>> INDUSTRY INSIDER: THE CHANGING LEGAL LANDSCAPE FOR MUSICIANS

OCTOBER 2008

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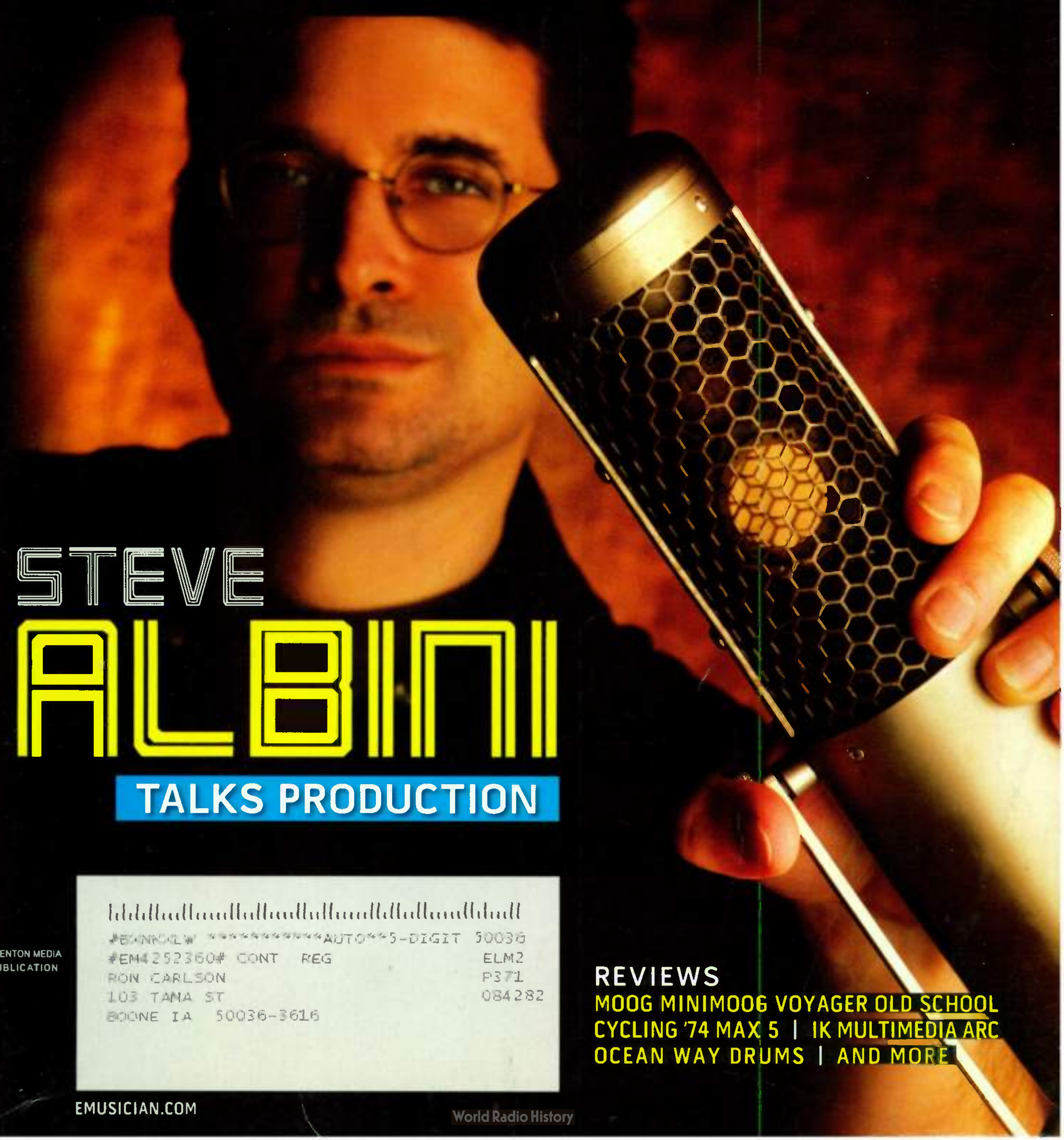
ELECTRONIC MUSICIAN

BETTER TONE THROUGH REAMPING

CREATIVE USES FOR MID-SIDE TECHNIQUE

10 GRANULAR SYNTHS COMPARED



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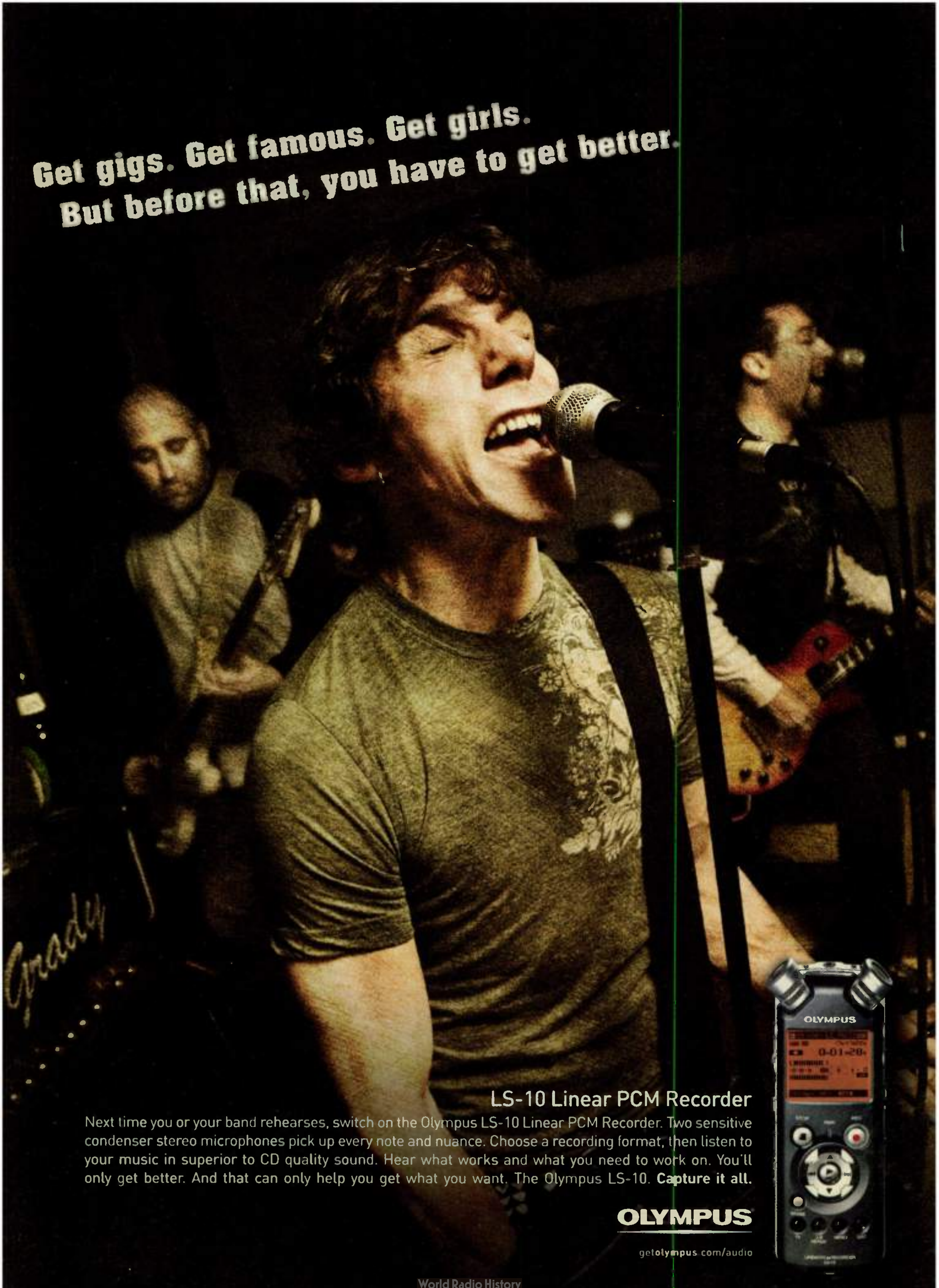
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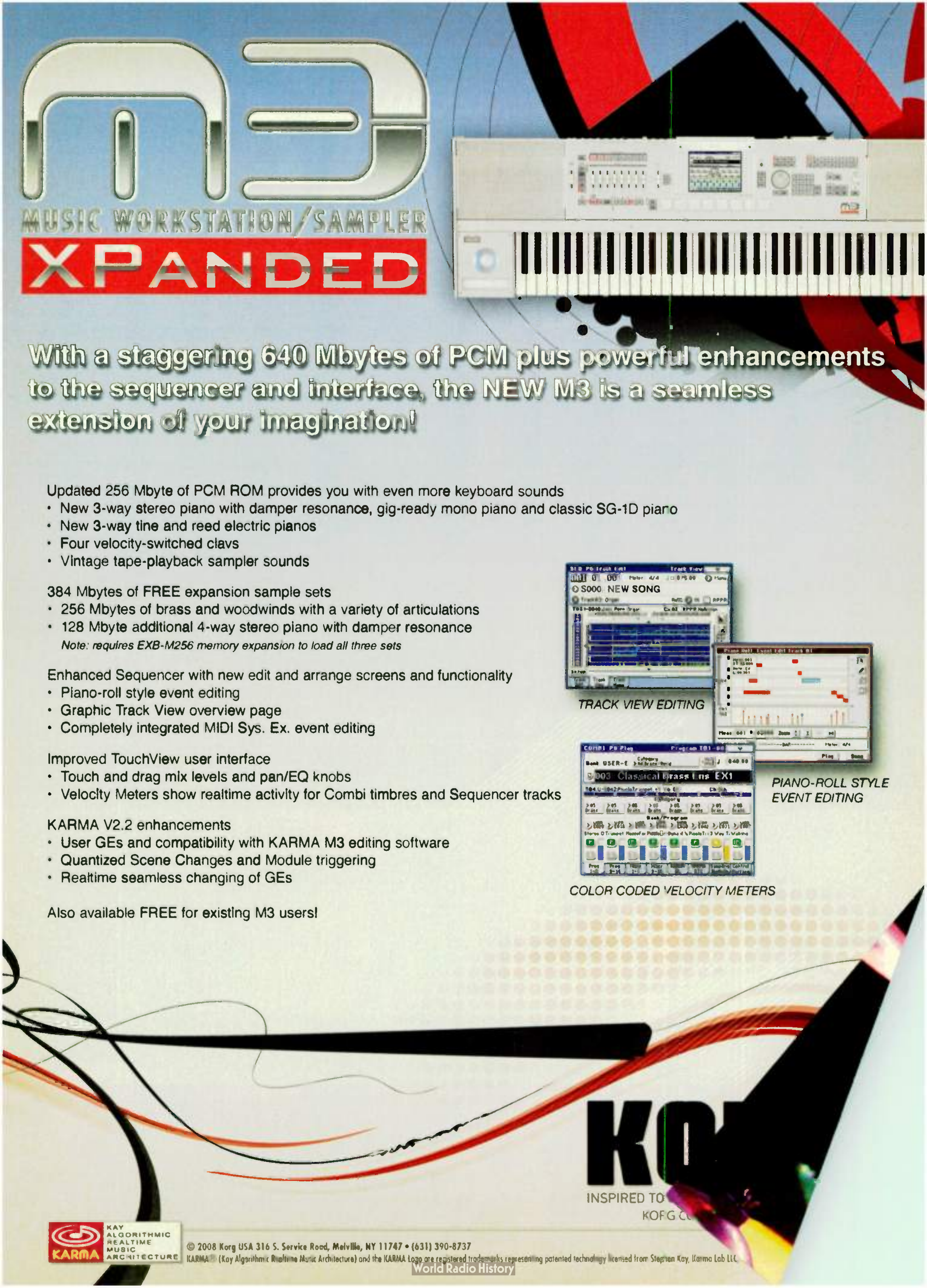
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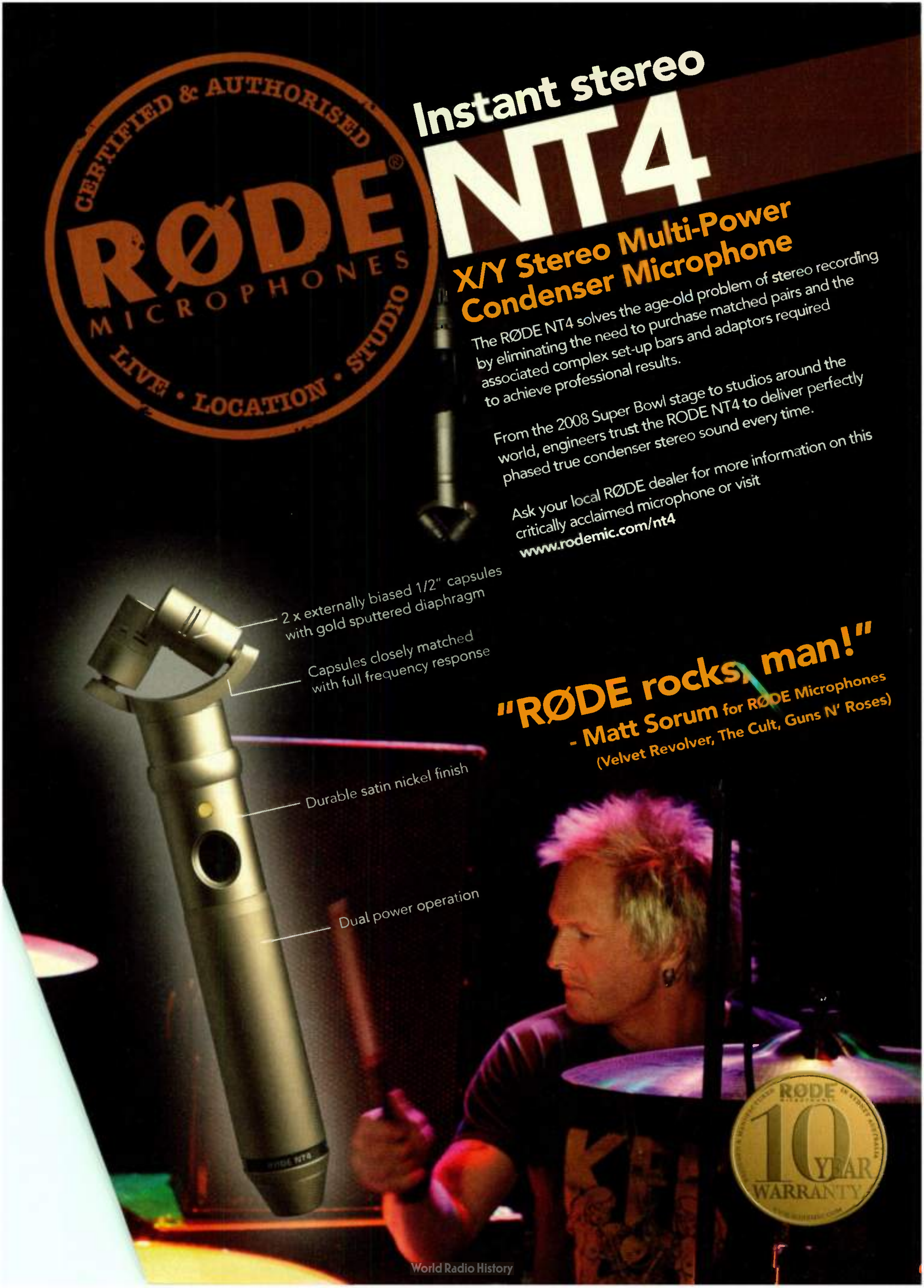
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FEATURES

RICH MARKESE

40 ELECTRICAL ENGINEERING

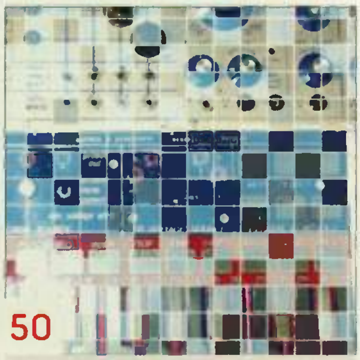
With his no-nonsense approach to recording and more than 25 years of experience, Steve Albini has crafted hit records for top-tier acts including Nirvana, Jimmy Page and Robert Plant, and the Pixies. In this interview, he shares production tips that can be used in any studio.

By Rich Wells

28 BETTER TONE THROUGH REAMPING

Why settle for a mediocre-sounding track when you can rerecord it through an amp? We show you how the pros use reamping to improve guitar and bass tracks, as well as drums, keyboards, and vocals.

By Mike Levine



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50 GOING WITH THE GRAIN

Granular synthesis has rapidly become one of the most popular new synthesis methods. We examine ten standalone granular apps that let you micromanage sound like never before.

By Dennis Miller

DEPARTMENTS

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More
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~~Instant~~ Inspiration, ~~Infinite~~ Possibilities...
The Motif XS is already the #1 synth workstation in the market, but we didn't stop there. Go to the #1 community support site in the industry – www.motifator.com to get **more**.
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>>FIRST TAKE

3 ≠ C

When describing the inhabitants of the fictional town of Lake Wobegon on *A Prairie Home Companion*, Garrison Keillor memorably notes that "all the women are strong, all the men are good-looking, and all the children are above average." Ah, if only it were a similar situation with audio products: it would make it a lot easier to write reviews!

Recently a distributor of a product that received a pair of 3s expressed great dismay, saying that such a rating would kill the item in the marketplace. While I know for a fact that the reviews in EM do not hold that kind of sway, he did bring up a popular misconception that some readers might have: that a rating of 3 means the product is average—in the negative sense—like a C grade for a high-school student who hopes to get into a prestigious university. But that's not the case in terms of EM's rating system.

