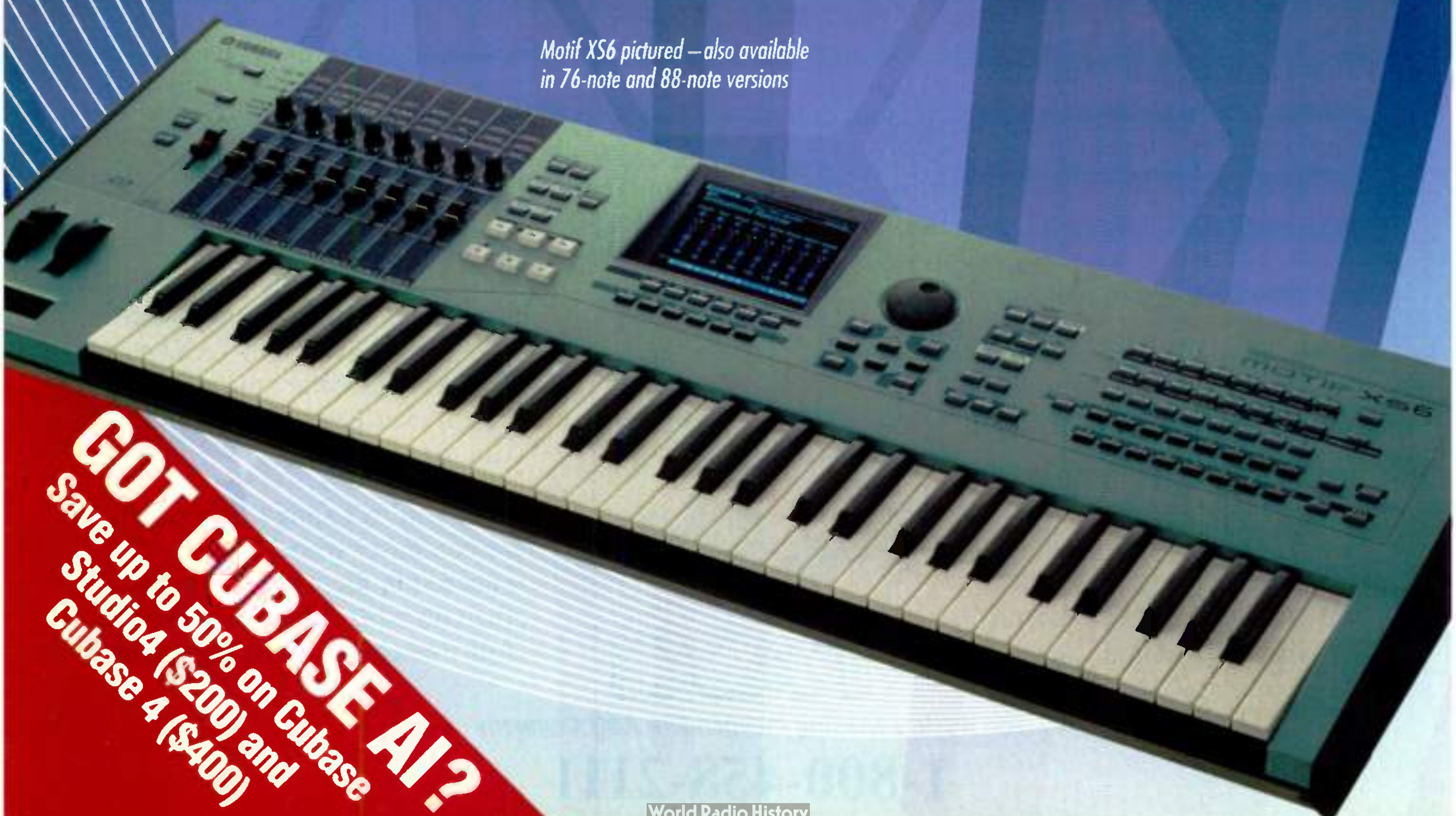
- **New UI with large color LCD and 8 knobs and sliders**
- **Xpanded Articulation synth engine with 355MB of wave ROM**
- **Studio-style mixing environment with Virtual Circuitry Modeling effects**
- **4 intelligent Arpeggiators with Direct Performance Recording**
- **Integrated Sampling Sequencer with 1GB of optional memory**
- **Total Computer Integration and Cubase AI software included**



Motif XS6 pictured – also available in 76-note and 88-note versions

GOT CUBASE AI?
Save up to 50% on Cubase Studio4 ($200) and Cubase 4 ($400)

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CUBASE

Whether you're using AI4 (the special Yamaha version of Cubase), Cubase Studio4 or Cubase 4, new Advanced Integration functions make sure your Yamaha hardware and Steinberg software work in perfect harmony. Download the AI extensions from www.Yamahasynth.com and features like true plug and play connectivity and direct importing of Motif XS songs will convince you that with Yamaha hardware and Steinberg software you really get the best of both worlds.

Cubase AI4

- 48 stereo audio playback tracks
- Up to 64 MIDI tracks
- HALion One Virtual Instrument with 150 Motif sounds
- 25 VST effects
- 8 FX return channels
- Notation with display quantize

Cubase Studio 4

- Unlimited stereo audio playback tracks
- Unlimited MIDI tracks
- HALion One with 600+ Motif sounds
- 33 VST effects
- 64 FX return channels
- Notation Plus Professional Scoring and Printing Support

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The new HS Series powered monitors were designed to be true studio reference monitors in the tradition of the famous NS10Ms. This means that mixes that sound good on Yamaha HS speakers will sound good on anything. In fact, that's the ultimate test of a reference monitor. Even better than that, HS series speakers not only sound good, they look great, too. Looking for a way to make your Motif XS sound even better? Pick up a pair of HS50Ms matched with an HS10W subwoofer and hear the Motif XS the way Yamaha did when they developed its sounds.

HS50M

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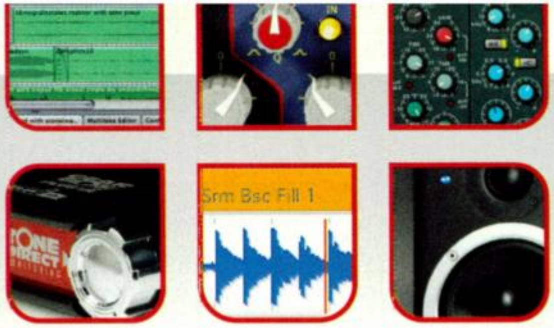
REVIEWS



PROPELLERHEAD SOFTWARE Reason 4 (Mac/Win)

If you're waiting for a Reason, here it is.

By Len Sasso

To borrow a slogan from a well-known winery, Propellerhead will sell no software before its time. It took a couple of years and then some to get Reason 4 out the door, and it was worth the wait. You get a new synth, improved sequencing, and additional pattern-based tools, as well as a bunch of smaller enhancements to make life easier.

The biggest wow factor belongs to the new Thor Polysonic Synthesizer, a semimodular, eminently programmable instrument with a huge palette of sounds. Steppin' out just got better with the introduction of the RPG-8 arpeggiator and the 32-channel ReGroove Mixer. But for those who use Reason for complete projects, the best news may be the much-needed redesign of the sequencer.

God of Thunder

Thor, the first new Reason instrument in two generations, definitely breaks some ground. For one thing, it offers five types of oscillator and one noise source, which you can mix and match at will. Similarly, you get four types of filter, each with multiple modes. Filters and oscillators are inserted into slots (three for each), and although the signal path is prewired, a number of switch points give you some control over the signal flow (see **Web Clip 1**).

Thor excels at modulation. A 13-row modulation matrix lets you route anything anywhere; beyond LFOs and envelopes, sources include Thor's audio modules and a variety of external signals. Then there's a built-in step sequencer with a lot of tricks up its sleeve. For a detailed description of Thor, see the **online bonus material** at www.emusician.com.

ONLINE BONUS MATERIAL

FIG. 1: The RPG-8 arpeggiator features a rhythm-pattern sequencer, octave shifting and repeating, several Insert modes, and the option to not repeat single notes.

Ups and Downs

One of my questions when I first saw Reason was, Where's the arpeggiator? I've asked that in reviews of each generation. The new RPG-8 module ends the wait—it has almost everything you'd want in an arpeggiator, along with a few things you've probably never thought of (see **Fig. 1**).

GUIDE TO EM METERS

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

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ASX Hardware Synth expansion board now available with hundreds of rich sonic possibilities.

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Needless to say, when you feed it a chord, the RPG-8 will cycle the notes up, down, up and down, randomly, or in the order played. It will offset the pattern by as much as three octaves and repeat the pattern over as many as four octaves. Four Insert modes—Low, Hi, 3-1, and 4-2—offer a bit more variety. The first two insert the lowest or highest held note every other beat. The latter two arpeggiate three or four notes, then jump back one or two notes before continuing the arpeggio.

For still more variety, you get a 16-step rhythm sequencer. Steps that are turned off don't play, of course, but in a nice touch, the arpeggio is spread across only the active steps. That gives you the same pitch sequences regardless of the rhythm pattern.

The MIDI Sustain Pedal (CC 64) activates the arpeggiator's Hold button to continue the arpeggio after you release the held notes, and the Single Note Repeat switch, when off, prevents arpeggiation when a single note is held. Using that, you can easily go back and forth between playing lead lines and arpeggiated chords.

Until now Reason offered no way to use MIDI note pitch or Velocity as a control source unless that option was built into a particular device. The RPG-8 converts MIDI Note messages to Reason's internal CV and Gate signals, and when turned off, it functions simply as a MIDI-to-CV and -Gate converter whose output you can route to other modules' CV and Gate inputs. For example, you could use that ability to make a pad synth's filter cutoff track the pitches of notes played in real time (through the RPG-8) while the pad synth itself is playing a recorded track.

PRODUCT SUMMARY

PROPELLERHEAD SOFTWARE Reason 4

synthesizer workstation
$399.99
upgrade, $129

FEATURES	4
EASE OF USE	4
QUALITY OF SOUNDS	5
VALUE	4

RATING PRODUCTS FROM 1 TO 5

PROS: Great-sounding new semimodular synth, Thor. Reason finally has an arpeggiator. Much-improved sequencer. Clever groove-quantization scheme.

CONS: Can't easily extract data to new lanes. MIDI file import and export lacks features. New overdub and alternate-take implementation is awkward.

MANUFACTURER
Propellerhead Software
www.propellerheads.se

Sequentially Yours

Reason's sequencer has undergone a major overhaul. Instead of being reassignable, tracks are hardwired to devices in the rack, and each device can have at most one track. Tracks have lanes for each type of data the device recognizes (notes, automation, and pattern select). You can have multiple note lanes, which play simultaneously but can be muted individually; other data types have at most one lane (see **Fig. 2**).

All data is now housed in clips. Clips are a great improvement on the data grouping in previous versions—they're much easier to move, copy, split, and join. Furthermore, you get numerical entry fields for all data points (note and automation values and positions). Unfortunately, there is no quick way to extract selected data to a new lane, for example to extract all notes for a particular drum to their own lane for groove quantizing.

Automation, which includes both device-parameter automation and MIDI performance data, may reside in note clips or in separate clips in automation lanes. During recording, MIDI performance data is placed in the current note clip. By default, parameter automation

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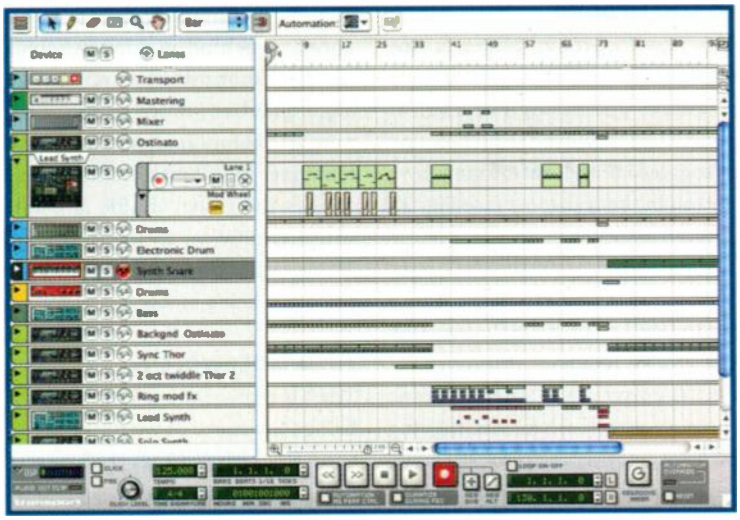


FIG. 2: Reason's greatly improved sequencer has multilane tracks with all data held in clips. Lanes can be folded for compactness or open (as the Lead Synth track is here) for easier editing.

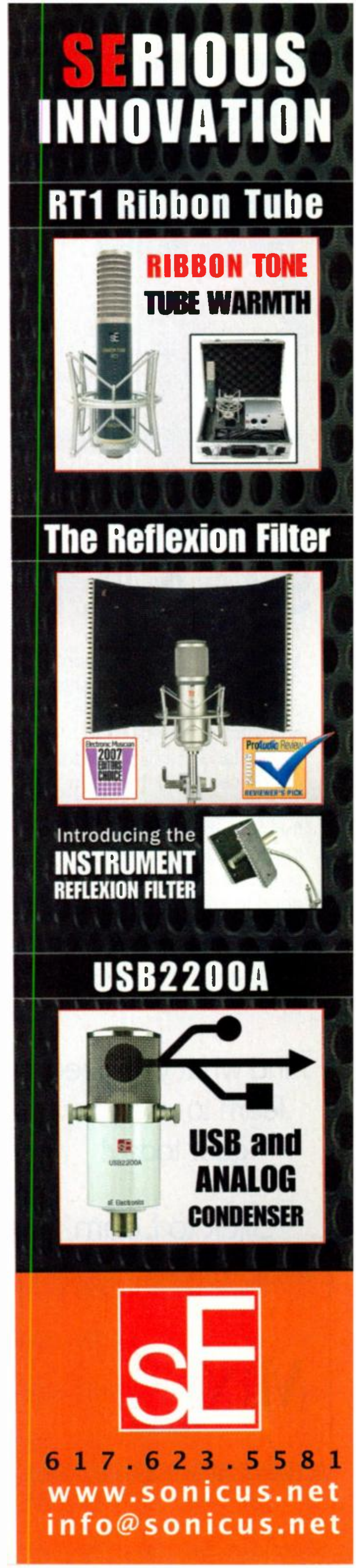
(moving device knobs and sliders) is recorded in separate automation lanes, although you can toggle that behavior. Automation data in either location is vector based (dots connected by lines), and you can edit it manually. You can also set the default value (Pitch Bend wheel centered, for instance) to be used on any portion of the lane where there is no clip. Pattern-select automation occupies its own lane, and you can now change the pattern number rather than having to reenter the automation.

The sequencer has a permanent Transport track at the top in which you can insert lanes for tempo and time-signature automation. The transport's tempo and time signature are in effect where there are no automation clips.

As in previous versions, two tools are assigned to the cursor, the alternate tool being accessed by pressing the Command (Mac) or Alt (PC) key. But unlike in previous versions, you can select any tool with a single keystroke—Selection (Q), Pencil (W), Erase (E), Razor (R), Magnify (T), and Hand (Y). That speeds up editing considerably.

You can import and export Standard MIDI Files, but owing to Reason's unorthodox MIDI implementation, the process is a bit clunky. Imported Type 1 (multitrack) MIDI files generate a new track assigned to a new, empty Combinator for each track in the MIDI file. Exported MIDI files are always Type 1, but each track is assigned to MIDI Channel 1 because Reason doesn't use MIDI channels. Unfortunately, tempo and time-signature changes are neither imported nor exported.

FIG. 3: The ReGroove Mixer lets you apply up to 32 different quantizing grooves to individual lanes.



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In the Groove

The new ReGroove Mixer is one of the slickest implementations of groove quantizing I've seen. It resides just below the sequencer and is toggled in and out of view by a button in the transport (see **Fig. 3**). You can assign each note lane in the sequencer to any of the ReGroove Mixer's 32 channels. Each note's timing will then be influenced by the groove template in that channel. A convenient Anchor Point setting controls where ReGrooving restarts. That is handy, for example, when you have an 8-bar intro and want to sync a 16-bar groove to bar 9 of the song.

Groove templates are, in effect, timing offsets for each 16th note over a span of one or more bars. You get a variety of factory templates, but more important, you can extract a template from any note clip in the sequencer. For instance, you can extract the groove from a REX file or an imported MIDI drum loop and apply it to a Redrum or Matrix Pattern Sequencer pattern once you've copied that pattern to a track (**see Web Clip 2**).

Each groove channel has a variety of settings that influence how the groove is applied. Those settings are made on the Groove tab of the floating Tool window (more on that in a moment). Sliders set the extent of the impact on timing, Velocity, and note length. For example, 50 percent timing moves notes halfway to the template position, 100 percent moves them all the way, and 200 percent offsets them to the opposite side of the template position. You use the Random slider calibrated in ticks (960 ticks per quarter note) to apply a random variation to the timing adjustments.

In addition to the settings in the Tool window, each ReGroove Mixer channel has a slider for groove amount, knobs for Shuffle and Slide, and Pre-align and Global Shuffle buttons. The groove-amount slider works relative to the sliders in the Tool window. The Shuffle knob offsets even-numbered 16th notes ahead or behind in time, whereas the Slide knob offsets all notes. Pre-align quantizes notes to the 16th-note grid before applying the groove template. The Global Shuffle button applies the global shuffle amount, which is now set in the ReGroove Mixer rather than the transport, to the notes being processed.