I do agree with the distributor that to the casual reader of the magazine—someone who might judge a product from a quick glance at the meters and maybe the last sentence of a review—anything less than a 5 could be construed as bad. But to an astute, longtime EM reader, who is interested in the details of a review *and* is seriously considering a purchase, a 3 means that the product delivers what it should in a specific category, such as Audio Quality. "Good; meets expectations" is how we've described that rating since we redefined our system in February 2006.

We all use products like that. Sometimes the feature set is killer, but the audio quality is, well, just okay. I've also known the opposite to occur. For reviews, EM's editors make sure that the text and ratings are consistent, which can mean that the reviewer has to reconsider aspects of the review—positive or negative—before it goes to print. In fact, the editor will challenge the reviewer's conclusions if he thinks a rating is too high or too low based on the body of the review. We don't want to say something is "Unacceptably flawed" (a 1 rating) or "Amazing; as good as it gets with current technology" (a 5) unless we believe it to be true.


JANE RICHEY

In addition, we don't rate products of various types, prices, and complexity against each other equally across the board in our reviews, because you cannot accurately compare a simple, inexpensive product to a highly complex, high-ticket item. When we give a product such as the Korg KO-1 Kaossilator (an item that costs less than $200) a rating of 4 for Ease of Use and Audio Quality, we are rating it against other products that are designed for a similar purpose—in this case, to make sound and loop audio—in a particular price class. It's not going head-to-head with the Korg OASYS, a full-featured keyboard workstation costing upwards of $8,000.

So if another product in the same issue (August 2008 in this case) gets 3s for Ease of Use and Audio Quality, such as Redmatica Compendium 1.5, does it mean that the inexpensive synth/looper sounds better and is easier to use than a specialized suite of software tools designed to do very complex things? Of course not, because we're not comparing a software bundle to a hardware instrument. Each is being rated on how it performs within its own class of tools. Although there might be nothing that does quite what the apps in the Compendium bundle do, for example, the user experience *can* be evaluated and rated after a period of time. Our reviews are designed to give you a real-world evaluation that should match the typical user experience, because our writers test the products for several weeks in their projects.

Fortunately for our editorial staff (and occasionally to the chagrin of our advertisers), we're not pressured by our sales staff to produce glowing reviews for "paying clients." The products we choose to cover are selected based on what we think is most relevant to our readers, and we judge these items on their merits alone.

In describing what the ratings signify, we've kept in mind the real-life expectations of our readers. As a reminder, we have reintroduced the ratings descriptions on the first page of the Reviews section (see p. 74). I hope that you find them useful in evaluating how the products we review fit your own production needs.

Gino Robair
Editor

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EDITOR
Gino Robair, grobair@emusician.com

EXECUTIVE EDITOR/SENIOR MEDIA PRODUCER
Mike Levine, mlevine@emusician.com

SENIOR EDITOR
Geary Yelton, gyelton@emusician.com

ASSOCIATE EDITORS
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

CONTRIBUTING EDITORS
Michael Cooper, Marty Cutler, Larry the O, Steve Oppenheimer, 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 PRODUCT DEVELOPMENT MANAGER
Tami Needham, Tami.Needham@penton.com

ONLINE AUDIENCE DEVELOPMENT MANAGER
Zach Smoot, Zach.Smoot@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

SENIOR VICE PRESIDENT
Kim Paulsen, Kim.Paulsen@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

EAST/EUROPE ADVERTISING DIRECTOR
Michele Kanatous, Michele.Kanatous@penton.com

WEST/MIDWEST ADVERTISING DIRECTOR
Erika Lopez, Erika.Lopez@penton.com

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

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

MARKETING COORDINATOR
Tyler Reed, Tyler.Reed@penton.com

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

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

CLASSIFIEDS PRODUCTION COORDINATOR
Linda Sargent, Linda.Sargent@penton.com

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

ADVERTISING PRODUCTION COORDINATOR
Jennifer Scott, Jennifer.Scott@penton.com

OFFICE MANAGER
Lara Duchnick, Lara.Duchnick@penton.com

Penton Media

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

CHIEF REVENUE OFFICER
Darrell Denny, Darrell.Denny@penton.com

EDITORIAL, ADVERTISING, AND BUSINESS OFFICES
6400 Hollis St., Suite 12, Emeryville, CA 94608, USA, (510) 653-3307

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FRONTPANEL... By Gino Robair

Download of the Month

d16 Group's Fazortan (Mac/Win) By Len Sasso

It's always a pleasure to find a plug-in that does one job, does it well, and makes it easy. Fazortan (about $42) from Polish software developer d16 Group (d16.pl) is a cross-platform AU and VST effects plug-in devoted solely to flanging and phasing. It is a true stereo effect: the right and left channels are processed separately, and a Stereo Phase knob lets you offset the phases of the LFOs affecting each channel by as much as 180 degrees. Among other things, you can use that to add stereo depth to a mono input (see **Web Clip 1**).

ONLINE BONUS MATERIAL

Audio entering Fazortan is passed through a notch filter with two, four, six, or eight notches. You use the Feedback knob to enhance the filter's effect by accentuating its peaks and notches. The dry/wet knob determines the mix of the filtered and unprocessed signal, and the Output Volume knob and associated LED meters let you set the output level. For motion, you get two multiwaveform LFOs whose signals are mixed to modulate the notch filter's frequency. The notches are spread evenly across the frequency spectrum from 5 Hz to 20 kHz, and the LFOs shift all notches equally. The LFO frequencies, depths, and waveforms are completely independent, and you can add a DC offset to the amplitude of the first LFO. That gives you a lot more waveshape flexibility when the LFOs are mixed.

Although Fazortan produces the usual whooshing and whirring effects associated with phasing and flanging, it also offers very subtle sweetening. The audio quality is excellent, and it can make an otherwise dull track stand out without taking over (see **Web Clip 2**). The 28 factory presets cover the basics, and you can save your own presets and preset banks using a categorized browser. Fazortan's MIDI Learn implementation lets you latch any control to any MIDI continuous controller, and you can save MIDI controller maps. Phaser/flanger plug-ins are common, but Fazortan's price and sound quality make it worth checking out.

OPTION-CLICK By David Battino

Load a 6-channel WAV file into ffmpegX and select the .AC3 output setting, and you'll get a 5.1 audio file you can burn to DVD.

Burn Your Own Surround Sound DVDs

Discover cool features lurking inside popular programs

There are many ways to distribute your music in 5.1-channel surround. But encoding it in Dolby Digital (AC-3) format will accommodate the most listeners, because all DVD players support AC-3. There are numerous AC-3 encoders online. You simply feed them six mono 16-bit, 48 kHz WAV files or an interleaved 6-channel WAV file and click on a button. Then you load the resulting AC-3 into a DVD-burning program, create some menus, and burn a disc. (I burn a disc image while testing.)

On the Mac, you can make surround DVDs with Roxio Toast (roxio.com; $79.99), but there's a secret: to import an AC-3 file, you have to Option-drag it onto the Toast window. In Windows, try Audio DVD Creator (audio-dvd-creator.com; $39.95). For AC-3 encoding on the Mac, I use De-Interleaver (scottwilson.ca; free) followed by ffmpegX (ffmpegx.com; $15), with WAVs arranged in L, C, R, Ls, Rs, LFE order. (For more about David Battino's work, visit batmosphere.com.)

Audio Innovations: Highlights from the 2008 TECnology Hall of Fame

1898
Telegraphone Wire Recorder
Danish engineer Valdemar Poulsen (1869–1942) invents a magnetic recorder

1928
Western Electric 618A Electrodynamic Transmitter
Bell Labs engineers E. C. Wente and Albert Thuras develop the first commercial (omni) dynamic mic

1940
Shure Unidyne Model 55
The first low-cost unidirectional (cardioid) dynamic microphone

1947
JBL D130 Loudspeaker
James B. Lansing's speaker design helps define the modern woofer

THIS MONTH'S SOUNDTRACK

These albums encompass a diverse range of styles and composition methods, from experimental and electro-pop to free improv and progressive rock.

1. Charles Cohen/Ed Wilcox: *Those Are Pearls That Were His Eyes* (Ruby Red)
2. The Hub: *Boundary Layer* (Tzadik)
3. Val-Inc: *On* (Innova)
4. Frank Zappa: *One Shot Deal* (Zappa Records)
5. Free Blood: *The Singles* (Rong/DFA)


1.
WENDY JANE HYATT

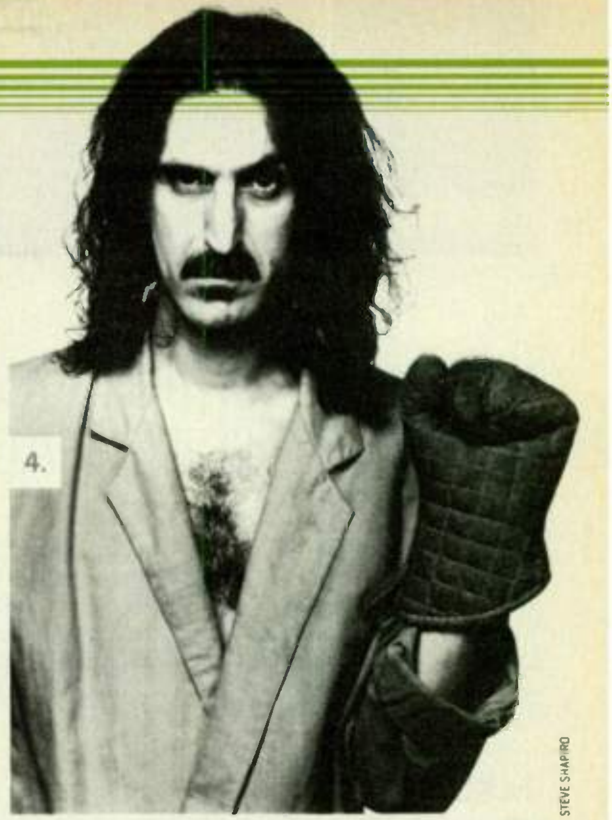
CHARLES COHEN/ED WILCOX
Delightful improvised duets, with Cohen on Buchla Music Easel and Wilcox on acoustic drums.

THE HUB
An engaging 3-disc set documenting 20-plus years of music from this seminal computer-network band.

VAL-INC
Percussion and vocal samples are the foundation of Val-Inc's engaging and politically charged audio collages.

FRANK ZAPPA
A collection of '70s live tracks—from free improv to fully structured—featuring a variety of FZ groups.

FREE BLOOD
The Brooklyn-based duo's debut offers a poppy blend of post–Talking Heads funkiness with electronic sensibilities.


4.
STEVE SHAPIRO


5.

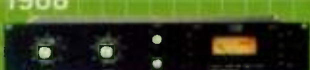

2.


3.
PA

EM POLL

Do you use a computer onstage? a) Yes, a Mac laptop b) yes, a PC laptop c) yes, a desktop Mac or PC d) no, I don't e) what, are you crazy? Submit your answer to this poll and others at emusician.com. This is not a scientific poll but a tabulation of readers' responses and is just for fun!

1968



Universal Audio/UREI 1176 Peak Limiter
Bill Putnam uses FETs as voltage-variable resistors in a gain-control device

1976

dbx 160 (VU) Compressor/Limiter
David Blackmer's level detection circuits change the world of pro audio

1977

Yamaha NS-10M Speaker
The most common studio reference monitor of the '80s and '90s

1983

Kurzweil Music Systems K250
Ray Kurzweil's ROM-based sampler has an 88-note, velocity-sensitive keyboard

>> For more **Hall of Fame innovations**, visit mixfoundation.org/hof/08techof.html.

By Len Sasso

MOTU 896MK3 FIREWIRE INTERFACE

MIXING IT UP

MOTU (motu.com) has introduced the third generation of its 896 series of FireWire audio interfaces with the 896MK3 (Mac/Win, $1,295). The unit has onboard effects and 16-bus digital mixing, and you can use it without a computer as a 2U rackmount standalone mixing solution. Included in its total of 28 inputs and 32 outputs, you'll find 8 XLR/RTS combo inputs with mic preamps, high-impedance guitar inputs, and 2 banks of optical I/O. The interface supports 192 kHz operation and 32-bit floating-point DSP. The onboard effects (reverb, 7-band parametric EQ, and compression/limiting) provide no-latency effects processing and monitoring. CueMix FX control software gives you access to all functions from your computer.

TAKE IT HOME

CAKEWALK SONAR HOME STUDIO 7

Cakewalk (cakewalk.com) has just released version 7 of Sonar Home Studio (Win, $99; upgrade, $39). The program sports a completely redesigned user interface stressing ease of use. New Track and Console views put many of Sonar's pro-level tools for music creation at your fingertips. In addition, you get assistant tools to help with everything from track setup to advanced signal routing. A new Loop Explorer view takes the guesswork out of integrating sounds from your loop collection into your music. The XL version ($159) adds more instruments, effects, and content.

BAG END PM6 POWERED MONITOR

IT'S IN THE BAG

With nearly 30 years' experience, Bag End (bagend.com) has earned a reputation for manufacturing high-quality professional monitors. The new PM6 ($898 [MSRP]) is a powered version of the company's popular M6 close-field monitor. Weighing in at 18 pounds, this 9 × 14 × 9–inch unit features a 100W amp, a 6-inch coaxial driver, and a 1-inch neodymium tweeter. Bag End claims flat response in both the frequency and time domains, with ±3 dB frequency response from 60 Hz to 20 kHz. Maximum output is rated at 107 dB SPL.

AUDIO-TECHNICA AT2035 AND AT2050 MICROPHONES

SIDE BY SIDE Audio-Technica (audio-technica.com) has expanded its line of 20-series side-address condenser microphones with the cardioid AT2035 ($249) and the multipattern AT2050 ($369). Both mics offer flat frequency response from 20 Hz to 20 kHz, high-SPL handling capability (148 dB, or 158 dB with built-in 10 dB pad), and low self-noise. The AT2050 offers omni, cardioid, and figure-8 polar patterns. Both mics feature a switchable 80 Hz highpass filter for low-end rolloff, low-impedance balanced XLRM connectors, a shockmount, and a protective pouch.

IK MULTIMEDIA TOTAL WORKSTATION RACK

MUSE AT WORK IK Multimedia (ikmultimedia.com) has teamed up with Muse Research (museresearch.com) to package the Powered By SampleTank virtual instruments series in the Muse Receptor plug-in-hosting hardware workstation. Total Workstation Rack ($1,999; cross-grade, $1,699) has SampleTank 2.5, Sonik Synth 2, Miroslav Philharmonik, SampleMoog, and SampleTron—all installed in Receptor and authorized. You get more than 10,000 sample-based sounds covering acoustic and electric pianos; orchestral strings, brass, and woodwinds; vocals, organs, guitars, basses, and synths; and drums and ethnic percussion. You can use Receptor as a standalone instrument on the gig and as a DAW plug-in in the studio.

GET SMART

Groovebox Corp.'s Designing Electronic Drums

The latest in Groovebox Corp.'s (grooveboxmusic.com) series of Power PAK tutorials is devoted to creating electronic drum sounds in Ableton Live 7. *Designing Electronic Drums* (Mac/Win; boxed, $39.99; online viewing, $14.99) consists of 15 tutorials, running more than 3 hours, by Craig McCullough. The tutorials cover all aspects of creating and processing individual drum sounds from samples and waveforms, and then combining them in complete kits using Live's Drum Racks.

St. Martin's The Indie Band Survival Guide

The Indie Band Survival Guide: The Complete Manual for the Do-It-Yourself Musician ($14.95), by Randy Chertkow and Jason Feehan, reveals how to successfully get your music out to the public based on the authors' experience with their band Beatnik Turtle. Topics include selling your music online; getting airplay on radio, TV, movies, and Podcasts; press relations; and Web-site design. You'll also find tips on how to protect your music and how to make money without being a jerk. Check out St. Martin's Press (indiebandsurvivalguide.com) for details.

HowAudio's Cubase 4: Essential Training

Cubase 4: Essential Training ($34.95) is an 8-hour tutorial hosted by independent Steinberg product specialist Matthew Loel T. Hepworth. The DVD's QuickStart Overview gets beginners up to speed with Cubase fundamentals. Other sections for veterans and beginners alike cover external hardware, surround sound, VST3 instruments and effects plug-ins, the Universal Sound Manager, and scoring. As an alternative to the DVD, HowAudio (howaudio.com) offers subscriptions ($19 monthly, $199 annually) for 24/7 online access to the more than 1,100 video tutorials in the company's catalog.

Sound Advice

Impact Soundworks' Impact: Steel

Impact: Steel (Mac/Win, $79) is the first product from Impact Soundworks (impactsoundworks.com), a collaboration between sound designers Andrew Aversa and Wilbert Roget II. Produced by Roget, this sound library for Native Instruments Kontakt, Steinberg HALion 3, and Tascam GigaStudio 3 features multisampled recordings of five metal objects: steel frame, metal cylinders (two sizes), metal spring, and metal cone. Articulations include hits, scrapes, rolls, and other textures. They are provided as separate instruments but are mapped so as to all be playable from a single keyboard. You also get 16 highly processed, pitch-mapped FX Patches.

Heavyocity's Evolve

Evolve (Mac/Win, $399 [MSRP]) from Heavyocity Media (heavyocity.com) is a collection of Native Instruments Kontakt Player 2 presets designed primarily for TV and video-game production. Noted producers Dave Fraser and Neil Goldberg (NFL, GE, Sony, and others) organized the collection for quick production. Categories include Rhythmic Suites (tuned and untuned percussion loops and arps), Percussive Kits (standard and found sounds), Stings and Transitions (sweeps, builds, and mangled instruments), and Tonality and FX (bass, melody, and pads). Mix and match instruments yourself, or start with one of 25 8-channel Multis designed for maximum compatibility.

Sonivox's Anatomy

If your percussion tracks lack that human feel, Anatomy (Mac/Win, $199) from Sonivox might be just the thing for you. This collection of roughly 800 presets for Native Instruments Kontakt 2 and 3 is produced entirely from melodic and percussive sounds created using parts of the human anatomy. The library is divided into two categories: Man (little processing beyond filter and reverb) and Machine (severely dealt with). Both categories contain drum kits, Foley and special effects, leads and basses, and pads. You'll find MP3 audio examples and Kontakt 2 demo instruments at sonivoxmi.com.

BLUE CAT AUDIO REMOTE CONTROL 2.0

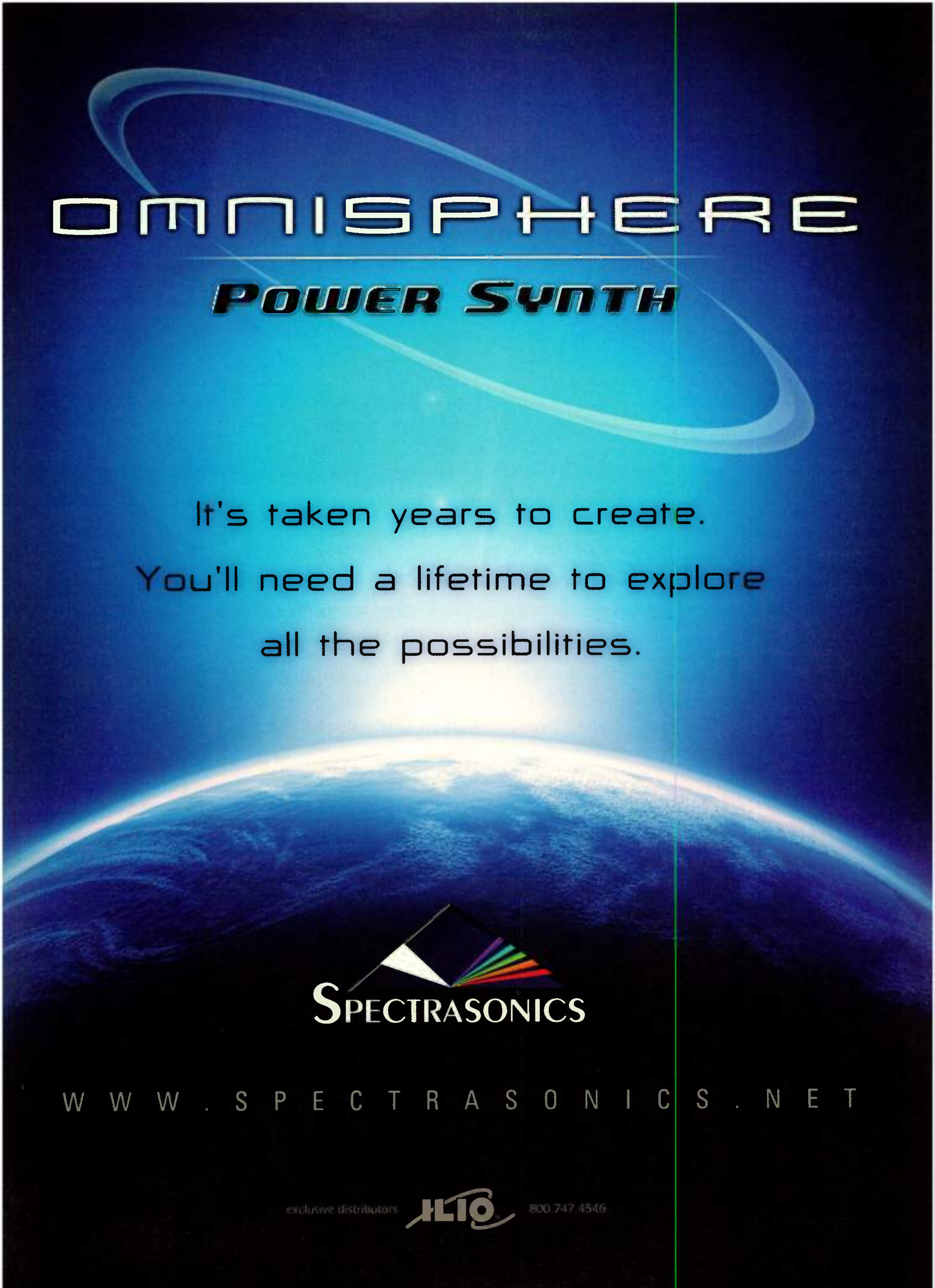
HANDS ON

Blue Cat Audio (bluecataudio.com) has come up with an unusual approach to MIDI remote control. Remote Control 2.0 (Win, $59) is a VST effects plug-in that puts a skinnable control surface on your PC desktop. Depending on the skin, you might have onscreen knobs, sliders, numericals, buttons, or *x-y* graphics for generating MIDI Control Change messages. Various kinds of meters and graphs monitor and relay incoming MIDI. The primary purpose of the plug-in is to simultaneously access the controls of several other plug-ins (using their MIDI Learn capability) from Remote Control's panel, thereby reducing desktop clutter.

APPLIED ACOUSTICS SYSTEMS STRUM ACOUSTIC GS-1

EASY PICKIN'

Applied Acoustics Systems (applied-acoustics.com) is renowned for the quality and realism of its physical-modeled virtual instruments. With Strum Acoustic GS-1 (Mac/Win, $229), the company has outdone itself by modeling not only the sound but also the voicings and playing techniques of acoustic guitar. You choose a guitar model or create your own using global and individual string parameters, and then play chords on your MIDI keyboard controller. Strum chooses guitar-chord voicings based on settings you make, then either strums automatically or lets you trigger individual strings and strum styles by playing MIDI notes. em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» *Posters Fade*

Getting More from Less

Derby produces its latest on an aging Mac with almost no plug-ins.

The Portland, Oregon, trio Derby—Nat Johnson on lead vocals, guitar, and bass; Dave Gulick on vocals, keys, and guitar; and Isaac Frost on drums and percussion—crafts a brand of tight, rocking alt-pop that harkens back to some of the post-new-wave bands of the '80s and '90s. But Derby's latest album, *Posters Fade* (Green Submarine Records, 2008), boasts a sound that's less retro than it is old school; it's fresh, but created with a relatively simple and unadorned recording process.

By Emile Menasché

The band records in a basement home studio and runs Digidesign Pro Tools LE through a Digi 002 into an old Apple Power Mac G4 Dual 450. "We use that to process, and it writes to a LaCie FireWire drive," says guitarist and front man Johnson, who does the bulk of the engineering. "We have two 20-inch [computer] monitors that take 10 minutes each to warm up. I'm not sure if that's normal, or even safe!"

One reason Derby can get by with an aging computer is that the band uses a minimal amount of plug-ins, opting instead to record organically and get good sounds from their instruments and amps. "We take a lot of time to get tones," Johnson says. "We like how great things sound simply." The band usually records right in the control room, which Johnson describes as an acoustically dead space. For analog warmth, mics run through an Allen & Heath MixWizard3 16:2 mixer.

On the drums, the band uses Shure SM57s on the snare (one top, one bottom) and on the rack toms, an SM57 or SM58 on the floor tom, and an AKG D112 on the kick. "We put the mic in that hole and got so much low end and thump," Johnson says. "Mic placement became [a form of EQ]. We used cheapo condensers as overheads. They lacked a lot of body, but that was okay; they were pretty bright on the cymbals."

Vocals were recorded through a Røde NTK tube mic, straight into the computer. But keys—especially an old Wurlitzer electric piano—were miked through amps. "We'd put the NTK tube mic in front of the amp and it would get very gritty," Johnson says. "Lead sounds were done via [Propellerhead] Reason, out through an amp. We never do the digital-to-digital transfer—it just doesn't happen."

The bass was captured direct but often was later sent through a Tech 21 SansAmp Bass Driver for tone shaping. "I did most of the guitar through a Gibson SG into an early '70s Fender Deluxe Reverb," Johnson continues. "The amp was so dynamic. I didn't really use pedals. I just used the natural gain you could get from the Deluxe."

To fight the inertia that can happen when musicians record at home, Derby works on one song at a time, not moving on to the next set of tracks until the first song is edited and mixed. The trio makes decisions about what to keep as they go, rather than comping from lots of takes after the fact. "I mix it all with Sennheiser HD 280 Pro headphones, which people find funny," Johnson says. "I reference in a million different places: car stereos, iPods, etc. I try to listen to a lot of other music that I think sounds good while I'm mixing.

"Our sounds are pretty organic, and the part that we enjoy is the arrangement and layering of all the simple sounds to create something different—hopefully musically familiar, yet unique and just off-kilter." em

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COURTESY GREGORY TAYLOR

GREGORY TAYLOR

Home base: Madison, Wisconsin

Primary software: Cycling '74 Max/MSP and Radial (on an Apple MacBook Pro), Audio Ease Altiverb

Audio interface: Apogee Duet

Web site: rtqe.net

» *Two Maps of Danaraja*

All the Pretty Algorithms

Gregory Taylor extends the reach of ambient improv

"The world is full of musicians who can humble us with their ability to play for really long periods of time," says mixologist and producer Gregory Taylor, explaining his fascination with Indian ragas and Indonesian tonal systems—just two of the many musical forms that propel his creative flow. "One of the great gifts of experimental music of the 1960s, and John Cage in particular, was the idea that what makes a piece of music meaningful is the *contemplation* of it. And in the last decade or so, the arrival of wholesale access to non-Western music also gave us the great gift of discovery that this wasn't our idea."

By Bill Murphy

A veteran of the indie "cassette culture" craze in the '80s, and a tireless musical traveler (with a weekly radio show on Madison's WORT-FM that has aired since 1986), Taylor knows a thing or two about the regenerative power of sound. His mastery of Cycling '74's Max/MSP opened the door for his contribution to another one of that company's products: Radial, a loop-based performance suite (since discontinued) developed by John Eichenseer. Taylor has used the program to stunning effect on a recent spate of releases: *The Desert Fathers: Coptic Icons* (pfMentum, 2007), recorded live with trumpeter Jeff Kaiser; PGT's *Temporary Habitations* (Loochtone, 2008), featuring mandolinist Terry Pender and programmer Brad Garton; and Taylor's solo outings, *Amalgam: Aluminum/Hydrogen* (Palace of Lights, 2007) and *Two Maps of Danaraja* (Stasisfield, 2008).

"I've found that the simplest thing you can do to sound different from everybody else is to make your own loops," Taylor explains. "Radial matches my sensibility because it's something that I can play with from zero as soon as I walk onstage. I tend to start with nothing when I play live—on *The Desert Fathers*, there might have been some filtered shortwave broadcasts or some water in a bathtub or something—and I do that because a lot of my work is informed by Indonesian tradition. With electronic instruments, it's easier to manage timbre if you work with just intonational variants of Indonesian tunings."

In his solo outings, Taylor explores the subtle timbral variations of the Indonesian gamelan ensemble with an even more drawn-out ambient approach; ringing sonic textures gradually morph and "breathe" to reveal new elements or to build on a haunting theme. In his collaborations, Taylor often delves into Radial's ability to cut up and rearrange samples in real time, adding a glitchlike rhythmic quality to his soundscapes.

"We're essentially designing big structures that we steer by using a single control," Taylor notes. "Instead of thinking about doing music that's about change, you do it about the *rate* of change. Instead of controlling a filter all the time, for example, the filter slewing is essentially under some other piece of control, and I'm controlling something else that controls that as a side effect."

As with any experimental music that's improvised almost entirely live, the end result does involve some risk. But when it comes to working in a group, the approach has its rewards. "There's not a performance tradition for using laptops," Taylor says. "I was afraid when I started working with other people that I wouldn't be able to adapt to that. And at a certain point, you realize that working in a community gives you the opportunity to fail. But the upside is that I've found myself in situations where I can play with people who are my friends. How does it get better than that?" em

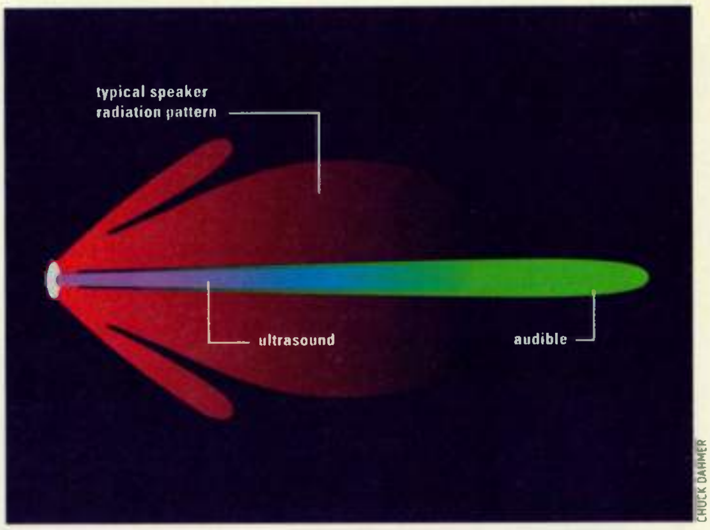

FIG. 1: The Audio Spotlight uses the interaction of ultrasonic sound waves with the air to reproduce audible sounds in a highly directional pattern. By contrast, a conventional speaker radiates sound in a wide pattern, often with side lobes that the Audio Spotlight avoids.

Getting Directions

Put sound just where you want it | By Scott Wilkinson

In most situations, electronic musicians want speakers to radiate sound evenly throughout a listening area, delivering a similar sonic experience to everyone within earshot. And most speakers do just that. But there are certain circumstances in which a highly directional beam of sound would be desirable—for example, playing dance music that only those in a certain area of a large space could hear, allowing others outside that area to talk without having to shout over the music.

This is the idea behind the Audio Spotlight from Holosonic Research Labs (holosonics.com). Based on underwater-sonar research from the 1960s, the Audio Spotlight was developed by company founder Dr. Joseph Pompei in the 1990s.

The original sonar research uncovered an interesting phenomenon: ultrasonic sound waves interact in a nonlinear manner with the medium in which they travel, distorting the original waveform. This distortion can include audible components, essentially transforming the medium itself into a highly directional speaker. The trick is to modulate the ultrasound so that the desired audio is heard.

Several large companies, including Matsushita (parent of Panasonic), Denon, and Ricoh, have tried to develop an audio speaker based on this principle, but they couldn't fully overcome problems such as cost, power, and high levels of distortion (which were greater than 50 percent THD). Pompei took up the challenge as a graduate student at Northwestern University and MIT, where he solved the problems that had plagued earlier attempts.

The Audio Spotlight starts with a normal audio signal, which is then processed in a manner that is precisely the inverse of the nonlinear distortion expected during playback. Next, a broadband ultrasonic carrier in the 65 kHz range is amplitude-modulated with the processed signal, amplified, and sent to an ultrasonic transducer, which converts the signal into sound waves. As the ultrasound interacts with the air in front of the transducer, the resulting nonlinear distortion demodulates the sound and counteracts the processing performed on the original signal, leaving just the intended audio, which the company claims is accurate to within a few percent THD.

But why is the sound so directional? Basically, the area of interaction between the ultrasound and air—which is long and narrow due to the directivity of high frequencies—acts like a very large end-fire line array of speakers, which is very directional because the size of the source is large compared with the audible wavelengths it produces. Another, less accurate analogy is a shotgun microphone, whose pickup pattern is highly directional because it's basically an end-fire array of mic capsules.

The nonlinear demodulation starts a small distance from the transducer, and the intended audio can be heard quite a bit farther away (see Fig. 1). By contrast, a conventional speaker radiates sound in a very wide pattern, often with side lobes (also depicted in Fig. 1). Side lobes do not exist in Audio Spotlight systems; the beam is much like that from a flashlight.

I asked Pompei if the Audio Spotlight could be used in a 2-channel sound system, but he says this is unnecessary. Because there is no room interaction, Pompei claims that sound from the Audio Spotlight tends to be perceived as "enveloping" the listener, almost like wearing headphones, reducing the need for 2-channel reproduction from regular speakers.