Also Noteworthy

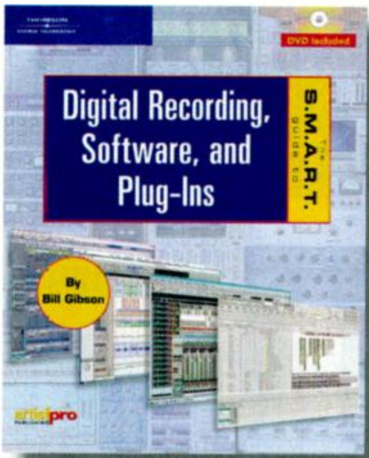
The aforementioned Tool window brings a variety of menu items to your fingertips. From its Browser tab, you can drag devices directly to the rack. The Tools tab provides sequencer-management tools for quantization; pitch, Velocity, legato, and note-length adjustment; tempo scaling; note scrambling; and automation cleanup.

The Combinator's Programmer lets you filter out MIDI Pitch Bend, Mod Wheel, Breath, Expression, Sustain, and Aftertouch messages. Also, all knob and button source fields are variable (you can route a single knob to ten destinations, for instance). The NN-XT sampler features chromatic automapping and multizone editing of sample settings.

In the transport, playback position is displayed in bars, beats, 16th notes, and ticks as well as hours, minutes, seconds, and milliseconds. All fields are editable. You also get buttons for precount as well as for overdub and alternate-take recording, although the latter two would be more useful as toggles than as triggers.

As with all previous upgrades, it's hard to imagine a Reason user not parting with the modest upgrade fee. If you're still not sure Reason is for you, this might be the time to grab the downloadable demo and have a listen. The patchable rack of instruments and effects is by itself worth the price of admission.

Len Sasso is an associate editor of EM. For an earful, visit his Web site at www.swiftkick.com.



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Whether you are recording voice, instruments, or field audio for a movie soundtrack, you need to understand the digital recording process and the seemingly limitless potential of all the available software and hardware tools. With this guide, you'll learn what you need to know about modern digital recording — whether you're operating in your home studio or in a professional recording facility.

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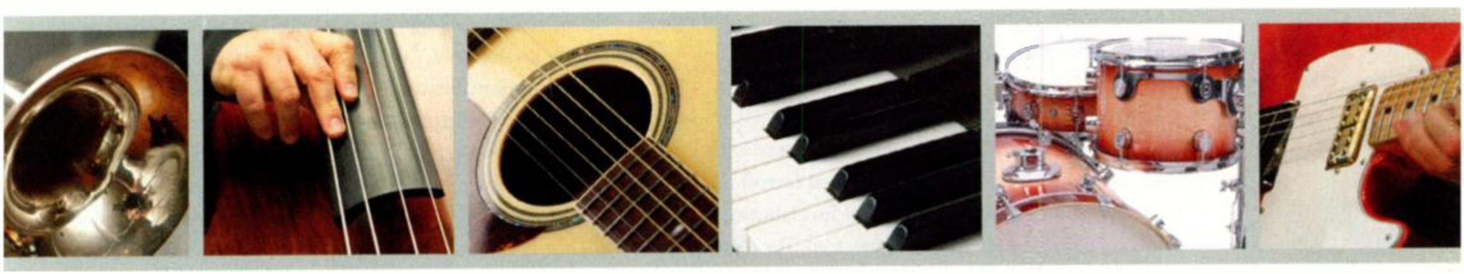

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FIG. 1: Digidesign's RM1 monitor features digital signal processing and an unusual design that produces clean bass in a cabinet that's neither sealed nor ported.

DIGIDESIGN RM1

Small but powerful monitors for studios of all sizes.

By Rusty Cutchin

Digidesign has expanded its product line aggressively in the last several years, and it was no surprise to see the company put its name on studio monitors. However, the new RM1 monitor (and its bigger sibling, the $1,749 RM2) isn't your everyday speaker. This monitor, the result of a collaboration between Digidesign and English speaker manufacturer PMC, is a precision instrument with onboard DSP, a revolutionary design, and a sound worthy of any studio, whether pro or project.

Dynamic Design

The RM1 is a compact biamplified monitor with a 5.5-inch low-frequency driver and a 1-inch ferrofluid-cooled high-frequency driver (see **Fig. 1**). It furnishes analog and digital (AES3) inputs on XLR connectors. The digital input accepts standard resolutions up to 24 bits and 96 kHz. Analog-controlled Class D amplifiers provide the power.

The RM1's front panel suggests elegant simplicity. The jet black finish (matte everywhere except for the glossy woofer cone) is broken only by the silver of the woofer mounting ring and the nameplate, along with the blue power indicator to the left of the soft-dome tweeter.

The rear panel is where clues to the RM1's distinctive design become apparent (see **Fig. 2**). A voltage selector with a slotted opening is placed just above the power toggle switch and cable receptacle. If you're traveling with the monitors and need to switch between the 115 and 230 VAC settings, then you must also switch fuses. The former requires a 3.15A fuse and the latter a 1.6A. The fuse holder is next to the power cable receptacle.

In the center of the rear panel are four threaded anchor points, similar to the VESA mount holes on flat-panel televisions. These are designed for connecting to custom tilt-and-swivel wall brackets. The RM1 manual says to check Digidesign's Web site for these mounts, but I found

> Electric bass balanced beautifully with the crisp highs and clean mids.

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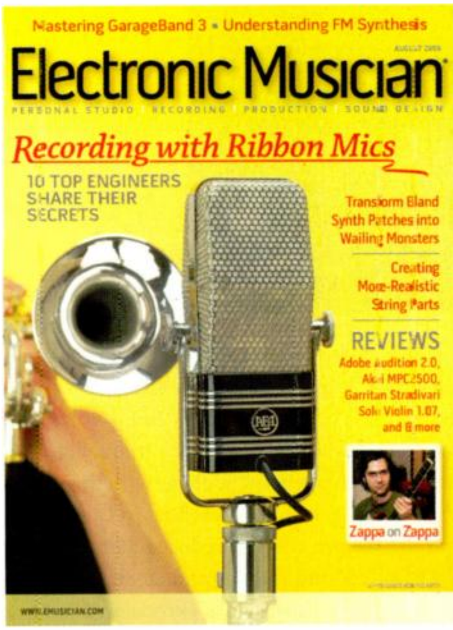

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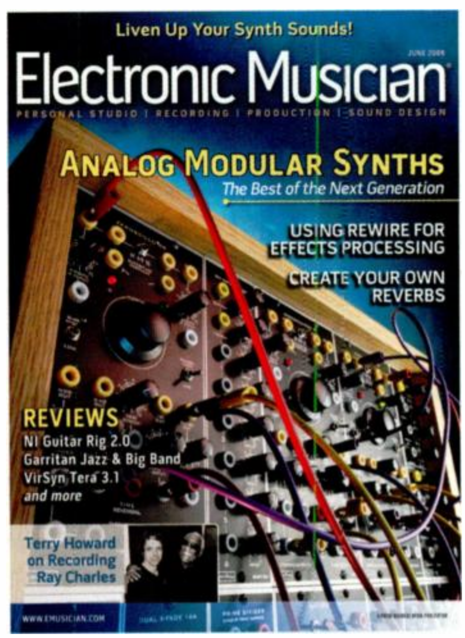

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FIG. 2: The RM1's rear panel supplies XLR connectors for analog and AES3 digital input, along with RJ45 jacks for passing digital audio to or accepting it from another RM1.

no information posted there. According to Digidesign, you can buy the appropriate wall mounts (the DB1 Wall Bracket) on PMC's Web site (www.pmc-speakers.com).

Next on the RM1's rear panel are the user controls. First up is a mini toggle switch for bass-port emulation. The RM1's design achieves excellent bass response without a port (more on this later), but engaging the circuit allows users to hear how a mix (quoting the manual) "translates to a ported speaker." According to the RM1's reference diagrams, the switch provides a boost that reaches about 2 dB from 50 to 500 Hz.

Next to the port-emulation toggle switch are the HF and LF gain-adjustment pots. These slotted posts protrude just enough for the nimble fingered to adjust them by hand. Both access shelving EQs. The HF pot provides adjustments from –4 to +3 dB in 0.5 dB steps with a 1 kHz corner frequency. The LF control gives you –4 to +3 dB in 0.5 dB steps with a 750 Hz corner (500 Hz on the larger RM2). Also onboard is a master gain-trim adjustment post, which gives you a range of 0 to –15 dB.

Digital Efficiency

At the bottom left of the RM1's rear panel are two RJ45 (Ethernet-type) sockets labeled Thru and In for connecting the second monitor in a stereo digital setup. You connect the first monitor with an XLR cable at the Digital In/AES 3 port. You connect the supplied RJ45 cable to that monitor's Thru port and the other monitor's In port. A Channel Assign mini toggle switch (next to the gain trim pot) tells the monitor which digital channel you want it to reproduce.

The RM1 manual provides no specific instructions for using the RM1 in a surround setup other than a generic speaker-placement diagram. If you have a 5.1 AES setup, you shouldn't have any trouble assigning center- and rear-channel data to multiple RM1s. Digidesign says that for 5.1 mixing, the RM1 and RM2 monitors match up well with PMC's TLE-1s sub, which would handle the surround LFE channel. For 2.1 mixing, you can use any sub you choose that offers bass management.

If you're monitoring in stereo and feeding analog signals to the RM1 (as I was doing for this review), you're likely to be supremely impressed (as I was) with the RM1's bass response, stereo imaging, and surprising sound-pressure levels (SPLs). The secret appears to be PMC's Advanced Transmission Line technology, which involves placing the main driver at the end of a long tunnel (the transmission line) that's composed of a series of internal baffles.

The tunnel is lined with acoustic material that absorbs upper bass frequencies and keeps the lowest frequencies in phase with the main driver, extending its LF output. According to Digidesign and PMC, the pressure created reduces distortion, makes the midrange sound cleaner and clearer, and produces a higher SPL and lower bass extension than ported or sealed cabinets of a similar size using identical drivers. The proof of these claims was definitely in the listening, especially regarding SPLs.

Blown Away

It's not easy to hide the fact that you've just hooked up a pair of RM1s. Considering that 80W RMS (woofer) and 50W RMS (tweeter) amps drive them, these babies are loud. I switched the XLR cables connected to my Mackie Onyx 1620 from my usual monitors (a pair with higher-powered amps and similarly sized drivers) to the RM1s and was stunned by the volume the latter produced at identical settings. After lowering the output from the board, I was pleased to hear no change in the level of detail. The RM1s held true to their claim of consistent frequency response at all volumes.

In addition, stereo imaging was excellent, and little details like string noise, sibilance, and bass guitar attacks, which can get lost or mangled by second-rate speakers, sat in their proper place on professionally finished mixes. Off-axis response showed the same consistency as frequency response. It was clear immediately that the English-built RM1s were a cut way above even the best budget-priced monitors (usually

PRODUCT SUMMARY

DIGIDESIGN RM1

active monitors
$1,249 each

FEATURES	4
EASE OF USE	5
AUDIO QUALITY	5
VALUE	3

RATING PRODUCTS FROM 1 TO 5

PROS: Excellent sound. Distinctive design. Superior bass and imaging. Efficient power handling.

CONS: Rear-panel adjustment posts hard to grasp. Pricey.

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of Asian origin) that have emerged in recent years.

On program material like the hard-rocking "Paralyzer," by Finger 11, guitar attacks were audibly distinct from hi-hat hits and identifiable in the stereo field of an otherwise typical rock mix heavy on the low mids. Fantasia's "When I See U" radio mix, which has only a kick drum and no musical bass, seemed to benefit from engaging the port emulation switch, while the near-distorted bass of "Make Me Better," by Fabolous with Ne-Yo, confirmed my suspicion that although mixing hip-hop on the RM1s would be reliable, players and DJs in the house would probably get a better vibe from the larger RM2 or with the RM1s matched to a high-quality subwoofer.

That's not to say that the RM1 has any problem with bass—far from it. On traditional material, like my own jazz-influenced country projects, or a traditional country song with a modern mix, like Vince Gill's "Cold Grey Light of Gone," electric bass balanced beautifully with the crisp highs and clean mids of the RM1. The growling low end of the Del McCoury band's upright bass on the Gill tune made me think I had the port emulation switch on, but I didn't. Still, lower-frequency jams like Fabolous's cry out for a range extending below 50 Hz.

Well Adjusted

Changing settings on the RM1 differs from adjusting analog powered monitors, because all internal processing, including the crossover, is digital. The HF, LF, and Gain Trim pots don't have stops; they can turn continuously. When you land on one of the marked settings, the adjustment becomes audible after a very short delay. Considering that many people will have to use a thumbnail or screwdriver to change these settings, it's good that you probably will have to set them only once. It would be nice to have easier access to the bass-port emulation switch, especially if you plan to mount the monitors on the wall.

But that's a minor complaint. On the whole, the RM1 reflects a successful marriage of the digital studio know-how of Digidesign and the audiophile design skills of PMC. The RM1 is an efficient, manageable, and great-sounding beast of a compact monitor that should be a match made in heaven for Pro Tools HD studios. With a footprint that a space-challenged studio owner could love, it should fit just as well into any high-end project studio.

Rusty Cutchin is a producer, engineer, and music journalist in the New York City area.

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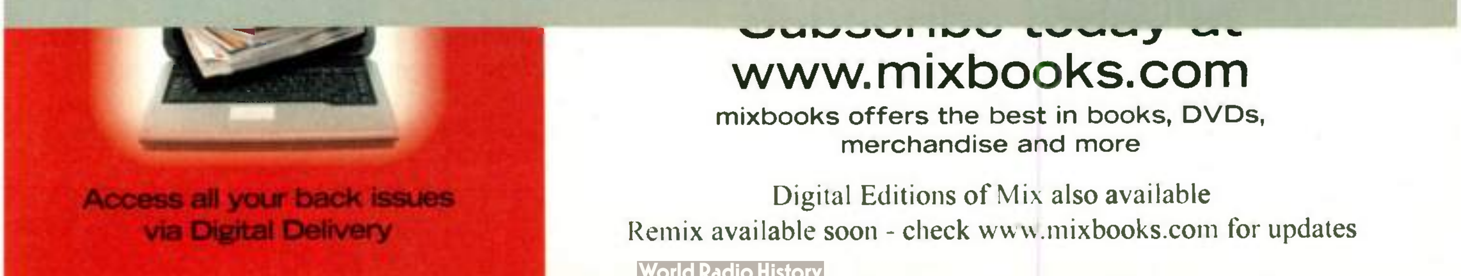

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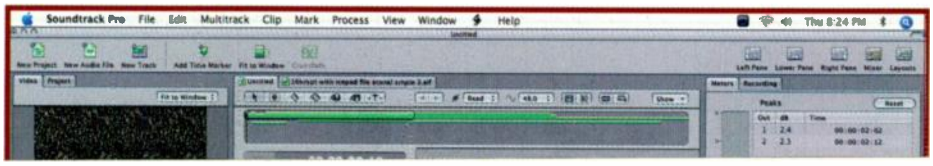

FINAL CUT STUDIO

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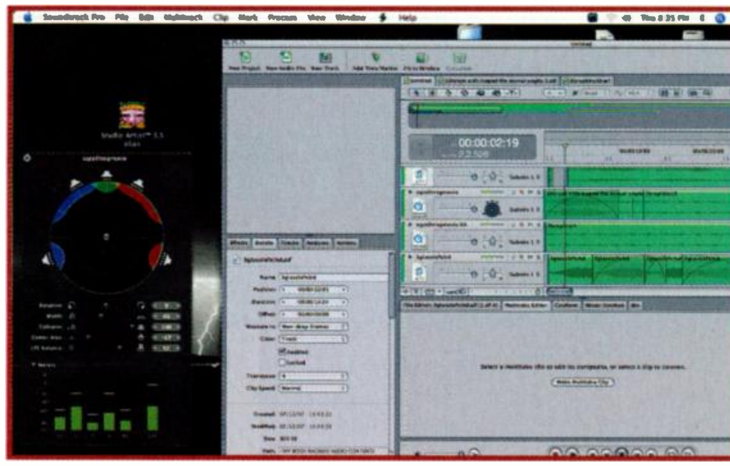

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FEATURES	4
EASE OF USE	4
DOCUMENTATION	4
VALUE	4

RATING PRODUCTS FROM 1 TO 5

PROS: Vast range of audio and video software. Good integration among programs in bundle. Good video-scoring tools.

CONS: Soundtrack Pro File Editor and multitrack Timeline need better integration.