The potential applications are many and varied, from limiting the area in which dance music can be heard to individual audio kiosks in a museum or store to audio billboards that can be heard only in a particular spot on the street. Although it might not be directly applicable to studio or live sound reinforcement, it's a very interesting technology nonetheless. em


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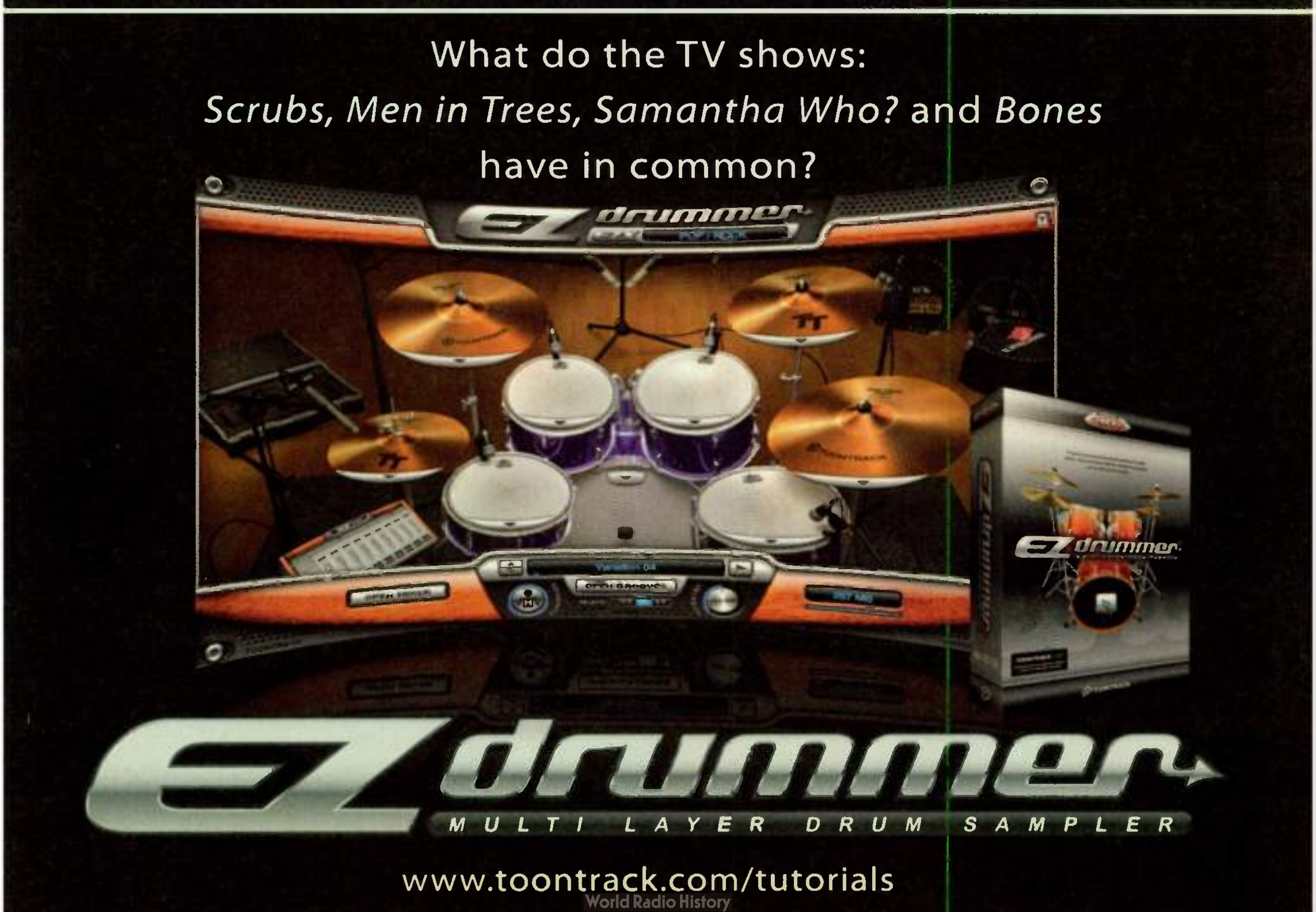
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By Mike Levine

Better Tone Through Reamping

Seven pros offer recipes for revitalizing your tracks.

Have you ever recorded a guitar track in your studio only to find, long after the guitarist had left, that the sound wasn't quite right for the song? Or perhaps you cut a bass part through a DI but wished later that you'd tracked it through an amp. Or maybe you were mixing a song with a soft-synth track that was well played, but the sound was lifeless.

In all those scenarios, and countless others, sonic improvement could have been readily achieved through the process commonly referred to as *reamping*—that is, sending your already recorded track through an instrument amplifier or an instrument-level processor (or both), and then bringing it back into your DAW as a new track. The term *reamping* is a bit of a misnomer because, as you'll see, the track being reamped has usually been recorded direct, not through an amp.

Although reamping most often involves guitar and bass tracks, you can apply it to any recorded audio. Drums, vocals, keyboards, or even a drum loop that needs spicing up can be improved through reamping. The only limit is your imagination. Ironically, in this era of digital signal processing, the devices that make reamping possible are completely analog. (I'll describe these boxes more in a bit.)

In researching this story, I talked to a number of pro engineers and producers who frequently use reamping in their studio work: Dave Bottrill is a producer-engineer whose client list includes Peter Gabriel, King Crimson, and Tool. Producer-engineer Butch Walker has worked with artists like Avril Lavigne, Hot Hot Heat, and Fall Out Boy, among others. Jesse Nichols is a staff engineer at Fantasy Studios in Berkeley, California, whose credits include the Donnas, the White Stripes, and Doves. Joel Hamilton is a New York City–based producer-engineer and musician who's worked with artists such as Tom Waits, Soulive, and Elvis Costello. Paul Antonell owns the Clubhouse, a commercial studio in Rhinebeck, New York, and has engineered for Natalie Merchant, Rusted Root, and Al Di Meola, among many others. Steve Skinner is a producer, composer, and frequent EM contributor whose credits include Akon, Jewel, and Celine Dion. I also spoke with engineer John Cuniberti, who designed and built the Reamp, the first commercially available reamping device. Cuniberti's credits include Joe Satriani, the Neville Brothers, and Train.

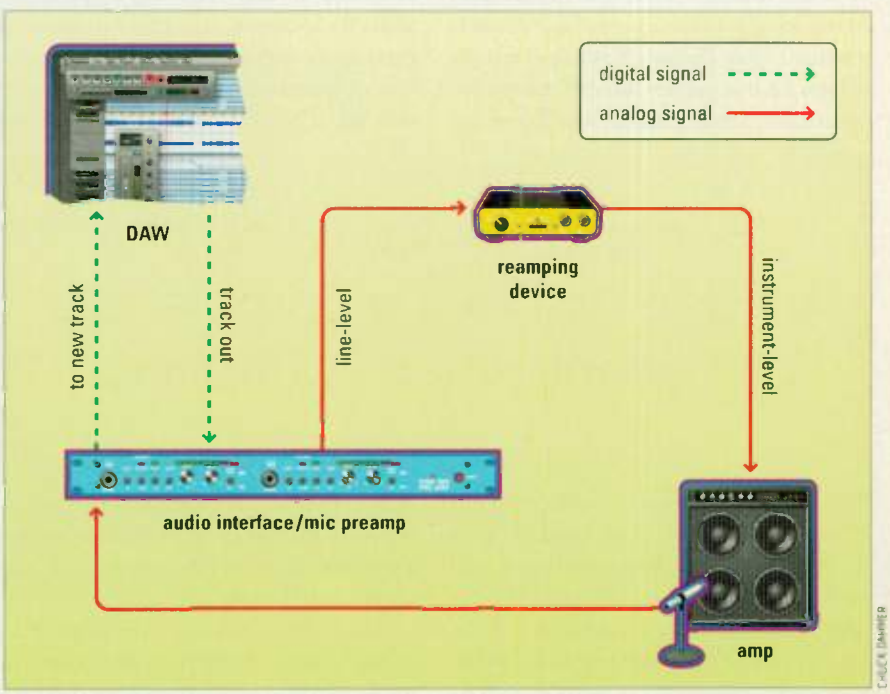

FIG. 1: This shows the basic signal path for reamping a track through a guitar amp.

Signal Impersonator

Before getting into various applications for reamping, let me explain how the process works. At the very basic level, reamping consists of taking a recorded track, routing it into an amplifier, and then capturing the amplified signal back into your multitrack.

However, it's not quite as simple as that. Here's why: your recorded tracks are low-impedance, line-level signals. They interface successfully with the line inputs of audio interfaces, mixing consoles, and most rack-mount processors. But they're not compatible with the high-impedance (and lower-level) signals needed to drive a guitar amp, bass amp, or stompbox. This is where the reamping processor comes into the picture (both passive and active models are available; see the sidebar "Reamping Tools"). These devices take those line-level signals and convert them to instrument level, as if they were coming out of the cable attached to your guitar. "As far as the guitar amp is concerned, it thinks there's a guitar plugged into it," says Cuniberti.

There are some differences, though. Reamping "will never be as good as having a guy plugged into an amp and standing next to it," says Cuniberti, because in that scenario, "there's going to be some sympathetic vibration produced from the guitar pickup—and the strings, for that matter—hearing the speaker in the room. There's going to be an interaction there that you just can't re-create; it's physically impossible. So I would never make the claim that it will be the same. Having said that, a lot of times that isn't important. Obviously, if you're talking about guitar players and recording studios, frequently the guitar amp is in a booth somewhere or covered with gobos or blankets or whatever [anyway]."

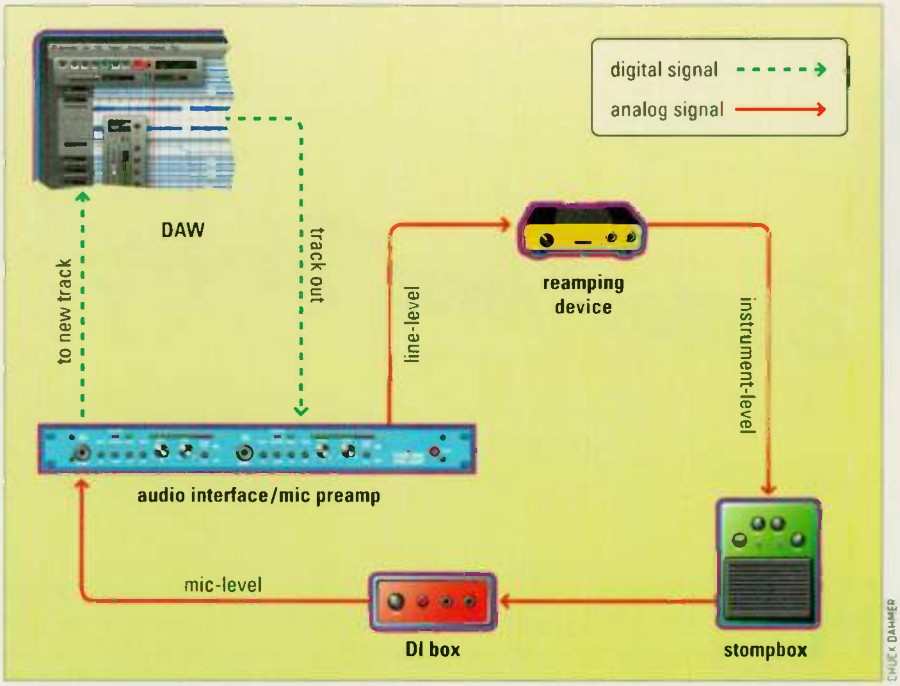

FIG. 2: In this case, the DAW track, after passing through the reamping device, goes not through an amp, but through a stompbox effect. Note that you have to go back into the DAW through a DI and a mic pre.

Once the reamping device is patched in and you hit play on your DAW, the signal for the target track goes out of your audio interface or console, through the reamping processor, where it's converted to instrument level, and then into the amp or stompbox or combination thereof. Simultaneously, you use one or more mics (or, if you're using a processor and no amp, a direct box) to record the newly amplified signal back into your DAW, where you can use it either in place of the original track or to supplement it (see Fig. 1).

One caveat: make sure the original DI track gets recorded noise-free. Any noises on it will be increased substantially when that track goes through an amplifier.

> "Sometimes we want the bridge to have a completely different sonic footprint."

Take Your Pick

Along with bass, guitar is probably the most commonly reamped of all instruments. As with bass, if you want the option to reamp later, you need to record a DI version of the performance—that is, you need to split the guitarist's signal during the tracking session using a DI box, with the XLR output going dry to the DAW and the DI's ¼-inch pass-through jack sending signal to the guitarist's amp, which also gets recorded. (You never know; you might end up with a sound you like, and therefore not need to reamp later.)

The clean DI track is essential as your reamping source, because you don't want to reamp an already amplified track. "It's hard to send a miked track out through an amp and have it sound like a guitar again," says Hamilton. "It gets incredibly blown out." Also, the DI track is like a blank slate, which lets you take the track in any tonal direction you want when you reamp it.

Some guitarists may be touchy about having their tone changed after tracking. "At times, I've had guitar players kind of give me a sour look if I want to take a clean signal," says Nichols. "It's like, 'What's wrong with my sound?'"

For the most part, though, Bottrill finds that guitarists are appreciative of the additional

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possibilities. "Guitar players are geekheads like everybody," he says. "'You have some more amps for me? Let's plug 'em all in.'" Engineers and producers certainly like the options that reamping provides. "If there's something going on in the song that requires a sound that's a little more driven or a little cleaner," says Hamilton, "or we all determine that it should be a touch more aggressive, at least we have the option, without having to retrack live. We're going to retrack, but from a previous performance. What I like is that in a way, it [reamping] separates the performance from the timbre" (see **Web Clip 1**).

ONLINE BONUS MATERIAL

"A lot of times," Antonell points out, "what's perceived as good for the basic tracks doesn't always work for the final mix." According to Cuniberti, it's a simple matter of prudence: "What you do is that you essentially take out what I would call an insurance policy where, yeah, you go ahead and record the amp, and you can put a microphone on it, but you can also simultaneously record it direct."

Another advantage to reamping a guitar (or bass) after the fact is that you can minimize leakage during a tracking session involving other musicians. If leakage is an issue, consider not recording the guitarist through an amp at all during tracking, and instead sending the DI guitar signal through an amp-modeling plug-in (meanwhile, you're printing the clean DI signal). That way, the guitarist can get an amplike sound during the session, but afterward, when leakage isn't an issue, you can reamp the DI track and get the precise tone you want.

With the proliferation of good-sounding amp-modeling plug-ins, you could dispense with the reamping processor entirely and generate the new guitar tone totally from software. If that gets you the sound you desire, it's a simpler option. However, many engineers and recording guitarists feel that the sound of a real amp is superior in plenty of situations.

"I can't pull the microphone back 6 feet from [Line 6] Amp Farm or [Native Instruments] Guitar Rig or have a unique acoustic space affect the overall timbre," says Hamilton. "Again, to suit the track, sometimes we want the bridge to have a completely different sonic footprint than the solo does, and I run out of colors with Amp Farm."

Cuniberti suggests that when recording a band live, splitting the guitarist's track through a DI and capturing a clean track in addition to the amped one is also a good idea. "If there's a mistake made, it's certainly a lot easier to bring the guitar player back in and just play over that mistake on the direct guitar track," he explains. "Assuming that you have the same direct box that was used live, or something similar, you would be able to punch in and fix that little spot in the direct domain. Once you've matched

Reamping Tools

Here (in alphabetical order by manufacturer) are some of the most commonly used reamping boxes. Some are dedicated devices, while others are multifunction units with reamping capabilities.

Creation Audio Labs' MW1 Studio Tool ($1,350 direct) is a combination DI and reamping device that offers variable impedance, clean boost, flexible signal routing, and more. creationaudiolabs.com

John Cuniberti's Reamp ($199 direct; see Fig. A) is the original reamping device and was designed by John Cuniberti. Many of the other manufacturers of reamping devices license his design. It uses passive circuitry, including a transformer. reamp.com

FIG. A: John Cuniberti's Reamp box uses passive circuitry.

The Little Labs Red Eye Recording Tool ($250 direct) can switch between two functions: a passive reamping device and a direct box. The unit offers expansion capabilities, allowing it to be daisy-chained with other Red Eyes. (See the review of the Red Eye in the August 2003 issue of EM, available at emusician.com.) The **Multi Z PIP** ($625 direct) is an active DI, preamp, and reamping device. The **IBP Analog Phase Alignment Tool** ($600 direct) includes a reamping output as well. The **PCP Instrument Distro** ($1,100 direct) offers guitar splitting, reamping, and more. The IBP and the PCP were EM Editors' Choice Award winners in 1999 and 2003, respectively. Reviews can be found at emusician.com. littlelabs.com

The Millennia TD-1 ($1,795) is much more than just a reamping box; it's a full-featured tube/solid-state channel strip. Among its varied I/O are two reamping outputs: one to emulate the characteristics of Les Paul pickups and one that does the same for Strat pickups. (See the review of the TD-1 in the November 2005 issue of EM.) mil-media.com

Radial's X-Amp ($199.95) is a Class A active reamping box that offers two outputs for driving two separate amplifiers. The **Pro RMP** ($99.95) is a passive model with a single output. The **JD7** ($999) is a combination reamplifier and guitar-signal distribution system that lets you feed up to seven amps at the same time. radialeng.com

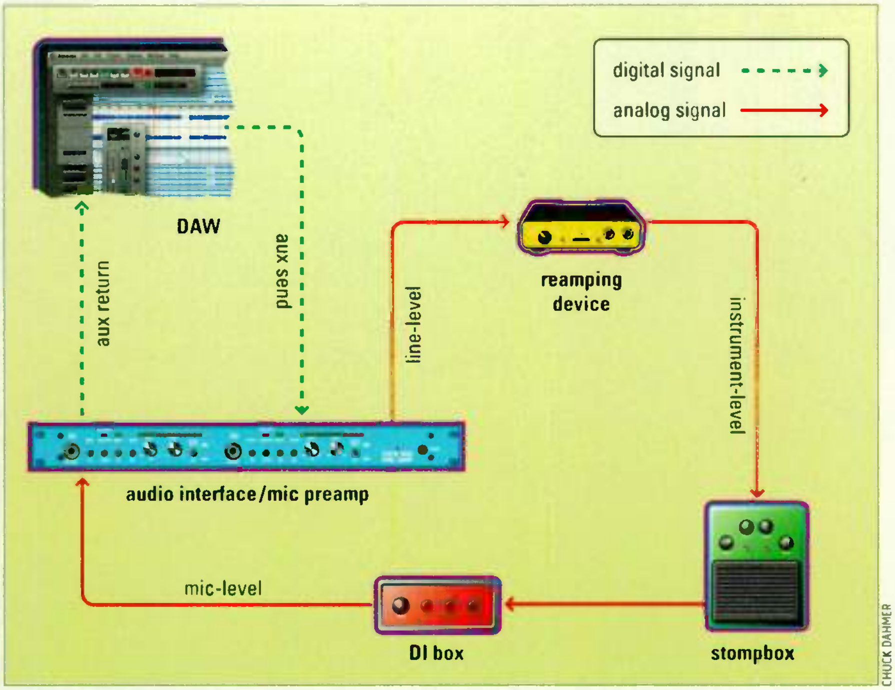

FIG. 3: You can also use a stompbox or other instrument-level effect on an aux send and return, as shown in this diagram.

up the direct signal for the punch-in and the cleanup section, then you can take that entire thing and run it out through an amp and rerecord the original performance."