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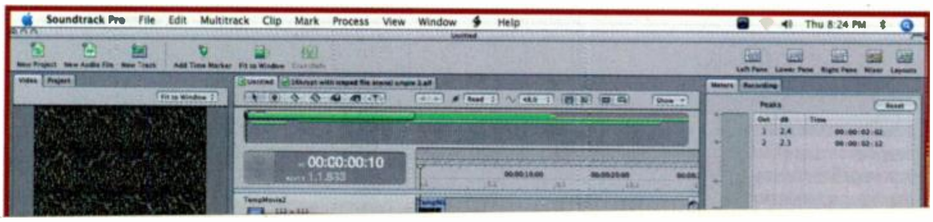

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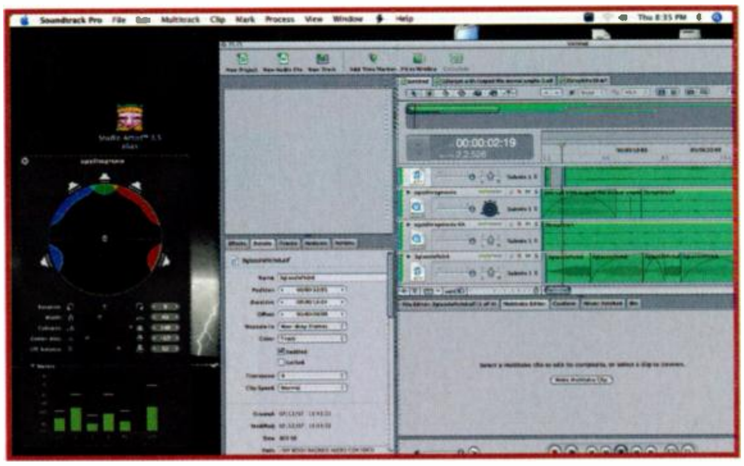

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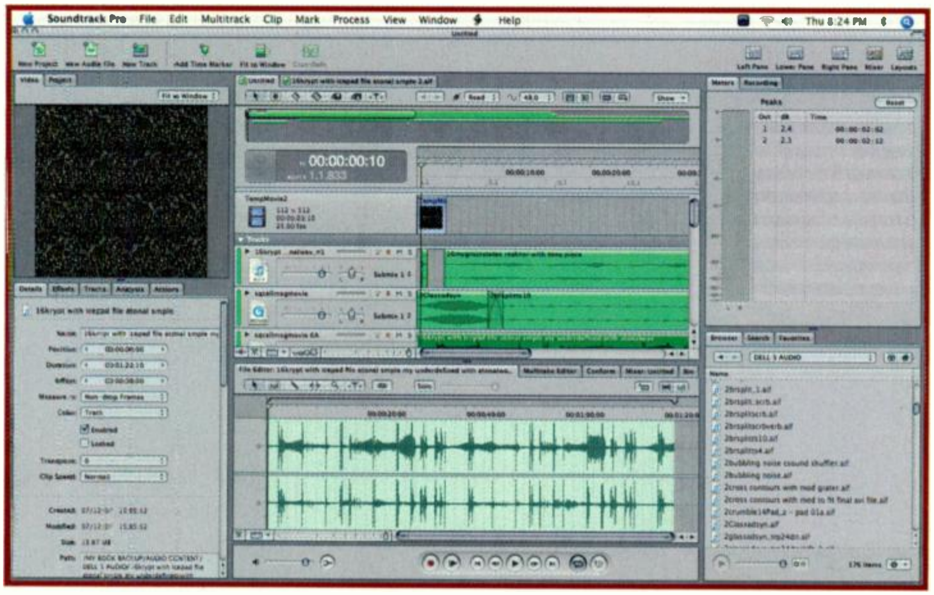

FIG. 1: Soundtrack Pro's interface includes the Timeline, which is a multitrack view, and the File Editor, where you work on a single mono, stereo, or multichannel file. You can save multiple layouts that include whichever work areas you want.

APPLE Final Cut Studio 2 (Mac)

A multitrack video and audio editor and a lot more.

By Dennis Miller

Apple's Final Cut Studio 2 (FCS 2) is a suite of high-end professional audio- and video-editing tools. Of particular interest to musicians will be the new version of the multitrack audio editor Soundtrack Pro (SP 2), which is the focus of this review. SP 2 has undergone a number of major enhancements, including support for multichannel files and multichannel metering (up to 24 channels), surround mixing, new DSP functions, and overall efficiency boosts. As expected, SP 2 also provides better integration with the video apps in the suite.

> You can save as many customized screen layouts as you need.

We've reviewed Soundtrack Pro in the past, most recently version 1.0 in the September 2005 issue (available online at www.emusician.com), so I'll mostly cover the new features in this article. (For information about the main video software in the bundle, see the **online bonus material** at www.emusician.com.) Note that Soundtrack Pro 2 is also available in the newly released Logic Studio ($499 [MSRP]; upgrade from Logic Pro 6 or 7, $199), where it joins forces with Logic Pro 8 and a number of other music applications. Look for a review of Logic Studio in a forthcoming issue.

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Out of the Box

Like its predecessor, Soundtrack Pro 2 offers both a dedicated editing view called the File Editor, which can handle mono, stereo, and multichannel files, and a multitrack Timeline view (see **Fig. 1**). The Timeline supports an unlimited number of audio tracks as well as a single video track, but there's no support for MIDI (other than syncing to MTC). Some functions (time-stretch, for example) are available only in the File Editor, and it's easy to open a clip from a track in the Timeline in the Editor. However, once you've made a change to a clip, it's not so simple to replace the old version with the new one—you must first save it as a new file and then reload it. Fortunately, integration among other elements of the program is much better.

SP 2's interface consists of a number of modular windows that you can freely reposition. Unfortunately,

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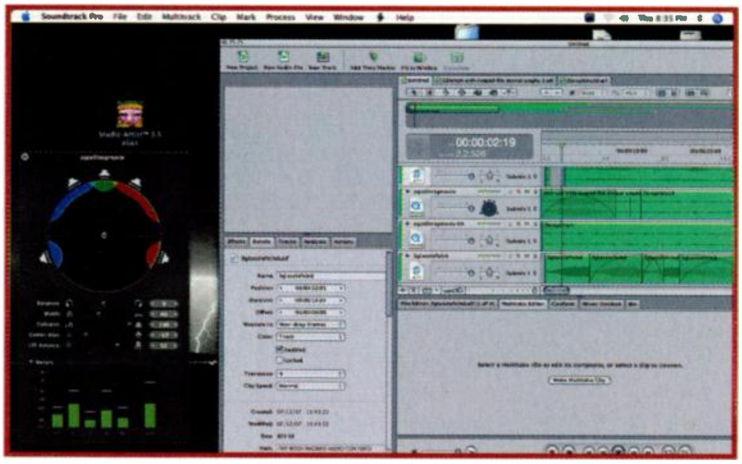

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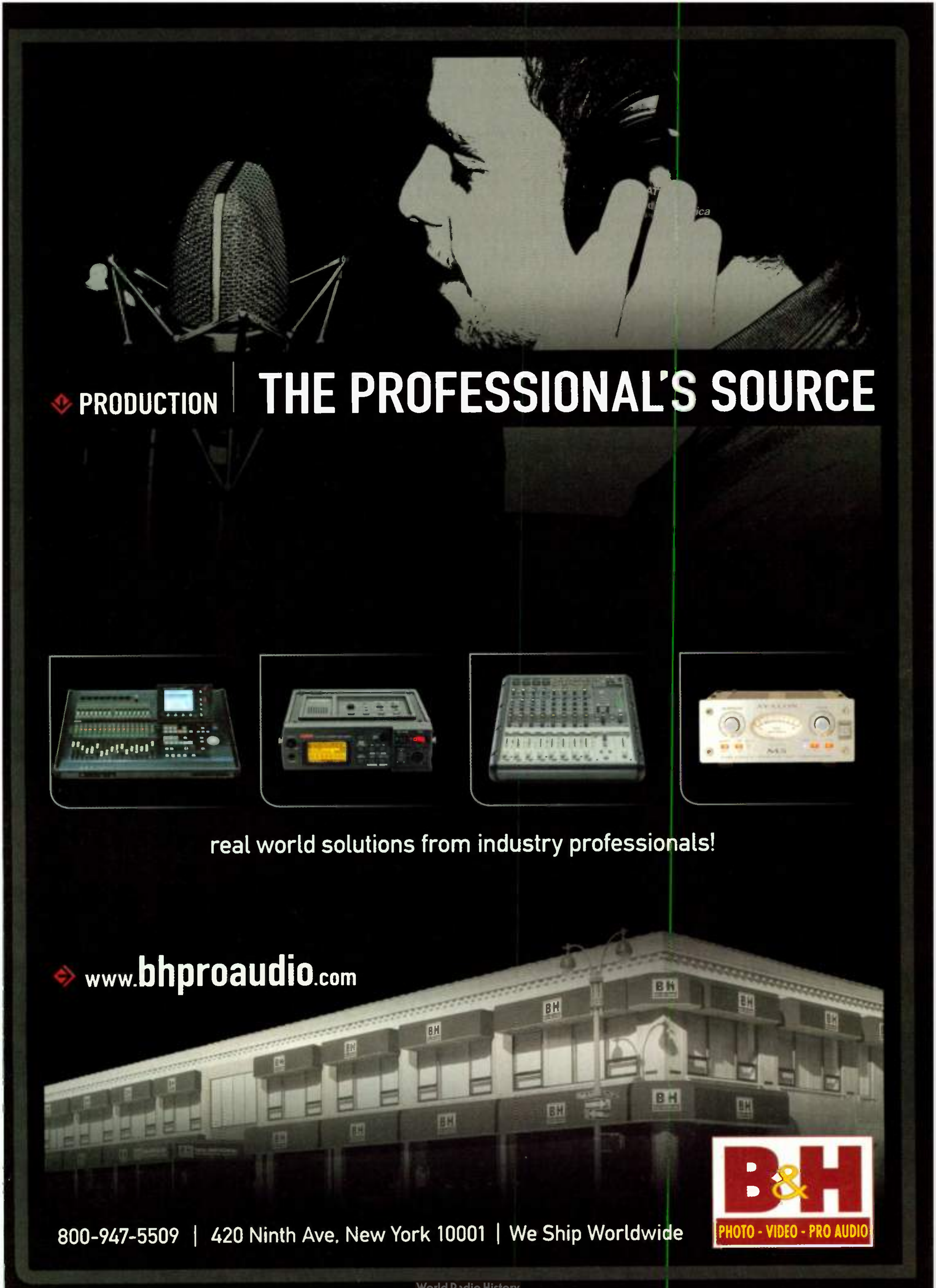

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and a larger HUD will appear (you can also access the Surround Panner HUD from the Mixer; see Fig. 2). All of the surround parameters can be automated, and you can create a surround mix even if your hardware doesn't support multichannel output. That option isn't offered by all audio editors.

The Spectrum Editor has received some minor enhancements; for example, the new Frequency Selection tool lets you isolate any range of frequencies and process only that area. The processing is limited to cut, copy, paste, and amplitude boost, however—unlike in Adobe Audition and Steinberg Wavelab (both Windows only), there's no way to add reverb, delay, or any other effect to just a highlighted range of frequencies. Speaking of effects, SP 2 ships with dozens of impulse responses for use with the newly included Space Designer convolution plug-in, and you can also use any AIFF file you want as an IR. Space Designer has a very intuitive interface and would be useful both for simulating room ambiences and for cross-synthesis purposes.

Take That!

The Multitake Editor is a handy new tool for managing recording sessions that require multiple takes. Set the Cycle Region to cover the duration of your recording (or use the entire track), arm the track, click on Record, and off you go. The program will record until the end of the specified duration and then automatically rewind and continue to record. When you've completed as many takes as you need, click on Stop, and you'll see each individual take appear as a separate clip on its own track in a new window below the main Timeline, along with another track that represents the composite of all the individual takes.

To create a finished composite take, use the Razor tool to split each take into as many segments as you want, then highlight each segment of each take that you plan to keep. The composite take automatically updates as you make selections, so you can preview your progress at any point. You can easily solo a complete take or just a segment, scrub individual takes, add fades at cut points, or move the clips forward or backward. Overall, the Multitake Editor makes creating seamless composites easy and keeps the Timeline screen from getting needlessly cluttered.

Point of View

SP 2 supports only a single video track, and the track can contain only a single video file that is always positioned at the very beginning of a project. As expected, though, it does have very capable features for syncing audio cues with video scenes.

Using the new Multipoint Video HUD, for instance, you can get a clear idea of what's happening in the video at the point where an audio clip starts and ends. Enable the Multipoint Video HUD and click on an audio clip, and the HUD will display the exact video frames at the start and end of the clip and at the location of the playhead (see Fig. 3). Then drag the right end of the audio clip to extend its length, and the right panel of the HUD will update to

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display the frame at the new clip end point. The same feature is available if you're moving or time-stretching a clip. (It would be nice if the video image also updated as you drew volume curves on a clip, for example, showing exactly where a fade-out started or a fade-in ended.)

Not surprisingly, you can display video on an external RGB monitor if you have the proper hardware, and you can remove or edit and replace the audio from a QuickTime movie. Soundtrack Pro also now has the ability to nudge an audio clip in increments of frames, which should make video editors happy.

Buying a bundle like Final Cut Studio 2 just for the audio capabilities is not something I would suggest. In fact, if you want only the multitrack editing features of SP 2, then the new Logic Studio combo is a better bet. But if you've decided to take the plunge into video, this well-conceived audio and video workstation is definitely worth a look. The tight integration of the programs in the bundle, and the enormous combined resources they offer, can take you from the back lot to the main stage.

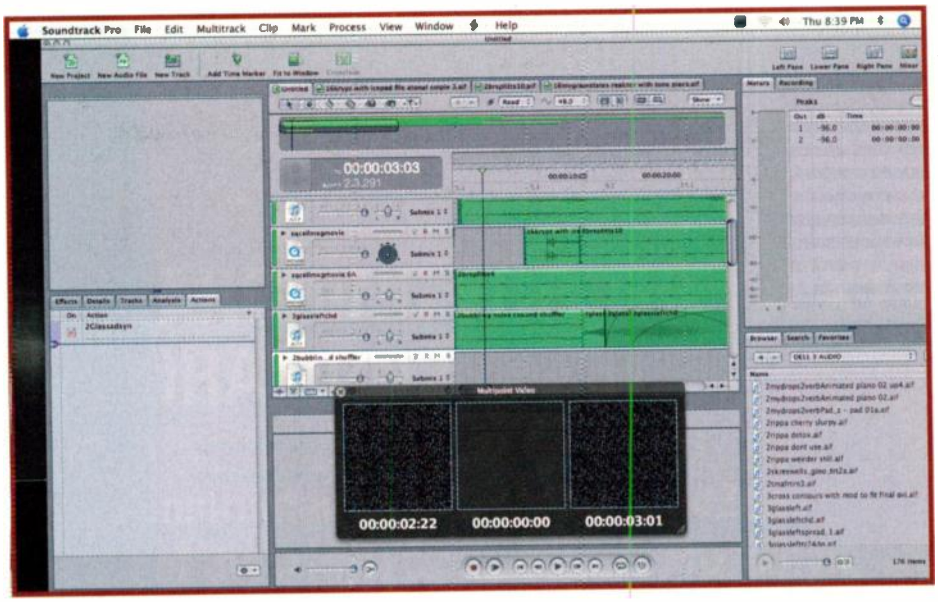

FIG. 3: The new Multipoint Video HUD makes it easy to line up an audio clip with a point in a video.

Associate Editor Dennis Miller is a composer and animator. Check out his work at www.dennismiller.neu.edu.

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and Tannoy PBM 8s—which run off an Adcom amplifier and are supplemented by a custom subwoofer. The KRKs were used with flat EQ, no limiting, and were also paired with the subwoofer during testing. The system level pot was set to +2 dB to match the level of the passive monitors.

The first thing I noticed on the KRK monitors was their amazingly sharp stereo imaging: the precise panning placement was so clear that it was a little hard to adjust to. I had never experienced this kind of imaging precision in my control room before, with obvious spatial gradations occurring in between my normal one o'clock/two o'clock panning increments.

I also noticed right off the bat that the test monitors had an enhanced midrange detail that I initially liked, even though this characteristic seemed excessive and unbalanced compared with the midrange-rich Event monitors. On a CD of alt-country ballads and up-tempo rockers I was auditioning—Val Esway's *Reason to Believe*—the VXT6 pushed the acoustic guitar, acoustic bass, and electric guitar to the foreground, but diminished airiness on the vocals, snare, and reverb tails compared with the Events. The KRK speakers were also a touch more sibilant compared with both sets of passive speakers.

On the other hand, the VXT6s were more revealing and detailed across the board than the Events, highlighting nuances in room sound, vocals, and occasional noise or distortion artifacts that I hadn't noticed before. On a pop-oriented number, the electric guitar and vocals were harsher around 3 kHz than I remembered them being on a number of other real-world speaker systems. Loud cymbal crashes were also more brittle on the KRKs compared with what I was used to hearing in these mixes. Overall, I found the KRK monitors to be much more complementary and listenable during the softer material on Esway's CD. But the rock numbers were too edgy for my taste, and slightly fatiguing to the ears after about 30 minutes of listening.

While auditioning a jazz-oriented studio sampler CD, I appreciated the depth and detail of the VXT6 pair's solid bass and how it conveyed the room sound. On a handful of these jazz tracks, the cymbals were again too bright in the upper mids for my taste, as were some saxes and brass instruments. But all in all, the jazz material was smooth and easy on the ears over the KRK monitors.

PRODUCT SUMMARY

KRK SYSTEMS VXT6

active monitors
$599 each (MSRP)

AUDIO QUALITY	3
VALUE	3

RATING PRODUCTS FROM 1 TO 5

PROS: Excellent imaging and detail. Range of useful EQ and other adjustments. High-quality electronics and design. Padded base.

CONS: Prominent midrange. System level-adjust pot is difficult to see.

MANUFACTURER
KRK Systems
www.krksys.com

FIG. 2: The switches for EQ, clip indication, and automute are protected by covers so that they can't be accidentally changed.

In the Mix

Those of you who have done a lot of mixing know that there is an inverse relationship between monitor coloration and the resulting mixes. For example, audiophile-type speakers with a lot of glossy high end sound great for listening but can yield mixes that are dull or lackluster. After the initial auditions with the VXT6, my concern was that they exaggerated elements—upper midrange, reverb, bass—that could, in turn, end up being underrepresented in a mix. The lack of high-end air could also lead engineers to boost the upper frequency range, although some of that compensation is often desirable, especially for pop music. So I set out to put these issues to the test with some mixing comparisons.

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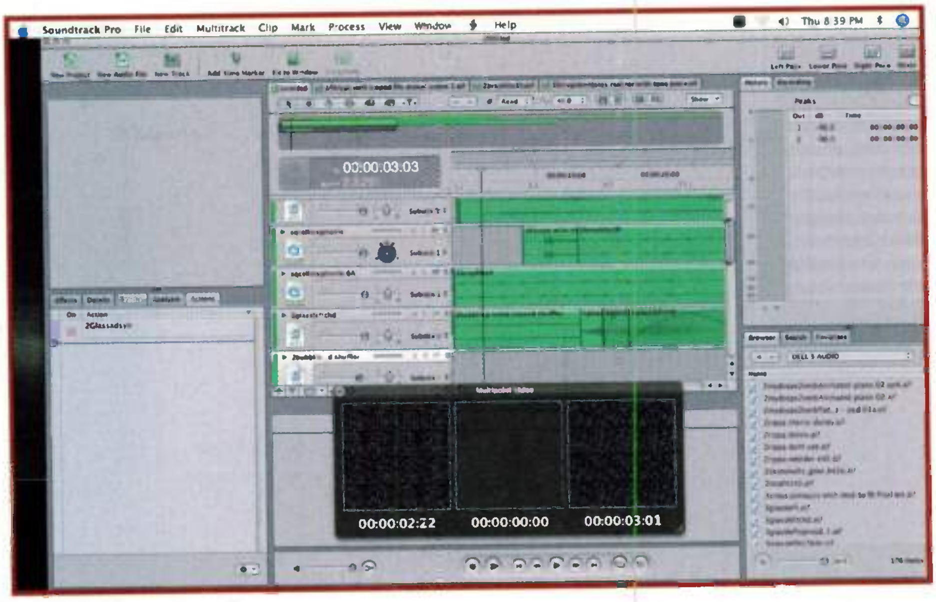

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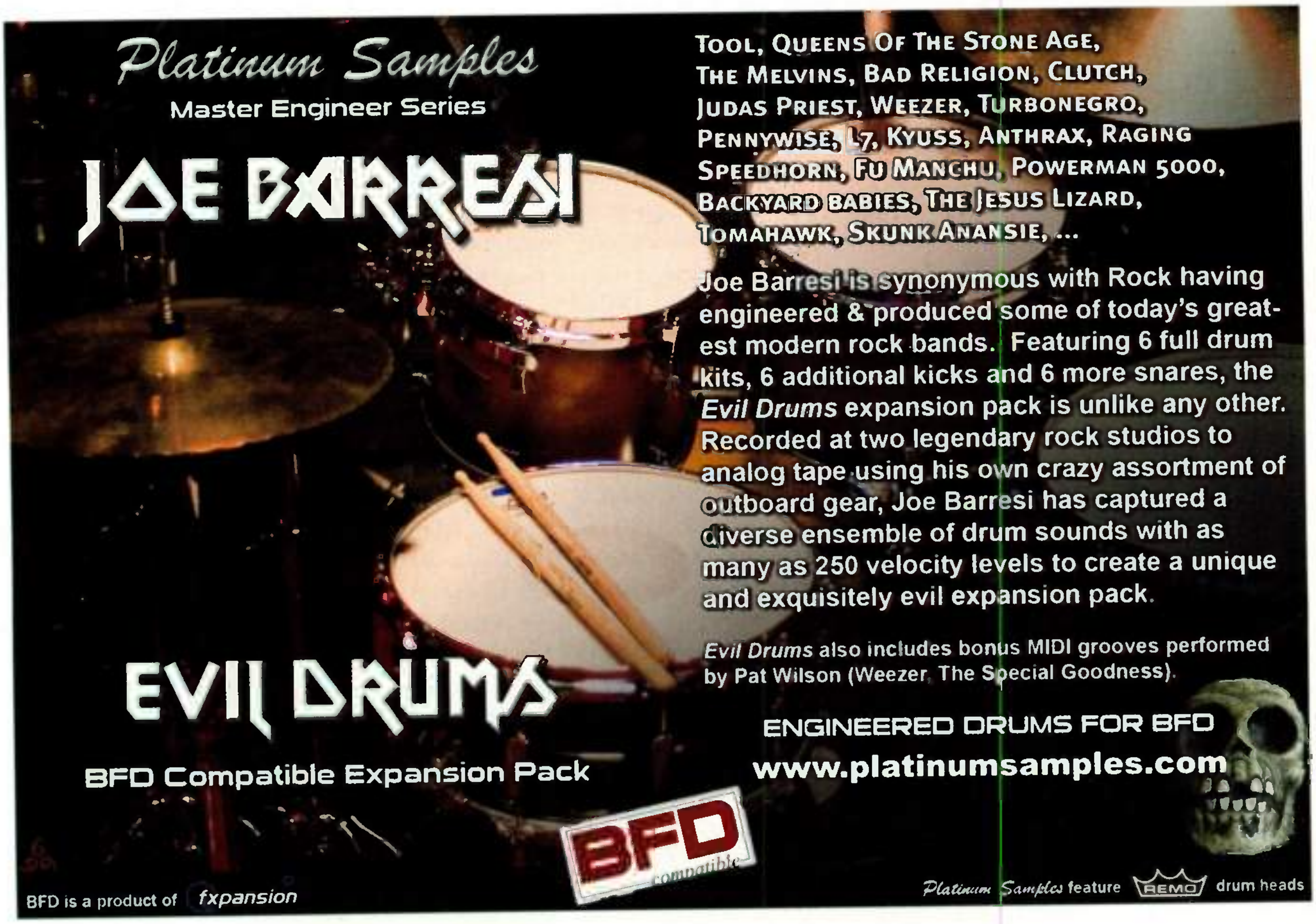

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FIG. 1: The VXT6's cabinet design helps give this monitor its accurate imaging.

KRK SYSTEMS VXT6

See Product Specs @emusician.com

A sleek new design with accurate imaging.

By Myles Boisen

KRK's new VXT studio monitor series includes three internally amplified models. The VXT6, the subject of this review, is a hefty 27-pound biamped close-field monitor with a 1-inch silk-dome tweeter, a 6-inch woven Kevlar woofer, and a sleek molded enclosure with no corners or right angles (see **Fig. 1**).

The VXT6 has numerous rear-panel adjustments for tuning its response and performance (see **Fig. 2**). A row of toggle switches at the top of the panel offers options for ground lifting, peak limiting/clipping indication, and automuting in the absence of an audio signal. (LEDs on the front of the speaker show the status of the amplifier peak limiting/clipping indication functions, when engaged.)

An HF EQ switch provides three equalization options: flat, –1 dB, and +1 dB (above 1 kHz). The LF adjust control—a 3-position switch marked whole (normal), quarter, and half—varies the monitor's low-end response below 200 Hz. The system level-adjust pot controls a range of input gain from –30 to +6 dB. Plastic covers protect all the toggle switches from getting bumped when you reach behind the speaker.

Also located on the rear panel are a Neutrik combo jack with XLR and ¼-inch TRS inputs, an IEC AC power connector, an on/off switch, and a switch for 110/220 VAC. The manual is thorough and well written, supplying explanations for all adjustments as well as a variety of setup and troubleshooting options.

With the Program

Installing the VXT6 monitors was quick and easy, with all rear-panel controls clearly marked and fairly self-explanatory. One drawback is the level-adjust trim pot, which has a very small black shaft and is hard to see, even in good light. However, this will be a set-it-and-forget-it adjustment for most users.

I appreciated the cabinet's rounded look, which provides sonic benefits such as randomizing reflections off the cabinet's surface to improve imaging. The nonslip foam-padded base is a thoughtful touch that is effective for reducing resonance when the monitor is placed on a shelf or speaker stand. Omnimount screw sockets are also provided in the base.

Once the KRK active monitors were connected and positioned for close-field use, I auditioned a variety of recordings I had engineered myself. Throughout the audition process, I compared the VXT6s with my standard control-room speakers—passive Event 20/20s

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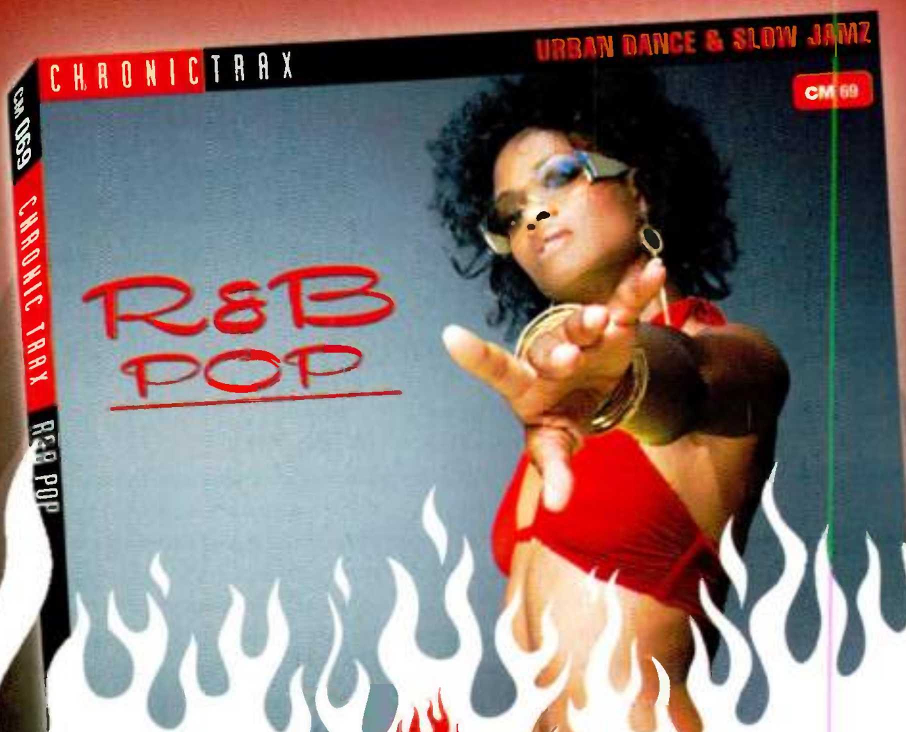

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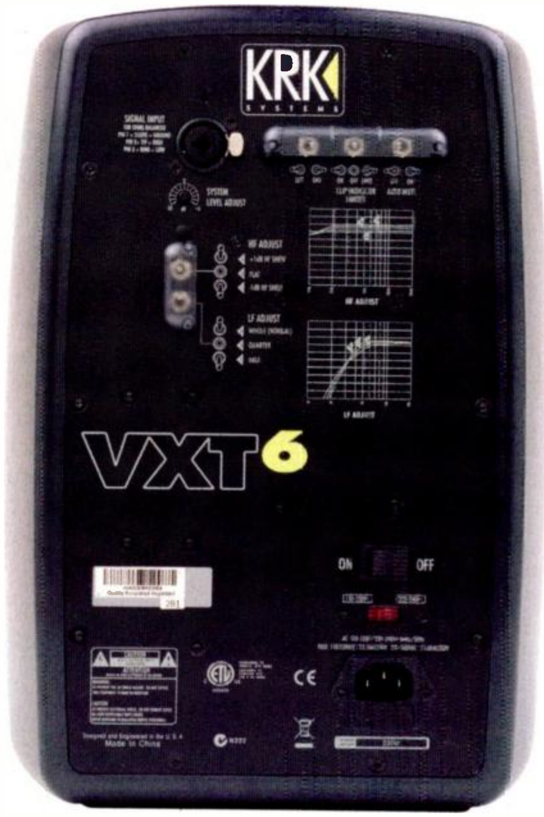

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vocal, as well as the bass, which seemed much easier to hear on this monitor pair. I also nudged the faders up on the kick drum, rhythm guitar, and drum room mic.

When comparing the second mix on both sets of control room speakers with the subwoofer on, I found that the VXT6s were much more sizzly in the highs and delivered much more kick drum. This contributed to a feeling of the mix being hotter without any actual level changes. The Event speakers were smoother and less fatiguing, with a midrange perspective that felt more trustworthy to my ears.

Listening back over the Dynaudio BM15 monitors in my mastering room, I felt that the Event mix was better balanced overall, with a more satisfactory kick/bass blend. In the midrange there was plenty of organ, even after pulling the level down, and a more equitable organ/guitar relationship. However, the KRK pair worked better for getting the vocal placement right. On the other hand, when mixing on the VXT6 monitors, the bass overshadowed the kick drum, and the organ and horns were too loud despite my worries about undermixing these midrange instruments.

After spending time with the VXT6s, engineer Bart Thurber, who shares my studio, found that the monitors were a good choice for tracking for his types of projects—mostly rock and punk—because the monitors allowed him to hear plenty of detail while recording. But he also felt that the monitors were not comfortable for mixing, due primarily to their forward midrange and prominent highs.

Thurber also noted an unevenness in the response around 120 Hz, which he described as a hole in the bass range, making it difficult to balance electric bass and kick drum. After my mixing experience, I concurred with him. Thurber also took the initiative to attenuate the highs, setting the HF switch to –1. This adjustment effectively dulled the brittleness but did not eliminate it.

Sound and Vision

Although the KRK VXT6 has a distinctive sound that takes some getting used to, it does have a lot going for it, including crystal-clear imaging, excellent detail and resolution, and an eye-catching modern look that delivers real sonic benefits. The quality of KRK's electronics and design is apparent, and it results in clean, undistorted audio that is in keeping with the monitor's price point (the street price is less than $1,000 a pair). Although my taste in monitors definitely leans toward a smoother, flatter sound, I could get accustomed to the VXT6 voicing for mellower styles such as jazz and classical music, or for the work of singer-songwriters.

Myles Boisen is the head engineer at Guerrilla Recording in Oakland, California. He can be reached through his Web site at www.mylesboisen.com.

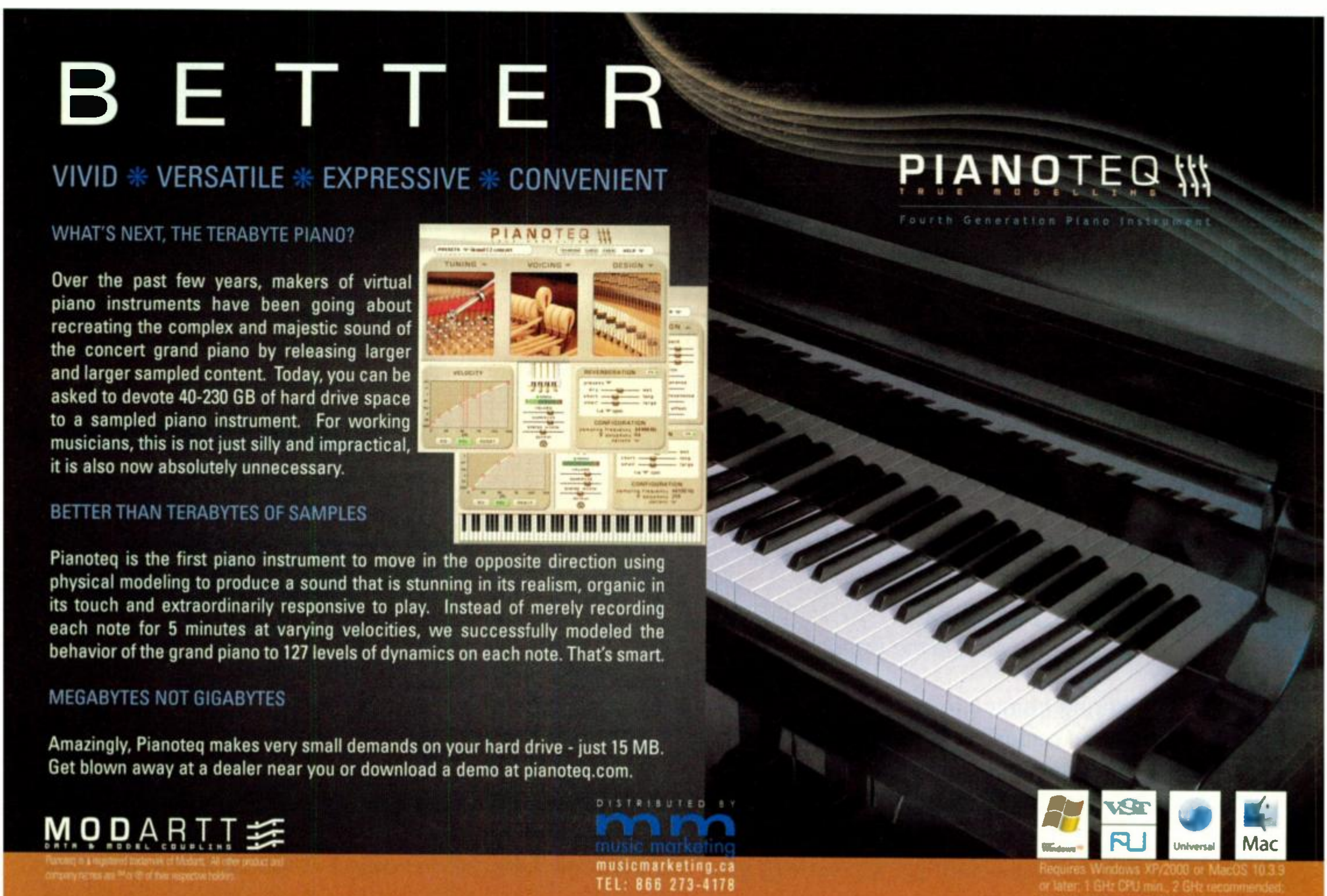

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FIG. 1: The Neve 88RS plug-in has features galore and keeps the Neve aesthetic intact.