Walker will often reamp, albeit without an amp, to get the sound of tape on a digitally recorded guitar track. "A lot of the time, if it was recorded dry, I'll send that signal out [through a reamping box] and back into the tape echo and bring it back in," he explains. "I'll set the tape echo to the shortest setting, and I'll get rid of the dry guitar and use only the effected signal. Then I'll time-align it so it fits in the track where it was supposed to be. That way it sounds like a warmer, lo-fi, crunchier guitar."

Reamping also allows you to go for some completely out-there effects. Nichols describes a particularly unusual one, which involves reamping a guitar through an amp stack, with the speaker cabinet placed under a piano. "You set the 2 × 12 [speaker cabinet] underneath a piano, pointing at the soundboard. And then, depending on what notes are played, it gets all these freaky sympathetic chords and dissonant stuff—octaves and weird stuff coming through the strings. So you'd stick some mics in the piano as if you were recording piano, but instead you're playing a guitar through the bottom of it, and you get a total freak show. It's not something you would probably push up in the mix, but you can definitely make a real swimmy sound, which is pretty cool."

Getting Down

Bass guitars were the first instruments to get reamped, back when the process was new. "For some reason," says Hamilton, "bass players are more hip to the idea of DIs and they don't cringe like guitar players do. That's the less sort of exotic procedure, because everybody knows what it's about, to take a DI of the bass."

> "Bass players are more hip to the idea of DIs."

"If you don't like the sound [of the bass amp] or you're not sure about it or it's leaking all over the drums, forget about it," advises Cuniberti. "Just record a DI. It's perfectly usable for the tracking session, and for most of the overdubs, you don't have to worry about it. Later on, run it out through an amp—it can even be a small guitar amp, it doesn't have to be a bass rig—and there you can re-create your amp sound" (see **Web Clip 2**).

Key Factors

Keyboard players have so many sounds and so many audio-processing options at their disposal, through the world of soft synths and samplers, that you might think they'd never have need to reamp their tracks. But even with all those options, there are times when reamping can really help.

Skinner, a keyboard player, frequently reamps his plug-ins. "If it's a sample like a Wurlitzer or Rhodes, it [reamping] really gives it a lot more edge," he points out (see **Web Clip 3**), "and it gives you distortion when you hit it hard."

It makes sense when you think about it, considering that vintage keyboards like Wurlitzers, Rhodeses, and Hammond organs are all instruments that listeners are accustomed to hearing played through amplifiers.

According to Skinner, some soft-synth sounds benefit greatly from being run through an amp. "The waveforms on most digital emulations of analog synthesizers just don't have the complexity," he says. "I usually find there's something missing. When you run something through an amp, it's really generating more overtones. So I'll do that to spice up the sound a little bit. You don't hear actual clipping, but it becomes a more complex sound" (see **Web Clip 4**).

Walker finds that reamping keyboards often helps them sit better in the mix. "I would normally heavily and drastically EQ them," he says, "to take out all these crazy, massive sounds that sound great when you're sitting in your bedroom playing on a keyboard, but [not] when you try to fit it into a mix with a bunch of other instruments." Reamping them through an amp with a 12- or 15-inch speaker, he says, "kind of EQ's those sounds to sit in the mix."


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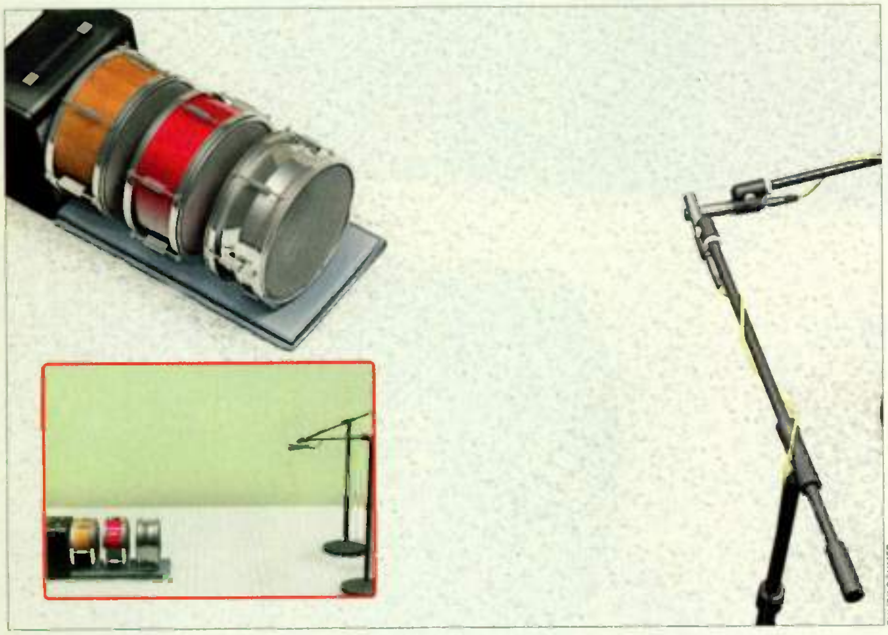

FIG. 4: Hamilton's snare-drum reamping trick. The sound pressure of the snare hits coming through the amp sets off the snares in the room, which are miked back into the DAW.

Pushing the Pedal

Beyond guitar, bass, and keyboards, virtually any recorded track can potentially benefit from being reamped. Antonell recommends a few more possibilities, including "getting a little more presence out of a vocal, making a blues harp sound a little more bluesy, or putting a little more grit on a vocal effect or a solo instrument."

Walker uses the same technique with the tape echo mentioned previously for guitar on vocals as well, or sometimes he'll use an analog tape machine instead of the Echoplex. "I'll amp the vocal out to that, or the guitar out to that, when I want, and set it on the repro head, and run that back in to get a different kind of character," he says. "And just like the Echoplex vibe, you can mess around with it in [Digidesign] Pro Tools to either time-align it so it replaces the original or use it as a slapback, and you can move it so you can set your own variable delay."

Nichols sometimes reamps through stompboxes without using an amp. The setup is similar to Walker's tape echo example. Send the track out of the DAW, into a reamping box, and into one or more stompboxes. Bring it back into the DAW through a DI and a mic pre (see Fig. 2). Nichols gives an example: "If you have a really clean Wurlie that you took direct and wanted to put some sort of phaser on it, you wouldn't need an amp." Such an approach, says Bottrill, allows you to "use the pedal at the level it's intended. You have the proper impedance and the proper level going to the pedal. If you just throw it back to a line input, then it just ends up being sort of a distorted mess sometimes."

Yet another option for using a reamping device with stompboxes or other instrument-level processors is through an aux send. In that case, you'd go out of your aux (either from a console or from an audio interface, depending on your setup), into the reamping device, into the stompbox, into a direct box, through a mic pre to get it back to line level, and then back to the aux return (see Fig. 3).

Supersize Your Loop

The reamping process is also very effective for adding life to dull or two-dimensional-sounding loops or programmed MIDI tracks. There are a couple of different approaches for reamping such tracks. One is simply to play them through your studio monitors, and capture them back into your DAW with a couple of microphones. (No reamping device is needed for that.) Obviously, the size and quality of the speakers and the sound of the room will come into play here. You could mic each speaker individually or use a coincident stereo configuration such as XY.

Antonell, who has the advantage of using the live room at the Clubhouse, likes to liven up tracks by running them through his Genelec 1031 studio monitors. "I'd use a stereo pair like [AKG] 451s about 10 feet away," he says.

He likes to liven up not only drum loops this way, but MIDI strings, too. "If somebody has a cheesy string sound and wants to make it better, I'll play it into the room and mic the room, and turn the direct sound lower," Antonell says. In other words, he's layering the sound of the reamped strings with the original track (see the section "Mirror Images").

Another approach to "decheesifying" loops and MIDI tracks is to do what you would with a guitar part, and send the offending track out to a reamping device and then through an amp, and capture it back to your DAW through mics.

Nichols says he might do that when a track "just won't sit right or is too clean and kind of cheesy. I'll run that through a guitar amp, with vibrato on it or any kind of weird stuff, and just smash it," he says. "Or maybe bring it back and run it through an old compressor and just squeeze it and get an interesting sound."

Beat It

Reamping a snare drum track is yet another application. Hamilton uses a pretty interesting setup for that. "Lately my favorite trick has been to put the snare through a little Fender Pro Junior and then put five or six snare drums on their side on a moving blanket, all just tuned randomly," he says. "And then I have a pair of stereo room mics picking up that racket" (see Fig. 4).

The sound pressure of the snare drum beats coming through the amp triggers the other snares to sound. "The duration of the pulse is exactly a snare drum length," explains Hamilton. "One thing that it helps me with a lot is if somebody has an incredibly poorly tuned snare and I'm presented with it in a mix situation, and there's a crazy ring right where the body of the snare is, something I can't notch out easily."

If the original snare track has too much leakage from the other drums to cleanly trigger the snares in front of the amp, Hamilton sometimes sends the track through a MIDI trigger, which keys a snare sample. The sample is sent through the reamping device into the amp. "I just get the sound of the snare sample

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coming through the amp perfectly clean," he says. "And acoustically that's keying the three or four snares sitting in front of the amp."

Mirror Images

Although many reamping scenarios include the replacement of the original sound with the reamped tone, it doesn't have to work that way. You can also add the reamped tone to complement the original one. This additive approach allows you to layer the original and reamped tracks, giving you a bigger sound.

In such an additive scenario, it can often help to move the mics back from the amp when recording the reamped track. "When I'm talking about keeping an original sound and adding to it, I usually put some distance between the mic and the source," says Hamilton. "It sits better with the original. It's like putting the original thing that came out of a box or out of a plug-in into the real world, and the real world has space around it."

He points out that if you're keeping the original track, you have more latitude to be experimental on the secondary, reamped tone. "You can kind of mess with where you point the mics and what you're willing to accept and how much you want to EQ the return. You can go bananas with it because it's not the primary sound in your mix."

Layering through reamping does not yield the same effect as layering through replaying a part. "Because you're working with the same performance, you wind up with what just sounds like a multiamp single performance," Hamilton says.

One thing to watch out for when using the additive approach is that your newly reamped track isn't out of phase with the original. Hamilton notes that phase is a problem particularly when adding multiple layers of reamped tracks. "If you're printing the same exact performance through the same exact mic position through the same amp and everything," he says, "you're going to have funky phase issues going on." By zooming in on the original and newly added tracks in your DAW, you can slide the latter to get it in phase. "You can look at it, find out where your peaks are, where your transients are, and put it in the right spot," says Bottrill.

Into the Mix

Clearly, reamping is a very useful creative tool that gives you greater control over the sound of your tracks, after they've been recorded. The applications described in this article are just some of the options available through reamping. If you experiment with the process, you're sure to find even more ways to use it.

(Check out the **online bonus material** at emusician.com for additional reamping techniques from Bottrill and Walker.) em

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Mike Levine is EM's executive editor and senior media producer. He hosts the twice-monthly Podcast "EM Cast" (emusician.com/podcasts).


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ELECTRICAL ENGINEERING

Steve Albini's no-nonsense approach to recording.

By Rich Wells

Over the course of his 25-plus years of studio work, punk icon and longtime recording engineer Steve Albini has recorded literally thousands of records in facilities around the world. To those who are unfamiliar with his work, he may most easily be described as the engineer who recorded the Pixies' *Surfer Rosa* (4AD, 1992), Nirvana's *In Utero* (DGC, 1993), and the Page and Plant album *Walking into Clarksdale* (Atlantic, 1998).

However, those are but a few bullet points on a very long list of accomplishments that contribute to Albini's renown. His first band, Big Black, along with other independent bands of the early '80s, helped string together the network of clubs and other resources, both in the United States and abroad, that created the underground culture we know today. (He also wrote widely for the underground 'zines of that era.) As a guitarist, he commands a singular sound and playing style, hair-trigger, full of treble, and instantly recognizable. For the music of Shellac of North America, Albini's current band, he grafts this guitar onto an increasingly stripped-down aesthetic and experimental arrangements, which together pleasantly upset the standard power-trio motif.

During practically every other spare minute, it would seem, he records other bands, and his regimen in the studio has remained essentially unchanged. Every recording is an entirely analog process from microphone to master tape, using the best available equipment, and with minimal gimmickry. (Although the studio is equipped with a Digidesign Pro Tools system, Albini himself doesn't use it.) The resulting albums share common aural attributes that have attracted scores of bands over the years: a naturalistic sound similar to that of the best jazz recordings, detailed and dynamic, albeit often with a more thunderous drum sound akin to that of, say, Led Zeppelin.

In 1997, with the completion of the Electrical Audio studios (electrical.com) in Chicago, Albini and his crew had created a world-class facility built to his specifications. The studio building is completely self-contained, with Albini's own living quarters located on-site, rooms available for clients to rent over the course of their projects, and a large lounge and kitchen area. The studios themselves are marvels of acoustics, flexibility in design, and thrift—the oak floorboards, for example, were reclaimed from the facade of the building itself.

The main purpose of this interview was to find out Albini's normal mode of operation in the studio for a variety of standard tasks. Although he touched on a few favored microphones for given situations, this was not the focus. You can find particular mic applications, and more, thoroughly documented on Electrical Audio's Web site.

During the course of the interview, we also touched on details of the studio's design and the building itself, both of which are, of course, integral to the overall sound of an Electrical recording. Additional text, as well as a short video tour of the studio (see **Web Clip 1**), can be found at emusician.com.

ONLINE BONUS MATERIAL

What's the general approach of Electrical Audio?

The vast majority of the records that I make are performance-based records, where you've got a band playing their material essentially the way they would in rehearsal or live, maybe with a few overdubs. But the idea behind the studio is that you can do essentially any project here. We have 48-track analog available. We also have a Pro Tools system, and other external equipment can be strapped in very easily if necessary. We have multicore patch bays so that you can replace the multitrack machines with a Pro Tools rig just by changing one cable. Overall, it was designed to be as flexible as we could make it.

FIG. 1: Steve Albini uses a stereo mic in front of the kit and often mics both sides of the kick and toms.

Let's start with a basic rock-band setup: drums, bass, guitar, vocals. Beginning with the drums, how do you typically mic them?

There's almost nothing that I do all the time. But normally I'll have close mics on all the drums, a stereo mic to pick up the drum kit sound as a whole, and then distant ambient mics [see Fig. 1]. The ambient mics are generally on the floor and triangulated from the seated position of the drummer, equidistant from the drum seat.

As normal practice, I combine multiple microphones for certain sounds. For example, I normally mic the top and bottom of tom-toms. I reverse the polarity of one of them because the two mics are pointed in opposite directions. But I'll sum those to one channel each, so there will be one track for the rack tom, and one track for the floor tom, for example. You just make sure the balance sounds good and then print that balance.

I'll often do that for the bass drum as well. I'll have a batter-side mic and a front-side mic, and I'll get a balance between those that sounds good and record that. I don't often use a snare drum bottom mic, but when I do, I'll do the same thing there.

I tend to use microphones as they were made and choose them accordingly. I'll put a microphone up, and if I like it I'll leave it, and if not I'll put something else up. Occasionally I'll think, "I like that, but there's a little low-frequency rumble in there that's not going to be helpful," so I'll roll that off. Most of the time, though, I'm using things flat, no EQ. In a typical tracking session, I'll probably end up brightening the top mics on toms to get a little more attack out of the toms, probably brighten up the top mic on the snare drum.

Almost any microphone you put on the snare drum is going to sound thick and meaty when it's right up next to the snare drum, but it may not give you enough of the impression of crispness. So I would expect to have to use an equalizer to brighten the snare drum, to brighten the top mics on the toms. But that would be about it.

Also, most of the time, the drummer's in the room by himself—I'd say 80, 90 percent of the time. But it's not as if the drums are recorded and mixed on their own and then the band is added later. My normal working method is to have the whole band play, and then from the first playback, you have about 80 to 90 percent of the record. That way, you can tell very quickly if there's a problem, and if there is a problem, you can stop and fix it before you continue.

What do you use for the stereo-mic setup on the drums?

For the stereo mics, I use Blumlein or M-S setups for a lot of stuff, especially in front of the drum kit.

And the choice between the two is dictated by what factor?

Whenever I'm bored with one, I'll throw the other one up. I don't make a real distinction between them. Blumlein stereo is slightly hollow in the middle, so if I feel like the majority of the drum sound is going to be coming from the stereo microphones, I'll probably use M-S rather than Blumlein. Whereas if I feel like the stereo mic is going to be mainly an addition to the close-mic sound, then I'm more likely to use Blumlein. But that's a real subtlety.

Do you run into phase issues from using so many mics on a drum kit?

It's not really that much of a problem. If it starts to sound weird, I'll just move a mic. I'm not afraid of getting out of the chair and moving a mic. It can be more of a pain with more mics, but that would never prevent me from doing something that I thought was the right thing to do.