UNIVERSAL AUDIO

Neve 88RS and LA-3A

(Mac/Win)

Two more solid plug-in contenders for the UAD.

By Eli Crews

Universal Audio's UAD DSP host system has garnered a lot of respect in the audio community, becoming a staple in many serious digital audio workstation setups. UA's knack for modeling the most revered analog devices in the world has set it apart in the crowded field of software emulations. Two new plug-ins—the Neve 88RS and the LA-3A—expand the possibilities for UAD system owners in the realm of high-quality digital processing.

Neve 88RS

The 88RS plug-in is based on a channel strip from the highly regarded 88RS, AMS Neve's 21st-century large-format analog console (see **Fig. 1**). The plug-in version has much of the functionality of its analog predecessor and is split into four basic sections. The Dynamics section has separate expander/gate and compressor/limiter circuits, each with its own switch to put it into the signal path. The 4-band EQ section is next, and then the high and low Cut Filters. The Global section is home to a polarity-reversal button, a switch for repositioning the EQ to be predynamics, and a button for moving the EQ into the dynamic sidechain for frequency-dependent compression (such as de-essing). A red power switch bypasses the entire effect.

Although I've not used an 88RS desk, I have worked a fair amount on earlier Neve VR-series consoles. The functionality is similar between the two, so I felt right at home with the exquisitely designed 88RS plug-in. It even has the uniquely Neveian quirk of putting some pots upside down, with the center position at six o'clock. I don't have the space here to fully explain all the features, so I'll concentrate on the ones I think set this plug-in apart from your average virtual channel strip.

The expander/gate circuit offers a great amount of control, allowing you to achieve a natural sound quality that most gate plug-ins can't touch. The Hysteresis control, which determines separate dB levels for the opening and closing of the gate, is a major factor in the musicality of the circuit. This control basically eliminates gate chatter, that annoying effect that happens

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FIG. 2: The LA-3A is much simpler than the Neve 88RS but gets the job done nicely.

when the signal is hovering right around your threshold, making the gate open and close rapidly. The rest of the controls are standard for a gate. Although there is no continuously variable attack time, a Fast switch reacts to the signal crossing the threshold in 50 microseconds, as opposed to 500 microseconds.

The compressor/limiter circuit is straightforward, although again you have only binary settings for your attack. The attack time is actually program dependent: the Fast switch shifts how quickly the compressor can react to a speedy transient, from 3 milliseconds down to 1 millisecond.

With 20 dB of cut and boost on each band, and widely overlapping frequency ranges, the EQ section has just about everything you would expect from a 4-band equalizer. However, only the two middle bands are fully parametric. The high and low bands are peak-type filters by default, and each has a Hi-Q switch to decrease the affected bandwidth. A Shelf button on each of the high and low bands allows you to change the shape to a shelving EQ. The two Cut Filters have individual engage switches and sweep from 31.5 Hz to 315 Hz (highpass) and from 18 kHz to 7.5 kHz (lowpass).

In practice, I found the Neve 88RS to be highly versatile and sonically robust. Using it is definitely the closest I've ever felt to manipulating large-format controls in the digital realm. The effectiveness and ease of use of all the sections make it my new first-reach channel strip for just about everything, from vocals and snare drum to trumpet and accordion. The EQ and dynamics are responsive, flexible, and free of undesirable artifacts, and the ability to use them together as a de-esser is a huge bonus. Considering how much each strip does, you can run a decent number of instances (13 mono or 9 stereo per UAD system). You can also free up DSP resources by bypassing individual sections of the strip that you aren't using.

LA-3A

The LA-3A plug-in is a faithful replica of its hardware namesake, which itself is the solid-state (FET) version of the tube-based LA-2A optical compressor. In a sense, the LA-3A is a cross between an LA-2A and the original Universal Audio/UREI compressor, the 1176. With only two knobs, two switches, and a VU meter, this plug-in is the model of functional simplicity—the perfect yin to the multifeatured 88RS's yang (see **Fig. 2**). One knob is for Peak Reduction and the other is for makeup gain. The switches are for changing between limiting and compressing and determining whether the VU meter shows gain reduction or output level. The VU switch also has an off position, which bypasses the effect and frees up DSP.

I used the LA-3A on a myriad of instruments and a number of mixes, and it held its own. A couple of times I opted for the UAD system's LA-2A or 1176LN plug-ins instead, as the sonic flavors are slightly different. For example, on horns and vocals in one song, I liked the LA-2A a little more for its gentler imprint on the sound. I found the LA-3A in this case to be more aggressive sounding, which just didn't work for that particular piece. However, the LA-3A sounded great on another song where I wanted the vocals to be more up front.

On a midtempo rock track, I preferred the LA-3A on the snare drum, because the LA-2A, with its inherently slower attack time, let too much of the transient of the drum through. In a different circumstance where you really need the snare drum to pop, the LA-2A might be preferable, so having both options is certainly nice.

On bass guitar and bass drum, I was surprised to find that I preferred the LA-3A, because the LA-2A has become my standard compressor for low-end instruments. The LA-3A cleared up some muddiness in the low mids but still let the sub energy through nicely.

PRODUCT SUMMARY

UNIVERSAL AUDIO Neve 88RS and LA-3A

UAD plug-ins
Neve 88RS, $299
LA-3A, $149

FEATURES	4
EASE OF USE	4
QUALITY OF SOUNDS	5
VALUE	4

RATING PRODUCTS FROM 1 TO 5

PROS: Both plug-ins sound fantastic and are very straightforward to use. Certain features of the Neve 88RS are unique.

CONS: Only work in conjunction with Universal Audio's UAD system. Dedicated DSP hardware causes inevitable latency.

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Universal Audio
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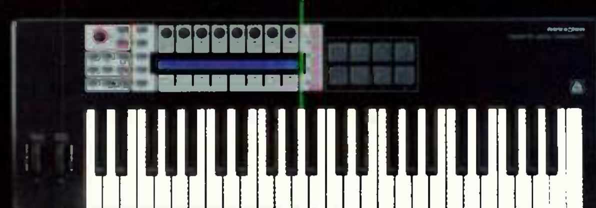

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Comparing the LA-3A to the 1176LN in all of these circumstances, I heard less difference in character, as long as my attack and release settings on the 1176 were in the middle of their range. Both sound great when slammed, giving you that brilliantly energetic overcompressed sound that is often perfect for drum room mics or backing vocals.

Play Your Card Right

If you already own a UAD system, you'll surely want to pick up these two gems at some point. If you don't already own a UAD system, Universal Audio is definitely making it harder to resist with each top-notch plug-in it releases. The company often offers promotional discounts or package deals for its plug-ins (such as the Nevana suite), so keep your eyes peeled.

If you already own a UAD system, you'll surely want to pick up these two gems.

The main drawback is that the DSP host adds a hefty amount of latency, which Digidesign Pro Tools LE doesn't automatically compensate for. The extra routing is a little cumbersome, but it's worth it—the UAD plug-ins have helped me elevate my DAW mixing to a higher level. Although I still prefer to use certain outboard pieces while mixing, Universal Audio has taken yet another step toward providing engineers with a fully professional internal mixing environment.

Eli Crews owns and operates New, Improved Recording (www.newimprovedrecording.com) in Oakland, California.

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FIG. 1: The Gold and Silver bundles both come with a combination dongle and instrument interface called the TonePort DI.

LINE 6 GearBox Plug-In Gold 3.10 (Mac/Win)

Hundreds of amps, effects, and more.

By Geary Yelton

For guitarists and anyone else who records guitar, the selection of gear-modeling software is greater than ever. Tones that for decades were available only to players with an extensive collection of stompboxes, amps, and speaker cabinets are now as accessible as loading a preset from a pop-up menu. Line 6 has long been a master in the field of modeling such hardware, and one of the latest products to take advantage of the company's experience is GearBox Plug-In.

Plug It In

GearBox Plug-In is based on the standalone application GearBox (Mac/Win), which has served as a software front end for Line 6's TonePort USB guitar interfaces since 2005. GearBox Plug-In is available in two bundles, Silver and Gold, which differ in the number of modeling algorithms they include (for more information, see the **online bonus material** at www.emusician.com).

ONLINE BONUS MATERIAL

In Line 6 parlance, a configuration of models and their settings is called a *tone*. Line 6 derived most of GearBox Plug-In's models and tones from the PODxt and Bass PODxt. The Gold bundle furnishes every model Line 6 makes; they're designed for guitar, bass, vocals, and general studio use.

GearBox Plug-In supports AU on the Mac, VST in Windows, and RTAS on both platforms (on the Mac, though, Steinberg Cubase users are out of luck). Because it works like any other effects plug-in, you can completely change your tone after you've recorded a track. You can't do that with the standalone GearBox (which is included with GearBox Plug-In), nor can you do that with a hardware processor without routing a send and return loop from your audio interface. You can also use different instances on different tracks for a variety of tones you can change at will. Another advantage is that plug-ins recall the settings you used for each track. You can save any tones you edit in the standalone GearBox, of course, but the plug-in saves you the step of having to load them manually when you open a sequencer file. You can also use sequencer automation to control plug-in parameters—something you can't do with a standalone program

GearBox Plug-In supports Mac OS X running on PowerPC and Intel processors, and Vista support is available to Windows users. I installed the software on my dual-processor 2.3 GHz Power Mac G5 with 4 GB of RAM

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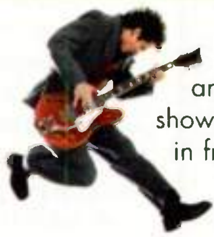

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and Mac OS X 10.4.10 and used it with MOTU Digital Performer 5.12 and Digidesign Pro Tools LE 7.3.

TonePort DI

GearBox Plug-In comes with its own low-latency audio interface, the TonePort DI (see Fig. 1). The TonePort also serves as a copy-protection device that must be plugged into your computer's USB port for the software to work. The front panel offers only the basics—a ¼-inch input, a volume knob, and a button to engage a pad for instruments with active pickups. The rear gives you four ¼-inch jacks—one balanced DI (direct injection) out, two monitor outs, and one headphone out—and a USB port, which powers the device. And because it's such a small but essential unit, a slot on the back panel lets you connect a security cable.

If you'd rather use your own audio interface entirely and forgo the DI, you can do that as long as you leave the DI connected for use as a dongle. If you already own one of Line 6's TonePort or GuitarPort interfaces or a PODxt or Bass PODxt, you can download the plug-in from Line 6's Web site for $199.99. It comes with the same collection of models as the Silver bundle, minus the DI, and represents a pretty substantial savings. If you've previously purchased any Model Packs for your hardware, they are automatically added to the plug-in's collection of models.

FIG. 2: GearBox Plug-In Gold gives you every modeled amp, cabinet, preamp, and effects processor that Line 6 makes. You can use it to process not just guitar and bass, but any audio track.

GearBox GUI

GearBox Plug-In is a slightly scaled-down version of the GearBox application (see Fig. 2). Its straightforward layout displays amplifier controls and a virtual effects pedalboard, along with knobs to control input and output levels and VU meters that track the plug-in's outputs. When you click on a stompbox, its controls appear in the effects control panel. You can select from hundreds of amp, cabinet, and effects models and save any modifications you make as new user tones. My biggest complaint about GearBox Plug-In is that you select models and tones using hierarchical pop-up menus—always a cumbersome technique with any software. You have no means to quickly jump to a selected preset or step through a series of presets, either manually or with Program Changes.

Although the application boasts several features that the plug-in lacks, you can run the standalone GearBox app anytime you need to use the tuner, adjust Hum Reducer parameters, or download and learn one of the hundreds of songs available from GuitarPort Online (a free 30-day membership is included). By the way, Hum Reducer is amazingly handy if your pickups are unshielded, as they are on my Fender Stratocaster.

Although the standalone version of GearBox responds to MIDI Control Changes, the plug-in version does not. Fortunately, it does respond to sequencer automation, but that doesn't help much if you want to control the volume or wah-wah pedal effects in real time.

Attractive Models

With every model in Line 6's bag of tricks, the Gold bundle furnishes quite a complete collection of virtual gear. Whether your taste in guitar amps leans toward Marshall, Fender, Vox, Hiwatt, Orange, or Line 6, GearBox Plug-In aims to please. Bass amps range from Ampeg and Fender to Eden and Gallien-Krueger. Emulations of guitar and bass cabinets made by all the same manufacturers are in plentiful supply, giving you everything from a single 6-inch to four 15-inch speakers. If you want to simulate a direct connection from your virtual amp head, just select No Cabinet. You can also switch off the amp modeling if you want to use only the effects, or you can use a mic preamp in place of an amp head.

The effects control panel contains ten categories, each devoted to a particular family of effects. Clicking on a slot reveals the controls for the chosen effects category, six

PRODUCT SUMMARY

LINE 6 GearBox Plug-In Gold 3.10

effects modeling bundle
$499.99

FEATURES	4
EASE OF USE	4
QUALITY OF SOUNDS	5
VALUE	3

RATING PRODUCTS FROM 1 TO 5

PROS: Hundreds of awesome sounds. Intuitive GUI. Extreme versatility. Low-latency audio interface. Includes standalone version with excellent additional features.

CONS: Hierarchical menu selection. No real-time control of wah and volume pedals. No Mac VST support. Fixed order of effects.

MANUFACTURER
Line 6
www.line6.com

World Radio History

of them with a pop-up menu for selecting presets (Gate, Volume, Compressor, and EQ have no presets). The Delay category delivers emulations ranging from an Electro-Harmonix Deluxe Memory Man analog delay to a Maestro EP-1 Echoplex tape delay. Mod effects include flangers, choruses, phase-shifters, rotary speakers, and the like. The Stomp category provides fuzz, distortion, ring mod, auto-wah, and quite a few unique effects that defy easy categorization. The Cab/ER category displays an image of a speaker cabinet and a microphone; you can click-and-drag the mic to determine its perceived distance from the cabinet and select from different mics for the guitar and bass cabinets.

Like other Line 6 products, GearBox Plug-In limits your ability to rearrange the order of effects. You can place the volume pedal, modulation, delay, and reverb after the amplifier using their Pre/Post switches, but otherwise it's strictly gate, volume pedal, wah, stomp, modulation, delay, reverb, cabinet/early reflection, compression, and EQ—in that order. Such rigidity does preclude some types of experimentation, but if you are experimentally minded, it's likely you have additional plug-ins that you can insert wherever you need them. Line 6 says that such limitations ensure compatibility with the PODxt's large library of tones and models.

Lest you think GuitarBox Plug-In is only for guitar and bass, many excellent presets are intended for processing vocals and even drums and other instruments. Six models are based on preamps from the likes of API, Neve, and Avalon, and 24 tones are specifically for vocals.

All Plugged In

As a plug-in as well as an application, GearBox now competes directly with similar native software such as IK Multimedia AmpliTube 2, Native Instruments Guitar Rig 3, Waves GTR, and Alien Connections ReValver Mk II. Not surprisingly, each has its strengths and weaknesses. Among GearBox Plug-In's strengths is its huge collection of awesome presets, many of them dead-on simulations of the setups used for particular hit songs. For processing all kinds of tracks—not just guitar and bass—GearBox Plug-In is one of the most versatile and useful multi-effects plug-ins I've ever used. Whether you're already a Line 6 convert, a guitarist who records with a computer, or a recordist who just wants a large repertoire of killer effects, GearBox Plug-In Gold is calling your name.

EM associate editor Geary Yelton has been playing guitar since high school and bass guitar since college. His virtual studio, Rea Road Tracks, is in Charlotte, North Carolina.

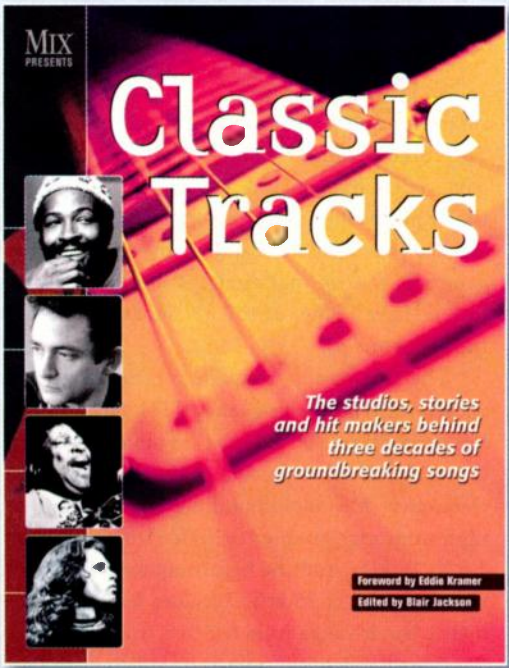

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FIG. 1: The Portico 5033's front panel is replete with continuously variable rotary controls and backlit push-button switches.

RUPERT NEVE DESIGNS Portico 5033

See Product Specs @emusician.com

Big tone in a tiny package.

By Michael Cooper

The 5033 single-channel, 5-band equalizer is the latest addition to Rupert Neve Designs' quickly expanding Portico line of pro-audio gear. The product line comprises highly compact and portable signal processors that can be interconnected to form an integrated system with much of the functionality of a large-format, modular console. That said, individual units are quite happy to be "home alone."

The analog (solid-state) 5033 features one band each of high and low shelving filters and three bands of fully parametric filters. Proprietary input and output transformers designed by the legendary Rupert Neve lend the 5033 a subtle yet distinctive timbre.

On the Face of It

The ½U 5033 packs a lot onto its steel front panel: 14 rotary controls and 5 push-button switches (see Fig. 1). All of the rotary controls are continuously variable, and the switches are backlit by LEDs when their related functions are active.

The 5033's two shelving bands each feature separate rotary controls for gain and corner-frequency selection. The three parametric bands each sport independent gain, center-frequency, and Q controls (the last adjusts the respective filter's bandwidth and slope), all of which are also rotary. All five bands' gain controls, as well as a gain-trim control that directly follows the input transformer, have a range of ±12 dB and are detented at 0 dB (no boost or cut).

The five bands' frequency ranges overlap fairly widely, with the low-frequency (LF) band's range extending from 30 to 300 Hz, the low-midrange-frequency (LMF) band covering 50 to 400 Hz, the midrange-frequency (MF) band handling 330 to 2,500 Hz, the high-midrange-frequency (HMF) band delegated to 1.8 to 16 kHz, and the high-frequency (HF) band taking care of 2.5 to 25 kHz. Q values for each of the parametric bands range from 0.7 to 5 (roughly 2 octaves to ⅓-octave bandwidth), which is sufficient to handle all tone-shaping tasks you're likely to encounter.

Separate bypass switches are provided for each of the three parametric bands, whereas the low and high shelving bands can only be switched in and out of circuit together. A separate all-bypass switch takes all of

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REVI

the filters out of circuit but leaves the trim control and transformers in the audio path. This design allows you to run audio through the 5033's flattering transformers without applying EQ, while also adjusting the input-trim control to avoid clipping and to optimize levels downstream of the 5033. (The unit has no output-gain control for such purposes.) But unfortunately, keeping the input trim always in circuit also makes for difficult A/B comparisons of audio that's been EQ'd versus that which is flat (where all filters are bypassed). That's because the always-active input-trim control cannot be used to compensate for the five filters' net gain boost or cut without also changing net gain in the all-bypass state. In other words, there's no way to instantaneously match output levels for EQ'd and flat setups with this design.

If portability and a small footprint are important to you, this is a great choice.

Bringing Up the Rear

The 5033's spartan rear panel sports XLR connectors for transformer-balanced, line-level audio I/O, as well as a switch and coaxial jack for power (the latter serving a lump-in-the-line power transformer and 2-pin AC connector), and two Buss output jacks (see **Fig. 2**). The Buss jacks are ¼-inch TRS and wired in parallel; they provide high-impedance feeds to other Portico gear equipped with Mix or Buss inputs, to fashion a modular-console assembly as mentioned earlier. (Suggested block and system diagrams for such setups will soon be available at www.rupertneve.com.)

If field recording is your thing, you can power a daisy-chained arrangement of Portico units from a single 12V car battery. Multiple Portico units can also be racked in a number of ways, including horizontally or in an optional vertical rack kit.

To the Test

I got very good to excellent results with every application of the 5033, although I didn't always find the unit to be entirely user friendly. On a thin, peaky-sounding lead vocal track, boosting the LF band several dB in the upper bass and the MF band around 800 Hz warmed up and filled out the track nicely. Moderate boost in the HF band added a smooth, flattering sheen to the sound. If I'm being nebulous about the specific frequencies I boosted in the LF and HF bands, it's because I really don't know—due to the crowded control layout, only one intermediate frequency setting is noted on the front panel for each band. That is, only the noon position and full clockwise and counterclockwise settings are titled. Likewise, only three settings for each gain and Q control are noted. That's all understandable considering the lack of available real estate, but the owner's manual provides no additional documentation. The company is aware of the oversight and says it will have many intermediate settings documented on its Web site by the time you read this.

On electric bass guitar, boosting the LF band below about 80 Hz added some wonderful thunder. The LF filter sounded quite tight for analog EQ. But I missed having a lowpass filter to steeply roll off fret noise and a highpass filter to eliminate thumping on the track.

On a somewhat cardboardy-sounding kick drum, the 5033 fleshed out the sound nicely when I cut a few dB at around 600 Hz and boosted at 30 Hz. In the same song, I could thin out the kick and toms in a stereo track for the overhead mics (using two 5033s, one for the left channel and the other for the right) by applying generous amounts of shelving cut at around 150 Hz. To test how well the 5033's HF filter section sounded (always the acid test for equalizers), I applied around 7 dB boost at the 3 o'clock position in that band on the overheads. I was not disappointed. The high end sounded silvery-sweet and smooth, with no audible ringing or edginess.

Next up was a problematic electric guitar track. Even miking with a Royer R-122 ribbon mic patched through a Universal Audio 2-610 tube preamp and LA-2A tube compressor, the sound of this player's nasty transistorized rig remained somewhat harsh. A bell-curve cut of a few dB centered at 6 kHz, with the 5033's Q set to its broadest setting (0.7), smoothed the track's sound beautifully.

On a rock mix that needed no 2-bus EQ, I nevertheless found that patching my mixer's stereo bus through a pair of 5033s in all-bypass mode sweetened the sound. The 5033s' transformers softened the highs slightly, moderating any digital edginess.

Play It by Ear

Experienced engineers who can listen to audio and readily identify specific frequencies needing treatment might be a bit frustrated by the dearth of titling for the 5033's intermediate frequency settings (especially

PRODUCT SUMMARY

RUPERT NEVE DESIGNS Portico 5033

5-band EQ
$1,795

FEATURES	4
EASE OF USE	3
AUDIO QUALITY	4
VALUE	3

RATING PRODUCTS FROM 1 TO 5

PROS: High-end sound quality. Five full-function EQ bands. Transformers can be used to sweeten the sound even with all filters bypassed. Frequency ranges for each band overlap. Highly portable.

CONS: No highpass or lowpass filters. Control layout is crowded. Few intermediate control settings are marked. Circuit topology makes unbiased A/B comparisons difficult. Pricey.

MANUFACTURER
Rupert Neve Designs
www.rupertneve.com

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FIG. 2: Buss jacks on the 5033's rear panel facilitate interconnection with other units in the company's Portico line.

when dealing with the ultracritical low end). But once it's dialed in, the 5033 sounds great.

If portability and a small footprint—without the loss of audio quality—are of paramount importance to you, this unit is a great choice. Considering that the 5033's list price per channel is comparable to that of other high-end equalizers with more-spacious control layouts, however, studio owners with room to spare will likely be a little put off by the 5033's crowded front panel. That said, most competing models feature fewer bands per channel compared with the 5033. If you're looking for high-end sound in a tiny package and are willing to accept the attendant ergonomic compromises, check out this flattering equalizer.

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

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QUICK PICKS

SUBMERSIBLE MUSIC

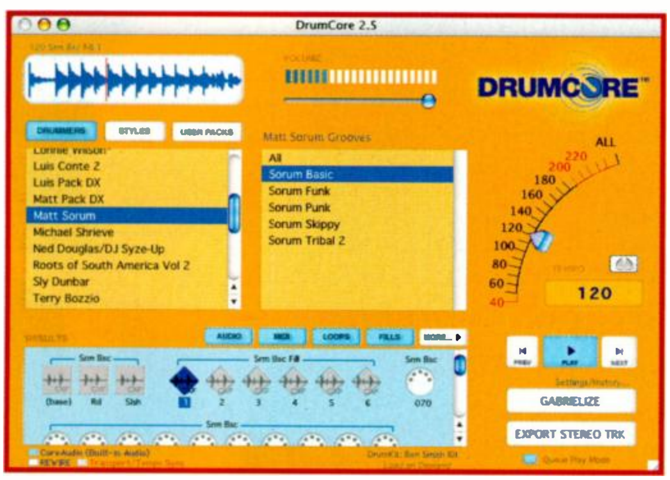
DrumCore 2.5 (Mac/Win)

By Mike Levine

The latest version of Submersible Music's DrumCore ($249, upgrade pricing also available) features some significant improvements, though none are earth-shaking. For those unfamiliar with the product, DrumCore is a multipurpose drum application that's designed to help you quickly and easily assemble drum parts. It hosts a large and expandable collection of drum loops (both audio and MIDI) that were played by a group of top pros and are organized into GrooveSets, which include basic beats and fills designed to work together in the same song. The loops can be exported (and even dragged-and-dropped) into your recording application or synced using ReWire at a wide range of tempos. DrumCore supports all the major digital audio sequencers.

The software also offers a MIDI drum module with kits for each of the DrumCore drummers made up of samples from their loop-recording sessions. It lets you import WAV, AIFF, REX2, Acid, Sound Designer II, and MIDI files and allows you to organize all your drum loops in a central database. (For more on DrumCore's legacy features, see the review in the February 2007 issue of EM, available at www.emusician.com.)

The improvements in DrumCore 2.5 make this stellar program even stronger.



The Latest

Many of the new features center around the application's MIDI drum module. One of the coolest is called Pad Swapping. There are 48 individual "pads" in a given kit—24 in the drum section and 24 in the percussion section—and now every component sound in each kit is interchangeable with its counterparts from different kits. That means you can move, say, Matt Sorum's snare drum into Zoro's kit, or make a kit with Michael Shrieve's kick, Sly Dunbar's snare, Terry Bozzio's China cymbals, Alan White's toms, and Luis Conte's guiro (see **Web Clip 1**).

Also new is drag-and-drop sample support within the MIDI Drumkit Editor window. This makes it possible to quickly set up a kit using your own or third-party samples. Just like the factory kits, each of the 48 pads in a user kit can have unlimited Velocity-switched layers. Setting the Velocity thresholds between samples is as easy as dragging a slider. I've always found multisampled layered sounds in drum samplers to be complicated and difficult to tweak, but in DrumCore it's a snap. Another new feature is the ability to hear the changes you make during a pad edit in real time, as you make them.

Version 2.5 also marks the debut of DrumCore as a standalone MIDI drum module. This will be particularly useful for those who want to use DrumCore for live performance work. You won't have to open it as a ReWire slave within a sequencer host. That saves a lot of potential hassle and signal-flow troubleshooting onstage.

Queuing Up

From DrumCore's main window, you can audition loops either in ReWire or standalone mode. Now with the new Queue Play mode, when you select a new pattern while another is playing, it waits until that previous pattern finishes and then continues playing the new pattern in time. You can also seamlessly switch between audio and MIDI loops, should you so desire.

Submersible offers a range of DrummerPacks ($79.99 each), which include additional content from many of the DrumCore drummers. Also available is DrumCore Deluxe ($670), which comes with nine DrummerPacks in addition to the basic collection and the DrumCore application. It's a lot more money, but you get a huge amount of very good content.

By the time you read this, Submersible should have released KitCore ($49.99, download only), an AU/VSTi/RTAS plug-in that contains the MIDI drum module (but not the audio loops) from DrumCore. KitCore Deluxe ($99), which contains all the kits from DrumCore Deluxe, will be out soon.

Core Values

DrumCore 2.5 is easy to use and compatible with most sequencers, and it offers a ton of great drum performances. Submersible makes an ongoing effort to improve the product, and the company seems attentive to user feedback. The only real limitation I find with DrumCore is that it only offers stereo audio loops. There's no multitrack option, so you have no choice but to use the mixes provided. Although they are uniformly quite good, there are times when a song calls for a different amount of kick or snare, or more processing on an individual element. No doubt, multitrack capabilities would make the product more complicated (not to mention expensive and disk space consuming), but perhaps down the road Submersible can implement such a feature.

That said, I've been using (and reviewing) DrumCore since its initial release, and it's definitely my go-to application for drum-track creation. This latest update only makes it better.

EM WEB CLIPS

Value (1 through 5): 4
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TELEFUNKEN USA

RM-5C

By Eli Crews

Designed by Silvia Classics and branded and distributed by Telefunken USA, the RM-5C cardioid ribbon microphone ($1,495) is a modern take on the classic shape of the RCA BK-5. The RM-5C has a striking, textured black finish and art deco grille housing, with arrows pointing to the business end to indicate that this is a front-address mic. (Anyone familiar with the RCA BK-5 will not be surprised by this.) However, the fact that it isn't a side-address mic with a figure-8 pattern is only one of the ways in which the RM-5C differs from your typical ribbon transducer.

Polar Opposites

I used the RM-5C beside four other ribbon microphones as well as a few dynamic and condenser mics. I auditioned it on vocals, electric and acoustic guitar, electric and upright bass, saxophone, accordion, and in various positions on drums. The mic comes with a leatherette pouch and large, spider shockmount, all within a wooden case.

Compared with other ribbon microphones I own, the RM-5C has unique, unribbon-like characteristics, such as an aggressive upper midrange and a less substantial low end. Because my RCA 77DX can be switched into cardioid mode, I did a number of tests comparing the directionality and timbre of the two mics to get my bearings, though it's not surprising that they sounded completely different. The RM-5C achieves a tighter directionality than the 77DX, picking up significantly less ambient room sound when close to the source. Additionally, the two mics have complementary frequency responses, with the 77DX accentuating the lows and low mids and the RM-5C heavy on the high mids and highs.

The differences were just as noticeable when I compared the RM-5C with Royer and Coles ribbon mics, both of which offer only figure-8 patterns. Because they had such different tonal characteristics, I often found myself leaving both the RM-5C and my reference ribbon mic up and blending the two for the best sound. This worked really well on an electric bass cabinet, where the RM-5C accentuated the midrange of the picked bass while the other mic filled in the low end nicely.

The RM-5C is a ribbon mic with a sound all its own.

In Action

The standout application was on upright bass, where the RM-5C sounded amazing. Situated about a foot and a half in front and a couple of inches above the bridge, the mic gave me a full, clear, rich tone, without the boomy bottom of many other mics in this position. The balance between the woody thump and the articulation of the fingerboard was about as good as one mic alone can capture.

The RM-5C also sounded good on acoustic guitar. I placed the mic about two feet away from the instrument, pointing between the 12th fret and the sound hole. Again, the combination of woody tone and pick articulation was very balanced. However, on a different guitar on a different day, I needed to blend the RM-5C with a small-diaphragm pencil condenser to fill out the sound. Likewise, on male vocals (and similarly on piano) the RM-5C felt a tad thin, but when blended with another, beefier mic, it gave me good definition and presence. As for electric guitar, I got excellent results using the RM-5C on small, tube combo amps, whereas on louder rock rigs, the mic imparted an extra fuzziness I didn't much care for.

On reeds, I got mixed results. On alto saxophone, the RM-5C didn't fare well: there was a nasal quality to the mic that was unflattering to the instrument's tone, as if I had cranked up 1,200 Hz on an EQ. I liked the RM-5C quite a bit on accordion, though; it really grabbed the reediness and growl of the instrument in a pleasing way.

The two best drum applications I found for the RM-5C were as a secondary bass drum mic to add some presence (positioned a few feet back and with a pop filter, of course), and as a room mic pointed away from the kit at the back wall. In general I wasn't happy about the way the RM-5C picked up the metal of the kit: it gave the cymbals and hi-hats an aggressive midrange that needed some subtractive EQ in the area of 1 to 3 kHz.

A Unique Voice

Although the RM-5C isn't good at everything—few mics are—the things that it *is* good at make it arguably worth having in your collection. The upright bass sound I captured with it alone makes me want to keep an RM-5C around.

The RM-5C wouldn't be my first choice for a single ribbon mic to own, especially at an introductory price of $1,495, because it's not quite as utilitarian as other ribbon mics. Nonetheless, it has a very interesting character, and I was able to get some great sounds with it. If variety in acoustic flavors is what you're after, give the RM-5C a taste.