What's your method for recording bass?

I tend to treat the bass guitar sound as it comes off the amplifier as "the sound." I often use two different mics on the speaker; one will be a microphone that has a more generous low end, and the other will have more-detailed high end [see Fig. 2]. I'll balance those microphones in the control room to get what sounds like the most accurate picture of the bass sound.

Most of the time, the microphone that favors the low frequencies has a couple dB of compression on it. There's a compressor that I really like on bass guitar, a UREI LA-22. It's a dual-detector compressor—it has a peak

FIG. 2: Albini combines two different mics on a bass rig in order to capture a wide frequency spectrum.

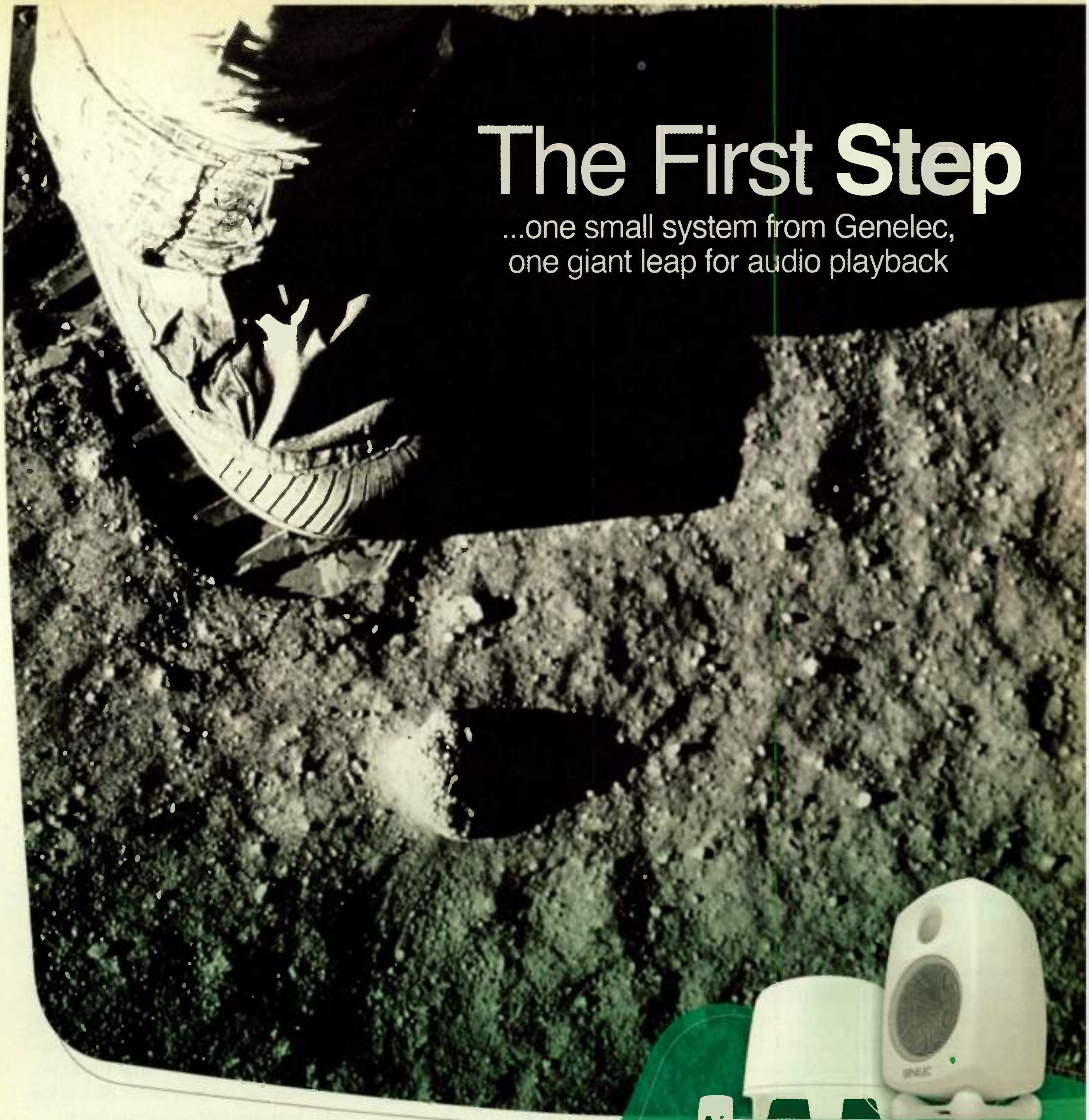

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FIG. 3: Here, a pair of mics are aimed at the middle of the speakers of a guitar cabinet.

detector and an RMS detector—and you can pan between the two to get the most flattering attack sound. I use that a lot on bass guitar, generally only taking a couple dB off, and generally only on the low-frequency microphone.

The reason for that is because in a lot of cases, bass players nowadays have distortion that they fire in now and again. That tends to not have much of an effect on the low frequencies, and in fact distortion often flattens the low-frequency information out. But it can give you a really big spike in the high frequencies and change the texture of the bass. I don't usually compress the brighter of the two microphones so that the effect of the distortion is more evident. It's a way of mimicking the way the bass sounds live, where clicking on a distortion pedal not only changes the sound quality but actually gets louder. Also, pick attacks sometimes trip compressors and make them start sounding odd, so it sounds more natural not to use compression for the microphone that favors high frequencies.

How about guitar amps?

It depends on the setup of the band. If it's a multiple guitarist scenario, I'll try to find out if there's one guitar that's more critical than the other, and I may keep them separate. I don't have any qualms about combining them together, though. If it's a band with one guitar player, sometimes it sounds better if you keep the mics separated and use them as a pseudo-stereo image. If the guitar player has a complex setup with multiple amplifiers, I'll try to keep those amplifiers discrete.

Normally for guitar amps, I just listen to the speakers to see which ones sound best, and pick a mic or two that sound flattering for the speakers. I've noticed that for close-miking, other people generally put the mics closer to speakers than I do. A lot of people take the mic and smash it up next to the grille; I've just had better results with microphones a little bit farther away—say, 8 or 10 inches away from the speaker cone. I'll usually position it on-axis, square in the middle of the speaker [see Fig. 3].

I use ribbon microphones on guitars a lot; in particular, I really like the RCA 74. I've been using that a lot in the past couple years. It's got kind of a slight crispiness to the high end, which I think is due to the fact that it was originally intended to be an announcer's mic. It was basically the budget version of the 44 or 77, which are the big announcer's and vocalist's mics. The 74 is a dinky desktop version, and they might have engineered a peak in the response for articulation's sake. It sounds fantastic on guitar cabinets; I use that mic all the time. I also use the STC [or Coles] 4038. That's also a fantastic microphone [see Fig. 4].

Also, ambient mics sound really good with guitar, even in a dead room. Sometimes just having 10 or 12 feet of distance from the cabinet makes it sound a little more convincing, more familiar. When you're playing guitar, you're not normally listening to it with your head leaning up against the cabinet. If it's a distant mic on a guitar cabinet in a live room, I'll use a condenser mic, just because it picks up the sound quality of the room a little better. But if it's just a mic that's picking up a distant signal in a dead room, I'll normally use a dynamic mic. Sometimes I'll even just use the talkback mic that's set up for the guitarist.

And for acoustic guitars?

Normally I'll have a mic on the bass side and another on the treble, very close to the player's picking hand. Occasionally I'll have a stereo microphone out in front of the instrument, sort of from an audience perspective. The mics by the picking hand tend to sound more like the guitar does to the player, and the mic out front tends to sound more like it does in the audience. For more-detailed guitar passages, I'm more likely to favor the microphones close to the player's hand. The more strummy and noncritical the playing articulation is, the more likely I am to favor the mic out in front.

Generally I'll have the person play a bit and I'll move my head around until I find a spot that sounds good and stick the mic there. I often use either the Neumann SM2 or the AKG C24 stereo mics for that. For the close mics, I tend to use small-diaphragm condenser mics. The Schoeps M221 is a favorite for that. I also like a couple of Lomo microphones; Lomo was a Russian company that made interesting microphones in the '50s and '60s, and there's one called the 1918 that's particularly good for acoustic instruments.

How do you typically treat vocals?

If I'm in a situation, for example, where I have to do four vocalists in the last hour and a half of the session, then I'm just going to put up some sort of bog-standard vocalist microphone, like an AKG C12. I can just open the fader and it'll be okay. It may not be ideal, but it'll be okay.

Practical considerations weigh very heavily in a lot of my decisions. A lot of the bands I work with have extremely limited budgets; they're sort of spending their rent money to come and make a record at all. As a matter of simple human decency, I don't want to waste their time. I don't want to indulge myself in experimenting and trying to find some nerdy, perfect thing when the meat-and-potatoes thing is fine. [For a discussion of one such nerdy, perfect thing, see Albini's take on the Josephson C700A in the online bonus material at emusician.com.]

Does the approach ever differ from that?

Sure. For example, when I did the Stooges sessions [*The Weirdness*, Virgin, 2007]. The Stooges have always recorded with Iggy singing live along with the band, so that's the way we did the record. Iggy was set up in an isolation room, he had an [Shure] SM58 and a [Neumann] U48, and he could either go for the classy U48 sound or the onstage SM58 sound. He had a monitor playing his vocals back at him in the room. There were vocal speakers

FIG. 4: Just a few of the microphones in Albini's collection, which includes transducers from AKG, Josephson, RCA, Neumann, and STC/Coles.

scattered around the rest of the band. So everyone could hear Iggy through the P.A., like always. That's not as common as overdubbing the vocals later, but I'm perfectly comfortable doing that if people want to do that.

Let's switch over to the mixing process. First, what do you use for monitoring playback?

I listen mostly on near-field monitors, and I've gravitated toward B&W near-fields. They're fantastic speakers. I can work on them all day and my ears never hurt, and that's not true of almost any other near-fields. My experience with monitors has been that after a very short amount of time, you get acclimated to whatever you're listening to. The choice of monitoring doesn't really matter in terms of whether you can make a good record with them. It matters much more whether or not they're going to be a nuisance. Like, if you turn them up to a comfortable listening level and you're clipping them, or if they have an irritating sound quality. Or if there's beaming in the high frequencies and where you move your head changes the sound quality a lot; that kind of stuff.

The choice of near-field monitors is mainly a matter of practicality. Genelec near-fields sound fantastic, for example; I enjoy listening to music on them, but they handle complex material very well. They tend not to make me think that there are problems that, intuitively, I think might be there. I'll listen to the Genelecs and think, "Oh that's fine," and it sometimes makes me a bit lazy.

The B&Ws, on the other hand, are just nice, neutral, "nothing special" speakers. But they can handle whatever abuse I can throw at them. I can play them at any volume I can stand to listen to them, and the speakers are going to be fine. And they're not fatiguing at all, which is an enormous benefit.

For mixing in general, you like to combine signals and keep the overall track count low. Does that mean you can usually stick to 2-inch 16-track?

Yes. For most of the sessions I do, 16-track is an ideal format. It's the perfect balance between flexibility and sound quality. For really involved sessions with a lot of extra musicians and a lot of overdubs, no, but for a straightforward recording of just about any performance ensemble, you can probably do it on 16-track. And the sound quality is outstanding. Having said that, 24-track doesn't suck.

As far as mixing is concerned, I've always treated it as an extension of the recording process. I don't record stuff that I know sounds bad thinking that I'll make it sound good later. And I don't record stuff that I know is useless thinking that I might erase it later. A lot of people record stuff not knowing whether they want it or not, thinking that they will have an infinite amount of time later to sort it out at the mixing stage. But I know that at the mixing stage, you're going to be making a million critical decisions.

So why record five microphones on the guitar if only one of them sounds good? Or why do ten versions of something and leave them all sitting there when you know that in the end you're going to have to pick one? Just pick one. If you take care of all the trivial details in advance, the more accurate you can be with the more important decisions at the mixing stage.

Like a lot of people, you started by recording yourself on a 4-track. And before Electrical, you had a recording studio in the basement of your house. What are some of the things you've learned along the way about the process of recording?

There's an analogy about recording that came to mind not that long ago. Think of three types of movies: a normal character/content/dialog movie; a super-high-tech movie, like *The Matrix;* and something technically bone simple, like *The Blair Witch Project*. Those three kinds of movies pretty much cover the spectrum of the technology of moviemaking.

Now, not every movie can be made on a camcorder. But a lot of movies can. If you took a character-driven film and made it more simply just by using camcorders instead of with big Hollywood production values, you wouldn't lose much of the movie. You'd still get the important aspects. But if you took a movie like *The Matrix* and tried to fake it with a camcorder, you wouldn't convince anybody.

With audio, it's similar: it's a matter of trying to make your production environment suitable for the music that you're recording. If you're recording music that has certain technical demands on it, and you can't satisfy those demands, you should move the job to another studio. Just be honest with yourself.

As far as the basement studio, it's where I learned how to do almost everything I do now. The whole attic of the house was a control

room; the playing areas were in the basement. There was a dead room and a live room, and, while they were quite modest, they both sounded good. But if I tried to do a session with 20 people there, it would be insane. It would be completely inappropriate.

The critical factor for any kind of small studio setup, though, is that no matter what you have, make sure you can get the most out of it. Before I had a 24-track machine, when I was first doing sessions in the basement studio in my house, I used an 8-track machine and a 16-channel board with four subgroups. I got the most out of that desk because I had to be inventive, I had to figure out how to solve problems one after another.

I think it would be good practice for anyone to start small. Start with a 4- or 8-track machine and a couple microphones and a modest mixing desk. Work your way up piece by piece, and learn everything as intimately as you can. And as you gradually accumulate equipment and gradually improve your recording environment, you'll have that same level of comprehension of everything in your studio, because you didn't try to attack it all at once; you've learned it one thing at a time.

When I was first starting out, if I had had a 36-channel board and a 24-channel machine and a raft of microphones and outboard gear and assistants and stuff like that, I would never have learned how to be as resourceful as I have. And I think that that resourcefulness and that willingness to solve problems in unconventional ways, that helps a lot when you go into bigger environments. There, you may be confronted with fewer issues, but the problems you do encounter generally require abstract thinking to solve.

It's very good to get yourself in the frame of mind that you can solve problems. Once problem solving becomes second nature to you, you'll know that you can work through any issue. Rather than coming at it from the mind-set of "There is a solution that I need to ask somebody for, and then I can do it," you're accustomed to thinking things through yourself. em

Rich Wells oversees the Supreme Reality, a recording studio in Portland, Oregon. Visit his Web site at thesupremereality.org.

Going with the Grain

Ten granular-synthesis programs to slice and dice your sounds. | By Dennis Miller

Judging by the number of hits you get when you Google it—several hundred thousand—granular synthesis would appear to be a very hot topic. You can also measure its popularity by looking at the large number of programs that either are dedicated to granular synthesis or include it as a feature. Though it's not the best way to produce cutting lead lines or thumping bass parts, granular synthesis has a vast range of musical uses, including slowing down sounds without changing their pitch, adding reverb or other ambient qualities to a sound, and, of course, producing all manner of amorphous sonic textures.

Granular synthesis is a technique intended to create clouds or clusters of small sonic events called *grains* (see "A World in a Grain of Sound" in the November 1999 issue, available at emusician.com, for an introduction to granular synthesis). Grains are typically in the realm of 5 to 100 ms long and are created from either synthetic waveforms or samples. When clumped into massive groups, individual grains are virtually indistinguishable, but the overall impression can be like rain or falling rocks, or more rhythmic, even pitched sounds, perhaps with recognizable bits of a source sample popping through.

In this article, I'll cover representative programs from the world of granular synthesis. Many dozens of programs support the technique, and the ten chosen for this roundup are just examples of what you'll find if you go looking. I'll focus only on software, acknowledging that hardware-based systems such as the Symbolic Sound Kyma System are extremely capable in this area. There are also hundreds of Native Instruments Reaktor patches that employ granular synthesis, not to mention patches designed for Csound, James McCartney's Super Collider, Cycling '74 Max/MSP, and other programming environments, so be sure to check out their users forums if you own any of those programs.

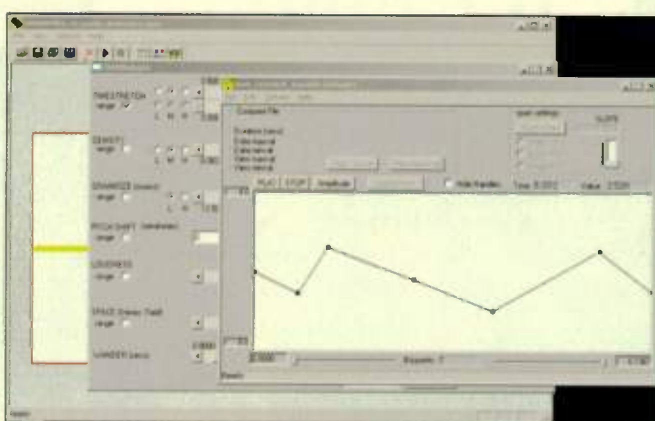

FIG. 1: GrainMill offers a window for building multisegment envelopes to control any of its parameters. The envelopes can be saved and reused.