Value (1 through 5): 4
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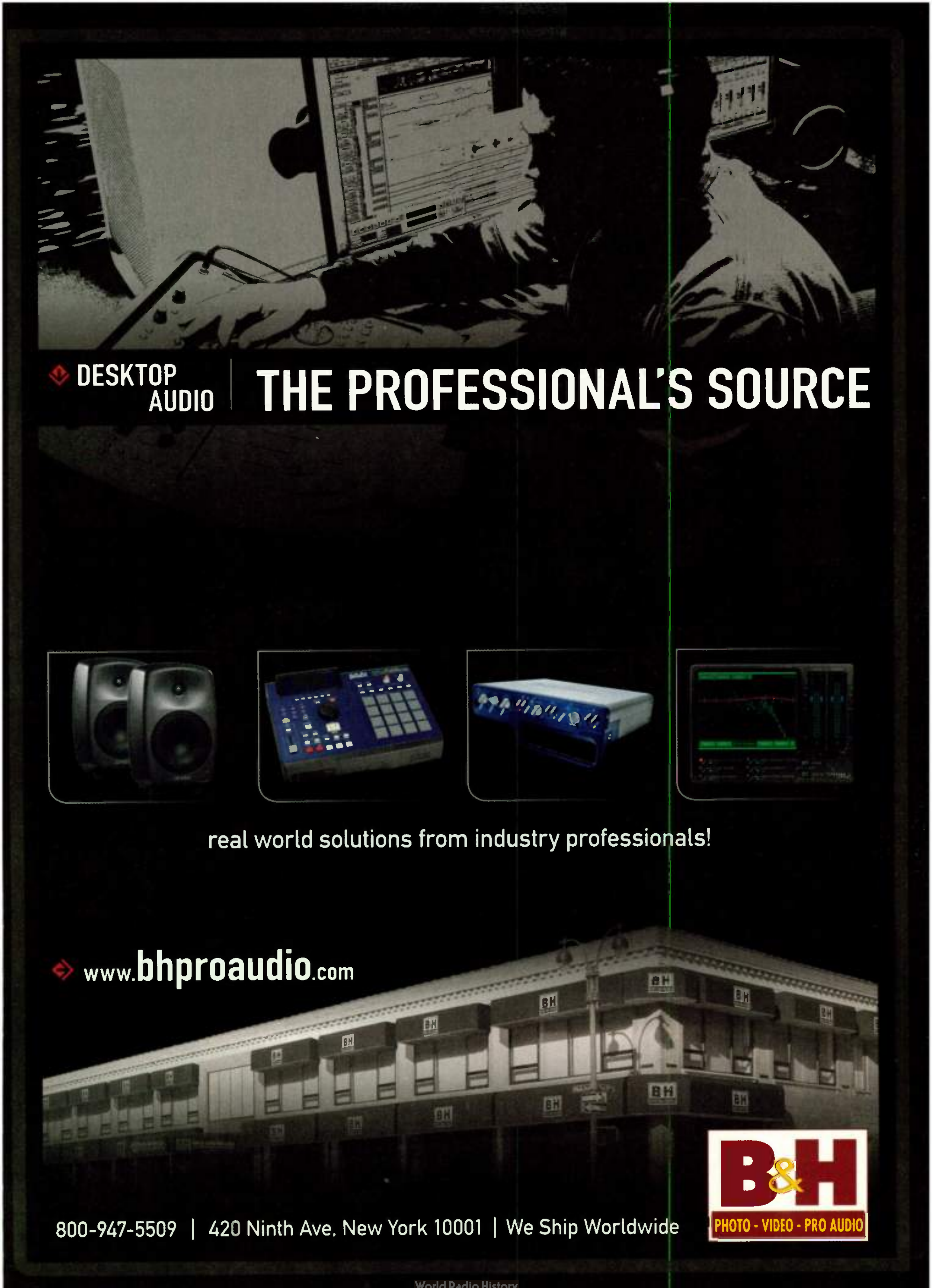

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TERRATEC

Axon AX 50 USB

By Orren Merton

TerraTec's Axon AX 50 USB guitar-to-MIDI converter ($549) is the company's second product to be based on Axon's neural-network-detection algorithms. It uses the same conversion technology as the Axon AX 100 mkII, which I reviewed in the March 2007 issue of EM (available online at www.emusician.com), and ships with a similar software editor.

Because the guitar-to-MIDI conversion technology was covered in the last review, I'll focus on the specific features of the Axon AX 50 USB here. You can also find a detailed description of the software editor in the March 2007 review.

Half Pint

The Axon AX 50 USB forgoes having editing buttons on the box for a sleek look and a half-rackspace footprint. The front panel of the unit includes a 13-pin input that can accept the output from any 13-pin compatible magnetic or piezo hexaphonic pickup. The front panel features a 3-digit, 7-segment LED window that doubles as the tuner and patch-number display. Next to the LED display is a tiny button that does triple duty: it activates the integrated tuner, adjusts the LED brightness, and prepares the device for firmware updates. The front panel also contains a large power button.

The rear panel is equally sparse. There is an input for the included 12V wall wart (which, by the way, is huge, even by wall-wart standards), an output for the normal magnetic pickup signal of the guitar, and MIDI In/Out/Thru jacks. Last but certainly not least, the middle of the rear panel features the centerpiece of the entire device: the USB jack for connection to a Mac or PC.

Direct Connect

One of the most glaring omissions in the Axon AX 100 mkII was a direct USB connection to a computer. The Axon AX 50 USB not only transfers MIDI data to and from the computer over a single USB cable, but it also uses the built-in USB MIDI support in Mac OS X and Windows XP, eliminating MIDI driver issues and making this a truly plug-and-play device. I plugged the unit into my Mac Pro, and all my software was able to see it. Perfect.

The software editor included with the Axon AX 50 USB features the Axon AX 100 mkII's unrivaled string, fret, and pick split functionality and parameters, with the added bonus that you can configure different splits not only by MIDI channel but by USB port as well. The Axon AX 50 USB appears to your computer as five separate USB MIDI devices. This gives you the flexibility to route your various splits to different applications or to different tracks within one application.

The Axon AX 50 USB software editor also differs from its big brother in that it doesn't include the latter's arpeggiator, step sequencer, and Hold-mode functions, due to the AX 50's lack of a momentary switch input. Considering that most software instruments and DAWs have their own arpeggiators and step sequencers, I didn't miss this functionality within the Axon editor.

Sound Improvement

One of the weak spots of the AX 100 mkII was its bland complement of internal GM wavetable sounds. The Axon AX 50 USB includes no internal sounds, instead shipping with Wave XTable VI, a sample-based virtual instrument that runs in Native Instruments' Kontakt Player 2. It contains nearly 900 MB of GM-compatible samples divided into 128 single instruments and 9 drum and sound-effects kits. All the sounds in Wave XTable VI are quite usable, high-quality samples. This is a vast improvement over the internal sounds in the wavetable chip inside the AX 100 mkII.

You can run Wave XTable VI standalone, utilizing the Axon AX 50 USB/Wave XTable VI combo as a true computer-based guitar synthesizer, or you can instantiate Kontakt Player 2 inside your host software. You can also load the Wave XTable library into a full version of Kontakt 2 for more-serious editing and sound design.

MIDI Me

It took a bit of work to optimize the Velocity sensitivity of the Axon to my playing style, but once I did, I found the Axon's speed and accuracy a dream. I used my Koll Tornado, which is equipped with a Ghost Hexpander 13-pin pickup, to play both Wave XTable VI standalone and software instruments in Apple Logic Pro, and what I played was always what I heard (see **Web Clip 1**).

I do wish that the sensitivity sliders in the software editor offered meters to aid in calibration, and that the manual provided more detail. But overall, guitarists interested in using their main instrument to control software instruments owe it to themselves to give the Axon AX 50 USB a serious look.

EMWEB CLIPS

Value (1 through 5): 4
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The Axon AX 50 USB is a guitar-to-MIDI converter that transmits MIDI data directly to your computer over a USB cable.

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CASCADE MICROPHONES

Fat Head II

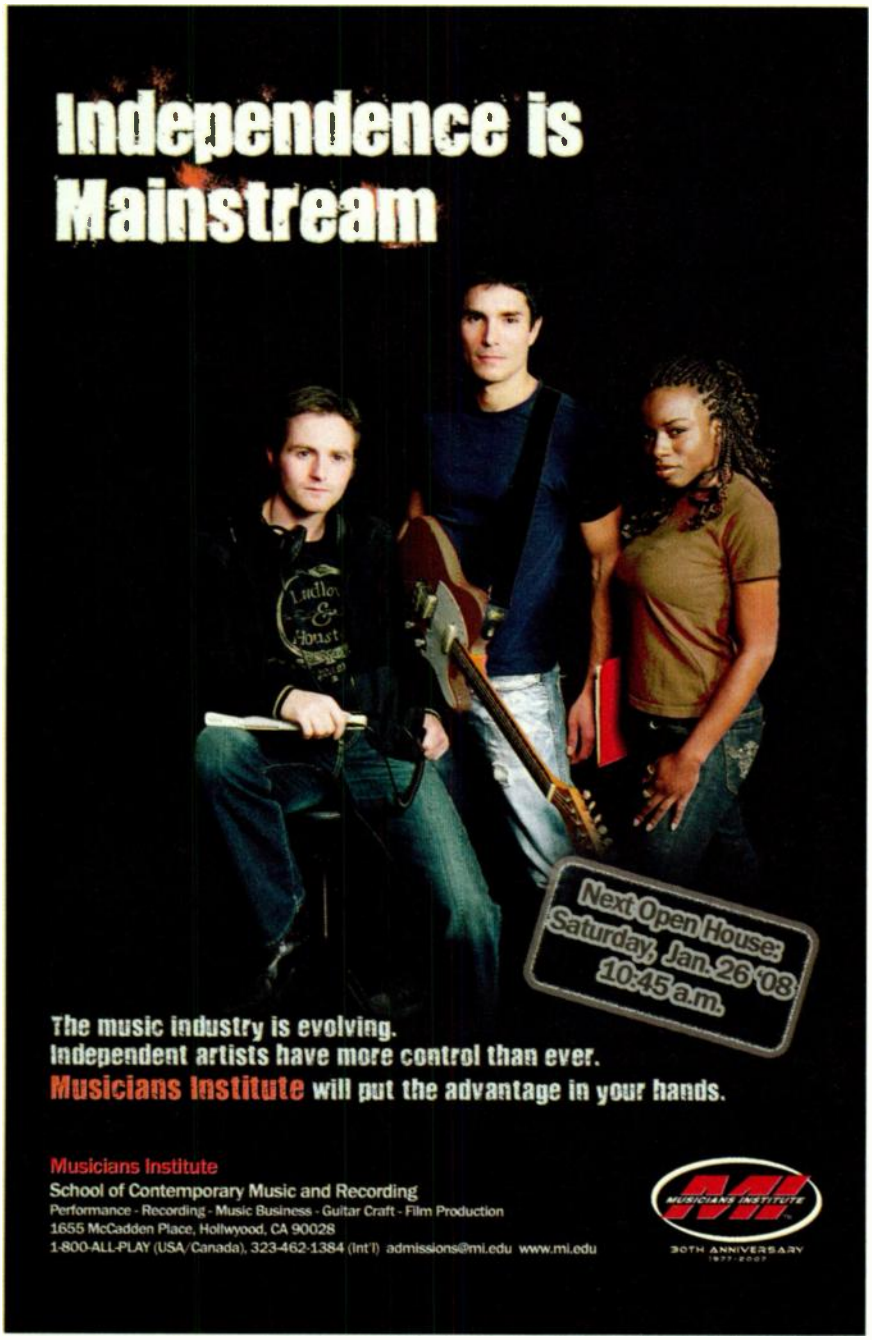
By Rudy Trubitt

Cascade Microphones first came to my attention last year through an email newsletter ad that offered a Fat Head II ribbon mic at a price so low, I ordered two. However, when I was assigned this review, Cascade sent a brand-new pair, because some of the manufacturing details had changed since I ordered my first set.

The Fat Head II ($199) is bigger than it looks in the photo. The barrel (available in your choice of silver or black) is about one inch in diameter, and the lollipop screened capsule is about two inches around.

Considering the price, Cascade doesn't skimp on the packaging. The mics arrived in individual foam-lined briefcases, with a wooden box inside and a large shock-mount mic clip. This is a step up from the original models, which came without the wooden box and included a slightly less-sturdy clip.

But there's a more important difference between the two versions that goes beyond the box: the earlier Fat Head IIs used an offset ribbon design that yielded a slightly different tone front to back. Some people appreciate this 2-tone option, but I prefer the current model, which provides a true symmetrical figure-8 response, making it better suited to stereo recording applications.



Out and About

The Fat Head II is appropriately named: it provides a warm, plump sound, with a smooth rolloff starting around 8 kHz that finishes around 15 kHz. In addition, the mic seems to thicken the upper bass frequencies, a coloration I often found myself EQ'ing out after the fact.

Typically, ribbon mics sound great on horns and guitar amps, and that's certainly true for the Fat Head II. A colleague and I tried the mics on sax, bass clarinet, acoustic guitar, an electric guitar amp, vocals, and drums. I also did some side-by-side simultaneous recordings comparing a Fat Head II and a new Shure SM57, because most EM readers would find an SM57 a familiar point of reference (see **Web Clips 1 through 3**).

For example, on a distorted electric guitar amp, with the mics on-axis at the center of the cone, the Fat Head II offered a thick, smooth sound that was less present and edgy than the SM57's. On a steel-string guitar where the SM57 highlighted my fingernail attack, the Fat Head II pleasingly smoothed the transients.

On some sources, such as acoustic and electric guitar, as well as my singing voice, I typically followed up a Fat Head II track with a good dose of EQ, a mild dip around 200 Hz and a somewhat stronger boost at 8 to 12 kHz. The Fat Head II recordings responded well to EQ: boosting the upper frequencies added detail without getting edgy or sibilant.

An important point to remember about directional mics (and those with a figure-8 pattern in particular) is that they are prone to the proximity effect, which increases bass response as the

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mic moves closer to the source. The combination of the Fat Head II's dark tonality and strong proximity effect can be a double whammy. Unless you're looking for a superfat sound, I'd avoid using a Fat Head II for extreme close-miking applications.

We also did a number of stereo recordings using a pair of Fat Head IIs in a Blumlein configuration. This cross-pair, coincident stereo technique requires the capsules to nearly touch but to be rotated 90 degrees from each other. (Cascade offers an optional stereo bar to simplify this setup.) We placed our crossed pair at the center of the bass clarinet quartet Edmund Wells, who set up in a squared circle. Supplemented with an assortment of condenser room mics, the Fat Head IIs provided a present, detailed recording without a hyped sound quality.

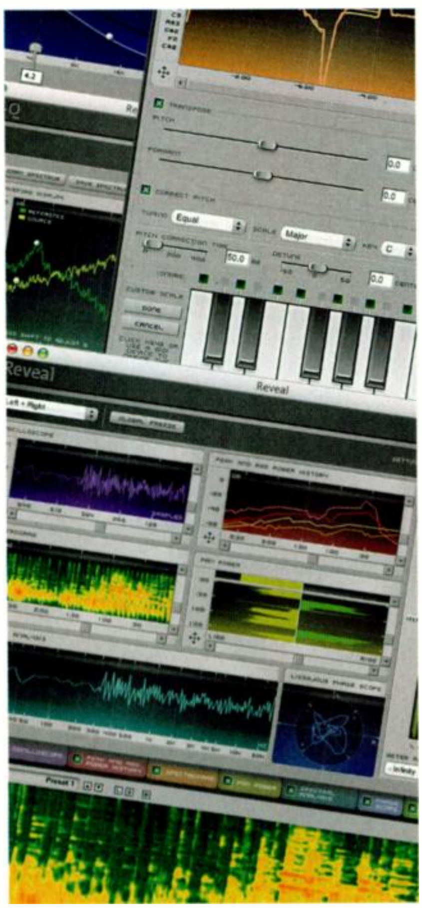
You can also take advantage of the mic's figure-8 null points. I set up a 90-degree crossed pair of Fat Head IIs in the horizontal axis to simultaneously record my vocal and a 12-string Guild acoustic. The two tracks had quite good isolation. What leakage remained was phase coherent between channels, as the capsules were physically aligned. That meant I could pan the guitar and vocal tracks to center without any undesirable coloration, but then process them individually, almost as if they had been tracked individually. This is a great way to cut a live guitar/vocal performance but still retain good flexibility during mixdown.

Dynamic Duo

The Fat Head II has a distinctive tonal character that is readily amenable to EQ, although its output is not particularly hot. However, I found myself using it on loud instruments—drum overheads, horns, guitar amps—and I didn't find its output level to be much of an issue. Overall, the Fat Head II is a good value and well worth considering if you're in the market for an inexpensive, all-around ribbon mic. **EM**

The Fat Head II has a symmetrical figure-8 response, making it great for stereo recording.

Value (1 through 5): 3
Cascade Microphones
www.cascademicrophones.com



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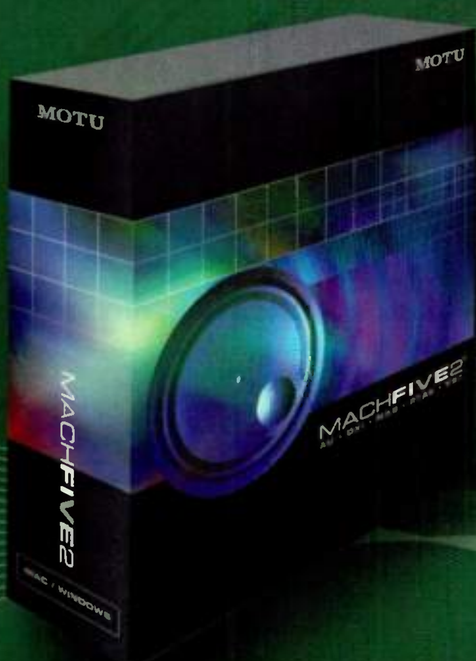
The all-new, all-in-one Apple iMac packs a complete, high-performance computer into a beautifully thin design. Available in 20- and 24-inch widescreen models, iMac is striking to behold, with its anodized aluminum frame and glass cover. The glossy widescreen display makes Digital Performer, MachFive 2 and your other music production software tools come alive with rich color. Now iMac offers greater performance than ever thanks to faster dual-core Intel processors and a speedier frontside bus. iMac features the Intel Core 2 Duo processor at speeds up to 2.4GHz. And for maximum power, you can upgrade the 24-inch iMac to an Intel Core 2 Extreme processor at 2.8GHz. With their 64-bit processor architecture, the Intel Core 2 processors can manipulate data and execute instructions in large chunks to deliver advanced computational power to iMac. That adds up to exceptional speed and agility, improved performance you'll notice when working with large multi-track Digital Performer sessions running all of your favorite plug-ins and virtual instruments. Augmenting the advanced Core 2 processors is an equally advanced system architecture, which includes an 800MHz system bus and 4MB of shared L2 cache. With such substantial L2 cache, data instructions can be kept close to the two processor cores, greatly increasing performance and allowing the entire system to work more efficiently. Your MOTU desktop studio is now more stylish, powerful and affordable than ever.



32GB of instruments and loops

MOTU MachFive 2 is a universal sampler that includes a whopping 32GB of instrument sounds and loops to get you laying down tracks in no time. Got libraries of your own? MachFive 2 reads all major sampler library and audio file formats directly, with no time-consuming importing or conversions — even advanced Giga libraries with multi-layer "dimension" presets. MachFive 2 delivers tons of advanced features, including unlimited parts, pop-out full-screen editors, Loop Lab beat-slice editing, integrated synthesis and much more.

Waves native processing

Waves has long been synonymous with quality plug-ins, and the Waves Platinum Bundle contains a huge range of top-quality Waves processing for your DP5 studio. The Platinum Bundle now includes Waves Tune LT, L3 Ultramaximizer, and IR-L Convolution Reverb as well as all the plug-ins found in the Waves Gold and Masters bundles. Platinum brings extraordinary signal processing power to DP5, for tracking, mixing, mastering, and sound design. From dynamics processing, equalization, and reverb to pitch correction, spatial imaging, and beyond, Waves Platinum Bundle is a must-have for every MOTU studio.

The MOTU experts at Sweetwater can build the perfect MOTU studio for you. We'll help you select the right components, and we can even install, configure and test the entire system for you. Why shop anywhere else?

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Acclaimed instruments from EastWest/Quantum Leap

An essential addition to your MOTU studio are the six new PLAY-powered Virtual Instruments just released by critically acclaimed sound developer EASTWEST/QUANTUM LEAP.

All of these Virtual Instruments include the EASTWEST PLAY advanced sample engine, with 64/32-bit support, easy to use browser, user controllable articulation window that allows users to build custom key-switches, built-in scripts, legato sensing, high quality convolution reverb (including 29 impulses), unique effects such as ADT (Automatic Double Tracking), and a high quality resampling engine.

There are too many other features unique to each product to cover here, so we urge you to go online and check out the video tutorials and audio demos.

- **SD2**
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- **FAB FOUR**
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- **GYPSY**
  Extremely detailed Gypsy style instruments essential for film, tv and game composers
- **QUANTUM LEAP PIANOS**
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Komplete control

For DP5 users who want it all: Native Instruments KOMPLETE 5 is a first-class collection of 11 groundbreaking instruments, including the all-new KONTAKT 3 and GUITAR RIG 3 Software Edition. You also get the award-winning MASSIVE, ABSYNTH 4, BATTERY 3, FM8, REAKTOR 5, B4II, AKOUSTIK PIANO, ELEKTRIK PIANO, and PRO-53. Add KORE 2 — the Super Instrument — and you have perfect synergy with KOMPLETE 5: Use the Sound Browser in KORE 2 to access any of the 7,500 presets from KOMPLETE 5 within seconds. Open the sound and instantly tweak it analog-style using the KORE 2 hardware controller. Access and tweak your software instruments like never before. KORE 2 and KOMPLETE 5 — an infinite universe of sound at your fingertips.

Vintage EQ/Compression

The Focusrite Liquid Mix is another Focusrite first and a true one-of-a-kind. Based on the award-winning Liquid Technology, the Liquid Mix provides 32 channels of simultaneous DSP powered vintage and modern EQ and Compression plug-ins into your DP5 mix without affecting your host computer's CPU. 40 classic compressors and 20 timeless EQs are included. Each EQ and Compressor emulation is painstakingly created though a process called Dynamic Convolution which is a huge step beyond modeling. Through Dynamic Convolution, every frequency at every possible combination of settings is perfectly sampled. That means that you get the true sound and feel of a vintage or modern classic. Tens of thousands of dollars of gear is now right at your fingertips! Think how great your Digital Performer tracks will sound with the Liquid Mix.

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Complete control

The M-Audio Axiom 25 represents an entire line of advanced Axiom MIDI controllers that are perfect for Digital Performer and your MOTU virtual instruments. Built around a rugged chassis, the Axiom 25 includes 25 semi-weighted velocity-sensitive keys with assignable aftertouch, eight MIDI trigger pads, six reassignable transport buttons and 20 non-volatile memory locations. Eight endless rotary encoder knobs let you get your hands on MachFive 2 parameters, Digital Performer's Mixing Board and more. Virtually everything is freely MIDI-assignable, and the backlit LCD screen makes programming easy and intuitive. If you need more keys and controller options, the 49-key Axiom 49 and the 61-key Axiom 61 complete the line.

Advanced waveform editing

Your DP mastering and processing lab awaits you: BIAS Peak Pro 5 delivers award winning editing and sound design tools, plus the world's very best native mastering solution for Mac OS X. With advanced playlisting. Superb final-stage processing. Disc burning. Plus PQ subcodes, DDP export (optional add on), and other 100% Redbook-compliant features. Need even more power? Check out our Peak Pro XT 5 bundle with over $1,000 worth of additional tools, including our acclaimed SoundSoap Pro, SoundSoap 2 (noise reduction and restoration), Sqweez-3 & 5 (linear phase multiband compression/ limiter/upward expander), Reveal (precision analysis suite), PitchCraft (super natural pitch correction/ transformation), Repli-Q (linear phase EQ matching), SuperFreq (4,6,8, & 10 band parametric EQ) and GateEx (advanced noise gate with downward expander) — all at an amazing price. So, when you're ready to master, Peak Pro 5 has everything you need. It's the perfect complement — and finishing touch — to Digital Performer 5.

Professional pad controller

The Akai Professional MPD24 is the velocity sensitive pad controller for musicians and DJs working with sampled sounds. The MPD24 features 16 MPC-style velocity and pressure sensitive pads plus transport controls for interfacing with Digital Performer and your virtual instruments. You get Akai's exclusive feel: either MPC 16 Levels or Full Level features for ultimate pad control. Now add four selectable pad banks totaling 64 pads, six assignable faders and eight assignable and 360 degree knobs for transmitting MIDI Control Change data. Included editor/librarian software gives you complete, intuitive programming and control for DP5 all of your other software titles. The MPD24 provides unprecedented creative freedom for manipulating sampled material.

The MOTU experts at Sweetwater can build your customized MOTU studio for you. We'll help you select the right components, and we can even install, configure and test the entire system for you. Why shop anywhere else?

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New Mackie monitoring

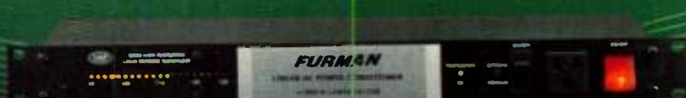
The high-resolution Mackie HR824mk2 active studio reference monitor sounds as smooth as it looks. The new Zero Edge Baffle™ minimizes diffraction for a crystal clear image of your Digital Performer mix, and controls sound waves for wide, even dispersion. The rear-firing, mass-loaded passive radiator ensures tight, detailed bass extension, down to 35Hz. And thanks to remarkably linear frequency response, you always get accurate mix translation. Acoustic Space, LF roll-off and HF controls let you tailor the sound to suit your MOTU studio space—and your taste. With all this and more, the HR824mk2 turns your Digital Performer desktop studio's sweet spot into a full-on sweet zone.

New hands-on control for DP5

The new Mackie Control Universal Pro control surface gives you ultimate hands-on control of your Digital Performer desktop studio. Nine motorized, touch-sensitive Penny + Giles faders, eight V-Pots and more than 50 master buttons let you tweak parameters to your heart's content. Unlike generic MIDI controllers, the MCU Pro employs a sophisticated communication protocol that delivers ultra-precise control, makes setup easy - no mapping required - and enables you to see your mix in action with real-time visual feedback via the huge backlit LCD and eight LED rings. Apply the custom overlay for Digital Performer for dedicated labeling of DP-specific functions. The MCU Pro is the ultimate way to mix in DP5!

Purified power

To get the most out of your MOTU studio gear, you need the cleanest power possible. The negative effects of poorly supplied wall outlet AC power on your gear can be dramatic, without your ever knowing how good your gear can really sound with properly supplied power. Furman Sound introduces the all-new Power Factor Pro with its ground-breaking Clear Tone Technology™, which actually lowers the AC line impedance supplied by your wall outlet while storing energy for peak current demands — over 45 amps of instantaneous current reserve. Additionally, Linear Filtering Technology™ (LiFT) dramatically lowers AC line noise to unprecedented levels in the critical audio frequency band. Also included are Furman's unique Series Multi-Stage Protection Plus (SMP+) surge protection and automatic Extreme Voltage Shutdown (EVS), which protect you from damaging voltage spikes or sustained voltage overload. Equipped with the same LiFT and SMP+ features, plus EVS Extreme, the Furman Sound IT-20 II ultra-low noise balanced isolation power conditioner is designed for the most critical, ultra-low noise installations. Delivering an astonishing 80dB of common noise reduction from 20Hz-20kHz, you're assured the lowest possible noise floor for all the gear in your MOTU studio. The IT-20 II's toroid transformer design assures a contained magnetic field for complete isolation from sensitive studio components nearby. The ultimate in purified power.



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Cut, Paste, and Process

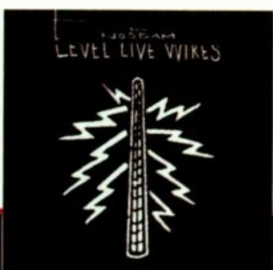
By Bill Murphy

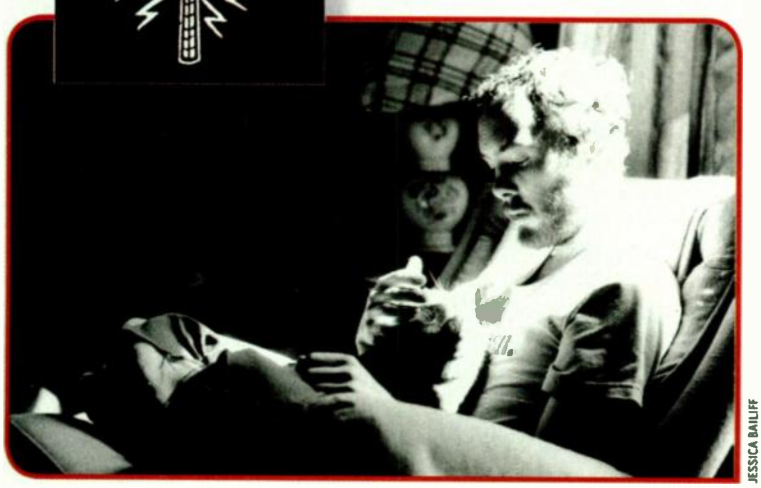
Odd Nosdam takes sampling to its freestyle extreme.

There's something beautifully off-kilter about the music of David P. Madson, aka Odd Nosdam. At first blush, his adventurous mix of dub and electronica seems to draw from the "audio collage" resurgence led by artists such as Prefuse 73, Four Tet, and Caribou. But a closer listen reveals strangely psychedelic, orchestral, and genre-bending aspects to his sound. Add in a few well-placed guest spots from the close circle of collaborators who orbit the Anticon label's oddball axis, and you've got a taste of *Level Live Wires* (Anticon Records, 2007), Madson's most radical effort yet.

"I'd say the album is about 90 percent sample based," Madson reveals. "I'm pretty obsessive about collecting—I try to find really random, unique old records at thrift stores or swap meets or even on the street. Every now and then, I come up with one that has a little sound or a couple of notes that I can sample and create my own melody with. Sometimes it'll be a break that I'm pretty into, but usually I'll just use sounds to come up with a strong melodic line, and then the beat comes after that."

Madson used to rely on a Boss SP-202 Dr. Sample (he gets by with only a Dictaphone and the built-in mic of his SP-202 for his nonvinyl "found sounds"). From there, he switched gears to an Akai MPC2000XL and most recently an E-mu SP-1200. "I'm really into the tiny details and the textures," he says, "and that's why I like the SP so much. It just has its own personality. If you sample harpsichords, pianos, or guitars on maybe 45 [rpm] and then tune them down, the SP transforms them into something totally different. Once I start compressing and EQ'ing that through the chain I have, it blows me away."





Level Live Wires/Odd Nosdam

The basic chain consists of the mix/EQ section of a vintage Tascam 488 Portastudio (whose trim control also serves as a lo-fi overdrive effect) and two Empirical Labs Distressors, which Madson uses to run a sound through several cycles before committing it to "tape"—in this case, Digidesign Pro Tools. He rarely maintains a Pro Tools session larger than 15 tracks, preferring to bounce down once he's found the right submix.

"I think it's very much like Joe Meek or Lee 'Scratch' Perry," he says, citing two of his key influences. "It's just about having confidence in what you're doing. With Perry, if you really start to dissect what you hear, you realize that a lot of it is him pounding on stuff and turning knobs and just bugging out. It's very much about how you manipulate things with your hands, and that's how I see my music."

Check out the deep spatial spread of "Fat Hooks" (with Jessica Bailiff on vocals) or the hypnotic harp melodies of "Kill Tone" (with Jeffrey "Jel" Logan playing live beats on an MPC2000), and Madson's meaning becomes clear. These and other tracks—the knife-edged "Burner" in particular, rendered even sharper with Madson's distorted Micromoog washes—are raw snapshots of handmade improvised genius, and are as compelling in what they transmit visually as what they do sonically.

"I make a lot of music," Madson says, "but with *Level Live Wires*, from the very get-go, I was really moved by something at the core of it. That's always what I hope will happen when I start building up my sounds—you take them completely out of context and make your own thing." EM

RIFFS

Odd Nosdam

Home base: Oakland, California
Key software: Digidesign Pro Tools HD
Favorite hardware sampler: E-mu SP-1200
Web site: www.anticon.com

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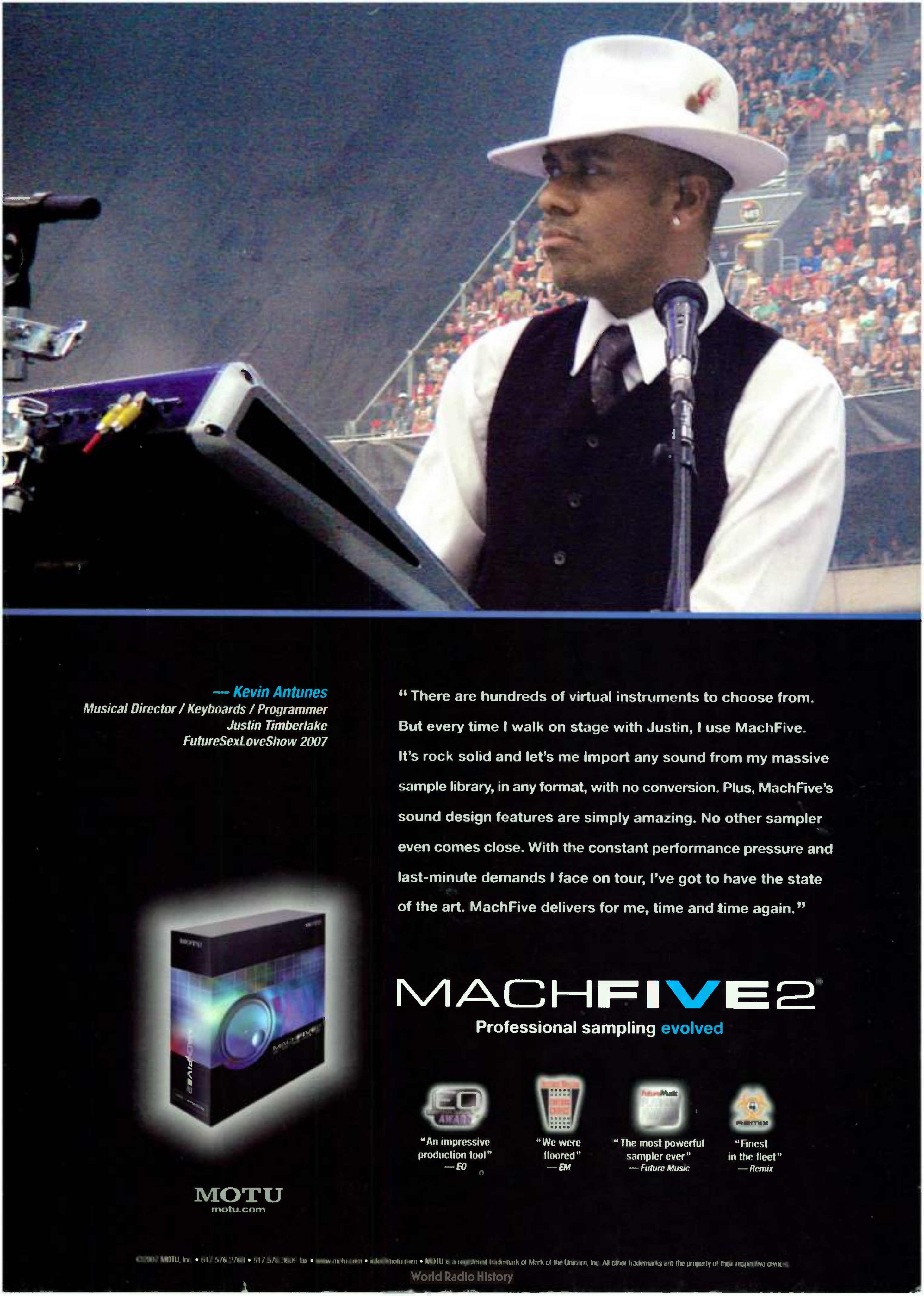

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