The programs I'll look at are Sinan Bökesoy's Stochos V6, CDP GrainMill 1.1, Karlheinz Essl's REplay PLAYer 3.2, Nicolas

Fournel's Granulator 1.1, Tom Gersic's Atomic Cloud 1.0, Nikola Jeremic's Organik 1.2, Christopher Keyes's Granular Cloud Generator 2003, LowNorth RTGS-X 2.4, Stefan Smulovitz's Kenaxis 2.2, and Jörg Stelkens's crusherX-Live 3.51. (See the sidebar "Manufacturer Contacts" for contact info on each program and the sidebar "Other Options" for a list of some granular apps that are not included in this article.) I won't go into detail on the sonic results each can produce, so be sure to check out the Web Clips and, where available, the demos for each program to get a sense of the sounds it can make.

In View

The granular-synthesis programs covered here share a number of features, though as you might expect, each also has its own take on the technique. All of the programs except GrainMill produce audio in real time in response to user input, and all except Stochos can record output to disk. All run as standalone applications, but Organik also includes a VST plug-in version. (CrusherX-Live includes a VST Bridge component that you load in a host if you want to use the program as a plug-in.) All but Organik let you load samples for granulation, and crusherX, Granular Cloud Generator, Organik, RTGS-X, and Stochos include one or more synthetic waveforms for use as grain sources. CrusherX, Kenaxis, and RTGS-X allow you to granulate real-time audio input.

Flexible granular synthesis involves controlling aspects of both individual grains and overall granular textures (often called *grain clouds*), and all of the programs have capabilities of those types. For example, you can modify both the pitch (frequency) and duration of individual grains in all the programs. Most of the programs use common and intuitive increments for these two parameters—hertz or MIDI Note Number for pitch and millisecond for grain duration—but Granulator only provides a range of 0 to 127 for both of these controls, with no indication of what precise values that span represents, and Atomic Cloud's controls read simply "0 - 100."

The range of available grain durations varies quite a bit, with some of the programs providing a more limited range than others. Organik, for instance, tops out at 80 ms for maximum grain duration, while Kenaxis and Stochos have no limit on the length of grains. GrainMill's maximum duration is 3.2 seconds,

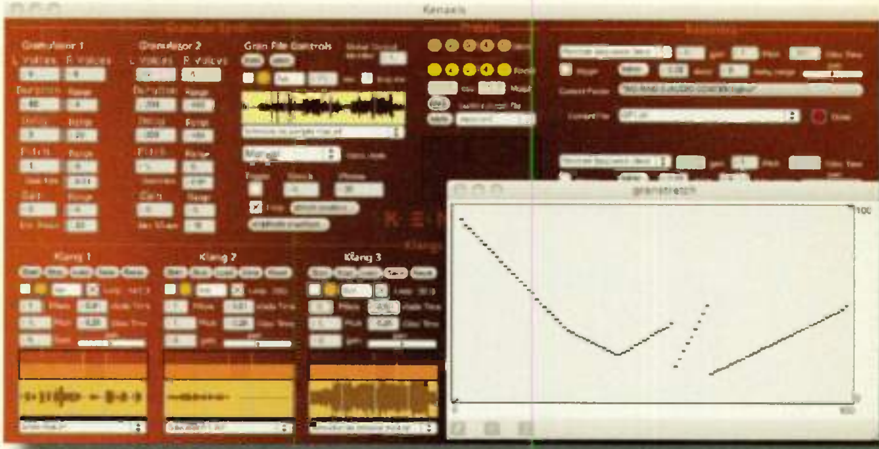

FIG. 2: Kenaxis has a very powerful and flexible system for controlling all aspects of sample granulation. Included is a two-dimensional graph (lower right) for drawing the sample's playback trajectory.

but the actual size you get depends on the size of the source file (see Web Clip 1). Its minimum size is fixed at 12.5 ms. As for all its parameters, GrainMill lets you apply a simple linear envelope to vary the value over time or build complex multisegment envelopes for that purpose (see Fig. 1). Note that grains of 1 second or more tend to retain the characteristics (pitch and timbre, for example) of their source. That could be an appealing creative option if, say, you're working with speech and want to retain some semblance of the original text.

ONLINE BONUS MATERIAL

You can choose what type of synthetic waveform you want for grains in several of the programs. Cloud Generator, for instance, includes only a sine wave (by far the most common), while Organik lets you choose from sine, saw, ramp, square, or pulse waves and, even more useful, create your own using a different 31-harmonic, user-defined additive-synthesis waveform for each of its four Generators. Stochos also supports additive synthesis as well as pulse-width, ring, and frequency modulation, and crusherX-Live offers sin, cos, triangle, rectangle, and random waveshapes, and a window for drawing and saving your own.

Zeroing In

Picking the region or range from which grains will be extracted from a sample is an important feature, and most of these programs offer flexible options in this area. Atomic Cloud, for example, has a Scan Rate slider that lets you determine how quickly the program will move through a sample and generate new grains. CrusherX has sliders for setting the region from which each of its four separate samplers will pick grains, and REplay PLAYer provides several modes (Walk, Jump, Regions, and so on) with adjustable parameters for the same purpose. GrainMill offers a Wander parameter to determine how random the process of picking grains will be: a value of 0 specifies that grains should be chosen in strict order, while a value of 100 specifies that grains should be chosen randomly from anywhere in the file.

RTGS-X provides a Buffer Position slider that can be automated to scan through a file at any speed, forward or backward, or to pick from random points, and Stochos can play back all or only a set range of a file in a variety of directions. Kenaxis has numerous flexible options for determining the portion of a sample that will be used, including a two-dimensional graph on which you can draw the playback trajectory (see Fig. 2). You can easily build a path that moves forward through the file quickly, then slows down, then reverses direction. (Graphs can be saved for reuse.)

Grain density, or the time between successive grains, is the parameter that most affects how quickly a massive grain cloud will build up. RTGS-X will produce grains from every 10 to every 5,000 ms, and REplay PLAYer lets you control density manually by using a two-dimensional grid or by applying a preset (Granular_lo and Granular_high, among others; see Web Clip 2). GrainMill's Density feature lets you set the number of new grains relative to the base grain size you've chosen (a value greater than 1 produces overlapping

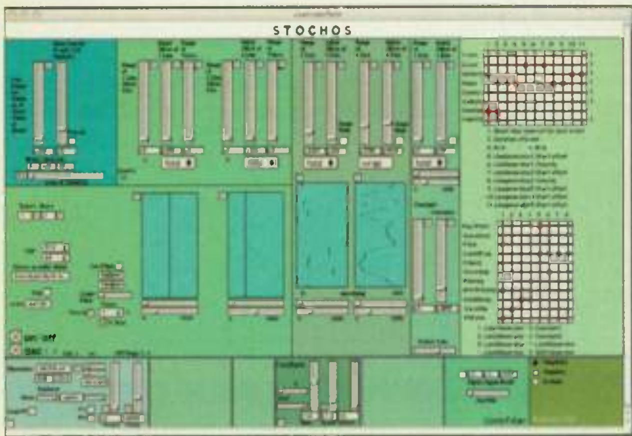

FIG. 3: Stochos uses a modulation grid (right) for assigning control sources to program parameters. Its numerous displays provide real-time feedback for other program components.

grains and a value less than 1 produces separation), and crusherX has a Birth parameter for setting grain density (from 10 to 1,000 ms) and can display a window that updates the actual number of grains produced in real time.

Windowing is the process of applying a smoothing envelope onto individual grains to avoid the clicks that such short-duration sonic events would typically produce—most common is a Gaussian (bell-shaped) curve, and linear fade-ins and -outs are also useful. All of the programs support this feature, and some offer several window types (or, in the case of REplay PLAYer, a Smoothing parameter) to choose from. CrusherX and RTGS-X even let you draw your own envelope shapes.

That's So Random

Because randomness and nonrepetition are such common qualities of grain clouds, you'll find random functions (called "variation," "jitter," "range," "random," or "drunken walk") to be part of many of the offerings. At the very least, most of the programs provide a base setting for their individual parameters, then let you add some amount of jitter or randomness that offsets above or below the base value.

The distribution or probability curves that several of the programs provide are even more powerful than simply adding a touch of random jitter. In addition to Gaussian, you'll find Weibull, Cauchy, Poisson, and drunken walk curves in Stochos and elsewhere, each of which defines different probabilities of events occurring. (Distribution curves typically establish a mean and some variance or standard deviation from that mean. See en.wikipedia.org/wiki/Probability_distribution for more information.) Cloud Generator has various options for randomizing grain frequency and offers controls to transpose grains using both hertz and cents increments.

All of the programs except Atomic Cloud and Organik allow you to automate changes in parameters via MIDI controller messages, internal envelopes, or other means. Several let you use distribution curves to determine parameter values (in addition to using them for added randomness). More typical are internal functions that sweep through a sample file to locate grains or the ability to apply a ramp function to the amount of pitch transposition (in Stochos, for instance). Granulator has a MIDI Controls window where you can map any MIDI controller number to any of the program's parameters, and both crusherX and Kenaxis have dedicated MIDI setup screens where you can assign MIDI data to various parameters and also scale or otherwise process the data before it reaches its target. REplay PLAYer has a hardwired mapping of controllers to program function.

RTGS-X provides a variety of ways to modify parameters over time, including an Envelope Follower, which will track the amplitude of incoming audio and map the values to parameters of your choosing. It also offers a Morph function that gradually moves values between two groups of settings, as well as four multisegment envelopes that you can use to modify parameter values over time. CrusherX also has a very nifty Morph feature that alters the time it takes (up to 3,600 seconds) to apply a change in a parameter once you click on a button or move a slider (see **Web Clip 3**), and GrainMill can extract an amplitude envelope from a preexisting audio file and apply it to any parameter you want.

Organik has a two-dimensional grid where you can morph between the settings of the program's four grain Generators in real time, and crusherX has several grids on which you can control parameters in real time. RTGS-X has what is no doubt the most unusual real-time control source in this group. Its Video Controller window is where you pick an input such as a Webcam or an iSight camera and track an object or a light source in the camera's field of view, then specify what program parameter the *x* and *y* coordinates of the object will control.

Added Attractions

Though the main purpose of most of these programs is granulating samples or synthetic waveforms, you'll find a few bonus features in some of them. Kenaxis, REplay PLAYer, and RTGS-X, for example, let you insert VST effects into their output, and Kenaxis also has an extensive sample-looping engine and a synth section with two oscillators and a noise generator. Organik includes its own reverb and delay, Cloud Generator offers reverb, and Stochos includes several filtering options. CrusherX has a limiter/expander as well as a reverb and stereo delay line, and RTGS-X also has a limiter and 5-band EQ that allows frequency, amplitude, and Q control via MIDI.

[The granular-synthesis programs covered here share a number of features.]

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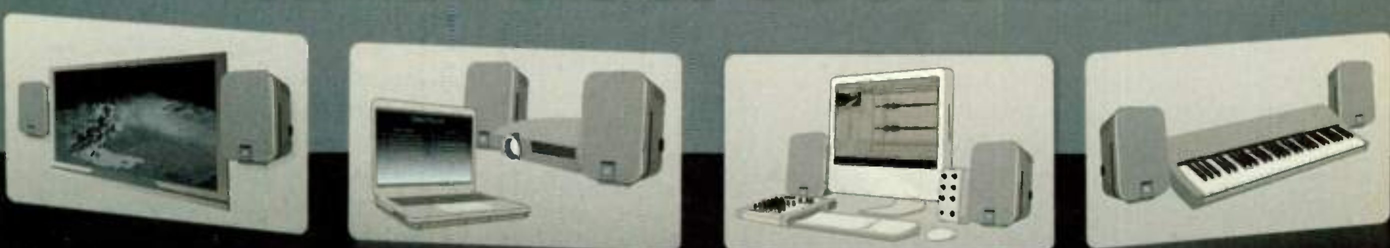

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Be aware that many of the programs were created by small, independent developers. Though they are free or relatively inexpensive, they may be lacking in support or thorough documentation. Moreover, there's no guarantee that any of them will continue to be developed by their creators—some have dates that are several years old. But all told, this collection includes a vast array of features for creating colorful and unique sounds, so be sure to give one or more of them a try.

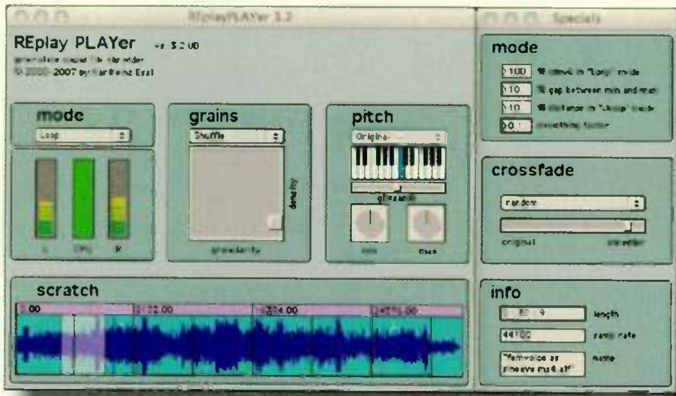


FIG. 4: REplay PLAYer's interface contains the main work areas that you can update in real time. Other controls are available from pull-down menus.

Sinan Bökesoy's Stochos V6 (Mac, free)

Stochos is one of several programs Bökesoy has developed that use granular methods to produce sound. Its control system is based on the interaction of a variety of elements, including a routing matrix where different parameters are matched to one of several different control sources (see **Fig. 3**). The control sources themselves can be configured to produce a fixed range of values or to use random distribution curves. You can also alter all parameter values in real time using sliders for that purpose (see **Web Clip 4**).

Stochos has three options for its basic sound sources: samples (Sampleobj), synths (Synthobj), and an oscillator bank (OscBank). To use a sample, you specify the folder that contains the file you want, pick the specific file, then set the sample's base frequency using a number box. There's a loop switch to enable looping and a range selector to specify how much of the sample will be looped. The synth options include FM, RM, additive, and noise, and you can morph between the various methods over a user-specified amount of time. The oscillator bank has somewhat fewer options than the others and uses Max/MSP's oscbank Object as its source. (You'll need the full version of Max/MSP or a runtime version, available at cycling74.com, to use Stochos.)

Stochos cannot record its output to disk, but you can use Cycling '74's free audio router, Soundflower, for the same purpose. Though that's not as easy as having a record feature, it's well worth the effort to explore this versatile program. You might also have a look at Bökesoy's latest project, a new granular tool called Cosmos, which according to its developer has even more "self-evolutionary" capabilities. If you're interested in the topic, Bökesoy's research in granular synthesis is definitely worth following.

CDP GrainMill 1.1 (Win, about $75)

The CDP toolkit (Mac/Win) is one of the most powerful yet underrecognized sound-processing libraries on the market. It offers several hundred functions, many of which work directly on a sound file and others that perform transformations on phase vocoder analysis data prior to resynthesis. Among these functions are a number that use granular techniques, but unlike those, GrainMill, which is Windows only, runs as a standalone application and is the only sound-processing tool sold separately.

GrainMill's strength lies in the powerful time-varying functions you can impose on all of its parameters. The main screen provides access to a simple Range feature that lets you specify start and end values for any parameter; the program will then randomly generate a value between those limits each time it creates a grain. But far more powerful is the Time Contours area, where you can create very complex multisegment envelopes to control any function. The envelopes can be saved to disk in GrainMill's BRK format, then reused on other parameters in the same session or in any future session. GrainMill also lets you assign extreme values for its parameters. Timestretch (and compress), for instance, supports a range from 4/1,000 to 256 times the original.

If you're interested in some serious sound-altering resources, the CDP library is unique in the range and quality of its offerings. Though you can buy GrainMill by itself, you might consider purchasing the entire tool set for not a whole lot more money (site licenses and student discounts are available).

Other Options

Ambient Grains	ambientgrains.com
AudioMulch	audiomulch.com/info.htm
Chaosynth	nyrsound.com
Cypher	maklott.com/cypher.htm
Emission Control	ftp.create.ucsb.edu/pub/EmissionControl
GranuLab	hem.passagen.se/rasmuse/Granny.htm
Hudak	homepage.ntlworld.com/david.resonant/Pages/downloads.html
KTGranulator	koen.smartelectronix.com/KTGranulator
Maelstrom (in Reason)	propellerheads.se
MetaSynth	metasynth.com
PulsarGenerator	create.ucsb.edu/PulsarGenerator
Real-time Granulator	www.hkbu.edu.hk/~lamer/download.htm
RiverRun	audioease.com
Sonix	jcproductionz.com/vst.html

Karlheinz Essl's REplay PLAYer 3.2 (Mac, $25)

Karlheinz Essl has been quietly producing creative music tools for years, and REplay PLAYer is just one of his many offerings. The program's interface is divided into four main areas: Mode, Grains, Pitch, and Scratch, with additional functions provided via pull-down menus (see Fig. 4). After you load a sample and turn on the audio engine, REplay will begin to generate grains from the source based on its default settings. You can use the Mode menu to choose how the sample will be scanned; options include scrubbing manually, Jump, which skips around in the file using distances you specify, and Loop, which has an adjustable speed setting (from 10 to 100 percent of the original speed).

Move to the Grains menu, and you'll find a 2-D grid for adjusting granularity (grain size) on the *x* axis and density on the *y*. Mouse movements that you make update parameters quickly and feel fluid with nary a glitch, but you can select from one of several presets if you prefer to leave control to the program. Pick the Random preset, for example, and you'll get a constantly changing grain cloud.

The Pitch area provides four ways to control grain pitch, which also include using presets and using sliders for adjusting pitch manually. There's also a virtual keyboard for specifying a transposition up or down two octaves and the ability to set a minimum and maximum range from which the program will pick randomly.

REplay has a feature that lets you crossfade between the original sound and the granulated version (with a random crossing amount, of course), the ability to host up to three VST plug-ins, EQ (also with a random option), and a random panning feature. All in all, it's a hefty toolkit for producing a wide variety of granular sounds.

FIG. 5: Atomic Cloud uses sliders to adjust its various grain parameters. Changes you make while a sound is playing update in real time.

Nicolas Fournel's Granulator 1.1 (Win, free)

Granulator offers just enough parameters to get your grain clouds flowing and could be a good entry point for exploring granulation of sample files. The ability to control every parameter via MIDI CC messages makes it suitable for a live environment and also allows a lot of spontaneity in a studio setting (though a plug-in version would be a huge enhancement in that environment; see Web Clip 5).

Among the different options are controls for grain density and duration, each of which offers a knob for setting a base value and another for random offset from the base. The same pair of controls is available to adjust the grain's pitch and the start point within the sample from which grains will be extracted. A filter with cutoff and resonance settings and an AR envelope with random offsets add to your sound-design options, and a delay with length and feedback controls and a final output stage with gain and pan settings round out the interface. You can also save presets of your settings and adjust the program's color scheme to suit your liking.

Tom Gersic's Atomic Cloud 1.0 (Win, $9.99)

Atomic Cloud's simple and intuitive interface places all the program's parameters onto a single screen (see Fig. 5). The software provides sliders for grain rate, grain size, scan rate, buffer rate, jitter, and amp level. There are also controls for setting the start and end points in the sample where grains will be generated. It's easy to produce effects with a high degree of randomness—a high Jitter value will serve nicely for that purpose—and you can produce backward-sounding grain clouds by setting the End parameter to a value lower than the Start value (see Web Clip 6).

Grain pitch is controlled by the Buffer Rate parameter, which offers a range of 0 to 100 (50 represents the original pitch). Scan Rate, in conjunction with Jitter, controls the rate at which the sample is scanned. Set it to 0 if you want to freeze the sound on a minute portion of your source file.

Atomic Cloud supports both 16- and 24-bit WAV output and offers a buffer-size setting that you can use to configure it for best performance on your system. The program comes with a small number of presets that illustrate several nice effects, and you can create and save your own.

Nikola Jeremic's Organik 1.2 (Win, free)

Organik is the only program in this roundup that can't granulate samples, but it offers plenty of sound-generating options nonetheless. Its single screen includes tabs to access parameters for its four synthetic-grain Generators, each of which can use one of the included waveform presets or employ an additive waveform that you design (see Web Clip 7). To start a project, you set a base grain size and grain rate for one or more of the Generators, then dial in some amount of random variation to those rates. (Note that you can enter values manually that exceed the values you can enter using the sliders.) You then trigger a note using either the onscreen keyboard or an external MIDI controller. You can modify the amount of randomness that will be added to the base pitch you play, or transpose the base pitch using the Octaves, Note, and Fine settings. All changes you make take effect in real time.

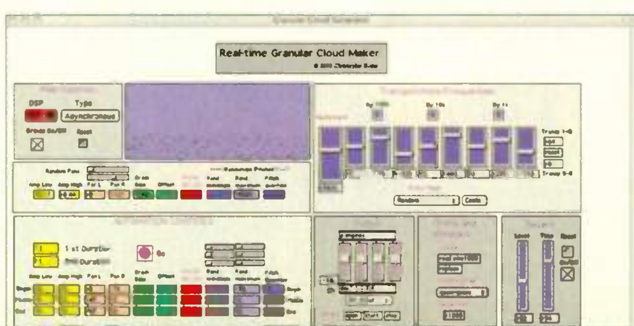
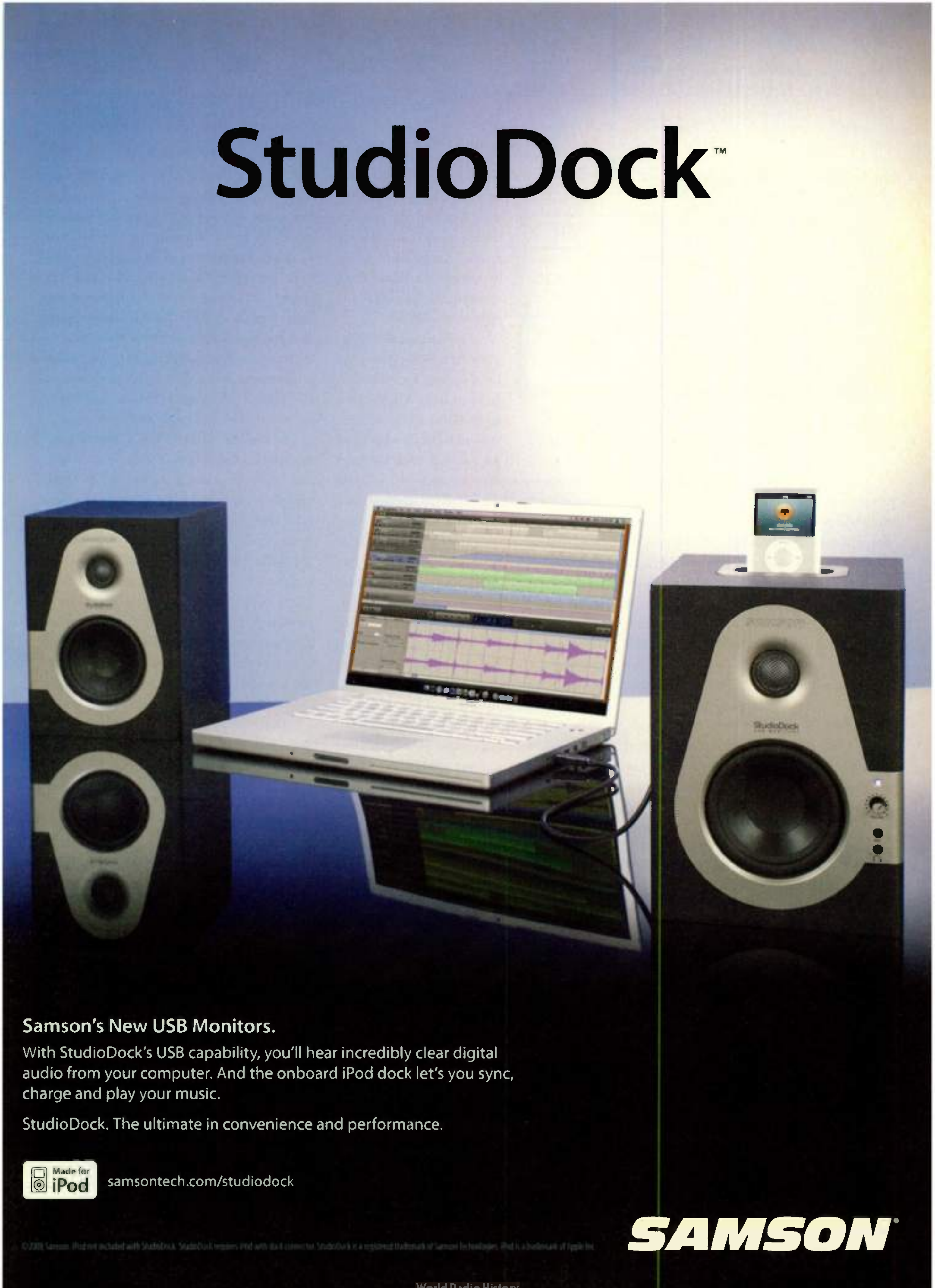


FIG. 6: Granular Cloud Generator supports up to 16 simultaneous grain streams and offers a flexible system (upper right) for modifying grain pitch or frequency.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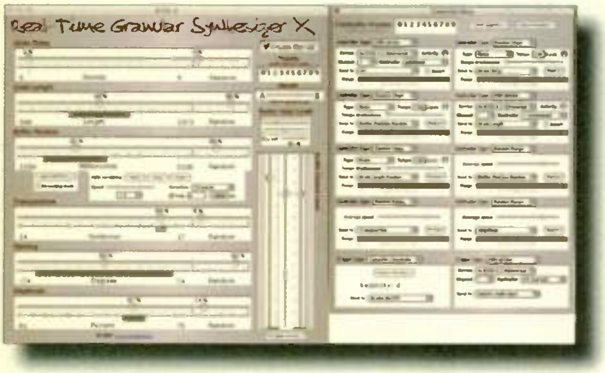

FIG. 7: RTGS-X has a dedicated Controllers Setup window (shown at right) for controlling its parameters in real time via MIDI or other sources.

Each Generator has its own volume and pan settings and a mute option, and you can morph between the Generators using a two-dimensional Cube interface. Movements on the Cube can't be recorded, however. Organik includes very serviceable reverb and delay effects, and there's also an LFO with adjustable shape, speed, and amount controls, and a single ADSR global envelope. You can also sprinkle a bit of noise into Organik's output using the Noise generator, which offers Range, Sparse, and Gain settings for that purpose.

Organik is not the most advanced program in this group, but its simple yet elegant approach is a good place to start if you're just entering the granular universe. At this price, there's no reason not to give it a try.

Christopher Keyes's Granular Cloud Generator 2003 (Mac/Win, free)

Cloud Generator is a straightforward program with some interesting variations on the standard grain-generation practice. It supports four audio output channels and can record its output to WAV, SND, or AIFF format, regardless of which platform you're running on. In addition to loading a sample, the program provides a sine wave with variable frequency for synthetic grains. It will generate grains up to 500 ms in length and can produce up to 16 simultaneous grain streams.

The Automation Controls window is the engine for grain generation. At the top of this area is a single row of grain-generating parameters that you can configure and then use for the task. But just below that is another area where you can assign beginning, middle, and end values to all or only some of the same parameters, then determine how long the program takes to move from one set to another. (It would be nice if the values you set above could automatically be assigned to the start row below.)

The Transpositions/Frequencies window gives you a lot of options for controlling grain pitch (see Fig. 6). If you set the Pitch mode to Hertz, you can then type in or move the sliders to enter the exact frequencies you want for each pair of grain streams (each slider controls two grain streams; see **Web Clip 8**). Choose Cents, on the other hand, and you can transpose the original pitch of the sample in increments as small as 1 cent (1/100 of a semitone). Buttons above the main sliders force the sliders to move in increments of 1s, 10s, or 100s of whichever value you are adjusting. A separate control toggles between synchronous (pitched or semipitched) or asynchronous (nonpitched) grain generation.

The values in the Transpositions/Frequencies area can also interact with the "Rand. minimum" and "Rand. maximum" values you'll find elsewhere. These last two parameters output a new value based on the "Grains per sec." control, which supports grain generation from 1 to 1,000 times per second. You can also set a high and low range for amplitude and for pan position (left and right channels only). Ten different window (smoothing) shapes provide additional subtle but noticeable timbral variations.

LowNorth RTGS-X 2.4 (Mac, $49)

RTGS-X is a robust program for creating granular textures and is well suited for live or studio use. Its Controllers Setup window (see Fig. 7) offers options for routing up to eight different control signals, which could be MIDI controllers or one of two different internal random generators, or two different Triggers (MIDI or a computer keystroke) to a variety of parameters (Grain Size, Grain Rate, and so on). You can record an entire session of slider moves and button presses using the Capture Stream option, then play back all movements at the original or any new speed you want. Pop-up help for every program component keeps your work session flowing, and a thorough HTML manual fills in any missing info you might need.

At the top of the RTGS-X interface are windows to control Grain Delay, Grain Length, and Buffer Position. The first two of these have sliders to adjust the percentage of a base value that you set in the Grains window and are especially handy in real-time performance. The Buffer Position also has a slider that represents where within a loaded sample file grains will be extracted. If you want to automate these parameters, you can create multisegment envelopes lasting up to 999 seconds and use them to control these or other program parameters. You can also map *x* and *y* mouse movements independently to three different parameters.

The Transposition slider controls grain pitch, and, as with the other parameters, you set a base value (in percent) and then apply some amount of random offset (in semitone increments; see **Web Clip 9**). There are also controls for panning and amplitude, with their associated random components.

RTGS-X limits its input to only the first 10

Manufacturer Contacts

Sinan Bökesoy	sonic-disorder.com/research.html
CDP	composersdesktop.com/options.html
Karlheinz Essl	essl.at/software.html
Nicolas Fournel	nicolasfournel.com/granulator.htm
Tom Gersic	atomiccloud.gersic.com
Nikola Jeremic	njeremic.ecobytes.net/njeremic/software.html
Christopher Keyes	www.hkbu.edu.hk/~lamer/download.htm
LowNorth	lownorth.nl/software/products/RTGS-x.html
Stefan Smulovitz	kenaxis.com
Jörg Stelkens	crusher-x.de

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FIG. 8: The crusherX-Live main screen can be configured to show only the areas you want. Its numerous displays show parameter values updating in real time.

seconds of the source file (AIFF, WAV, AU, or NeXT format), and loads only the left channel of a stereo file. Other than that, there's nothing really missing from the program except a Windows version! (Granular-synthesis guru Barry Truax has noted that granulation typically destroys all sense of stereo separation in the original file.)

Stefan Smulovitz's Kenaxis 2.2 (Mac/Win, $145)

Tucked inside a very powerful looping and sampling interface are Kenaxis's two granulators, each of which uses the same sample but has the ability to process it in separate ways. For example, one might generate a high density of grains while moving backward through the source file at half speed, whereas the second plays back a version of the file stretched to several times its length but with the pitch intact (see **Web Clip 10**). You can use the Random File Impulse command to instruct Kenaxis to pick sample files randomly from a folder you designate, or drag-and-drop a new file manually onto the Granulator file window even while the program is playing back. Kenaxis supports only mono files (or the left channel of a stereo file) for granulation but outputs audio in stereo.

Kenaxis controls randomness in a variety of ways. You can set a range within which grain duration, delay, pitch, and loudness will be chosen randomly, and you can quantize the pitch range to limit values to fixed increments (only multiples of 0.33 times the original, for instance). Or you can use the GranRnd window to control grain parameters with a high degree of specificity. Like the KlangRnd window, which controls the playback of sample files, GranRnd uses a drunken walk model to set the probability of events occurring. Both windows include a tempo control that you can use to determine how often different parameter values are updated. You can also create interesting rhythmic patterns by using the Amplitude Envelope window to impose "amplitude

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sequences" on grains at specific tempos.

Kenaxis includes a delay with delay times up to 75 seconds and a filter that you can apply to any of its sound sources. It has an extensive manual and is easily the most configurable of this group—there's even a dedicated window for setting up the response of a joystick or Wacom tablet. In addition to the stereo version, a second version, called Kenaxis VBAP, allows surround output and up to eight channels, and includes a mixer for panning each of the program's sound generators independently.

Jörg Stelkens's crusherX-Live 3.51 (Win, about $222)

What do you get when you combine four samplers and four multiwave oscillators with real-time audio, granular processing, unique effects, and a flexible routing system? That would equal only a portion of crusherX's sonic-mangling resources. The program offers a screenful of sliders and graphical controllers that would keep even the most hyperactive tweaker happy (see Fig. 8). Nearly every parameter has a randomize control, and every program parameter can be put under MIDI control.

Each of crusherX's sound sources can be routed directly (Dry) out to the mixer; to the Vapor modulators, which are the program's primary granulators; or to the Effects section. The Vapor modulators share several parameters that you control in various ways. You can use sliders to control things like a grain's base pitch or amount of modulation or assign those and other parameters to the unique Physical Model X/Y Display controller. This controller is a pseudo-3-D display on which you move a small blue light to change the values that the controller outputs. You can modify the shape of the display and the speed at which the light updates when you move it. You can also modify Vapor values by assigning a sine, square, or various other functions (random, for example). All in all, the modulation matrix is massive, to say the least.

CrusherX supports up to 199 simultaneous grain streams, and you can offset the streams to start in succession, from 0 to 1,000 ms apart. Moreover, each grain stream can be routed to up to ten different multichannel outputs of an ASIO audio interface. You can define the morph time for changes you make in the interface to take effect and also determine how much time it takes to change from the current to new parameter settings when you load a preset. You can also configure the interface to show only the parameters that you want to work with, or use the Newbie mode, which strips out a large portion of the program's features but still gives you plenty. Having a fast computer and a lot of time to experiment is the best preparation to master this very deep program. em

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Associate Editor Dennis Miller composes with music and images. Check out his work at www.dennismiller.neu.edu.

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Side Order

Creative effects processing using the mid-side matrix. | By Brian Smithers

The mid-side (M-S) stereo-miking technique is well known in both classical and commercial recording for its ability to control stereo width while maintaining mono compatibility. I'll show you how to adapt M-S principles to bring that same ability, and more, to your mastering.

Adding It Up

M-S miking is a coincident technique combining a cardioid mic with a side-facing bidirectional mic (see **Fig. 1**). Because the bidirectional mic's null faces the sound source, it misses much of the direct sound and picks up sound primarily from the left and right sides of the stage. Thus, the cardioid mic is the mid, and the bidirectional mic is the side. Increase the level of the mid, and the stereo image narrows; increase the level of the side, and the image widens. Because the two mics are coincident, they sum to mono very nicely.

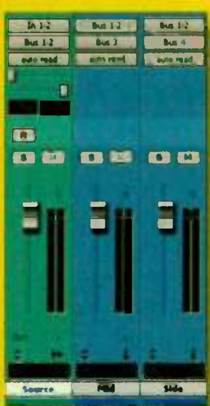
You derive discrete left and right stereo channels by adding and subtracting the two mic signals. That works because when we face the side mic to the left, sound from the right of the stage is picked up by the rear of the mic, pushing the diaphragm when it should pull, and vice versa. Therefore, the right-side signal has its polarity inverted—a negative of the actual sound. That means the side-mic signal is L – R, whereas the mid-mic signal is L + R. Sum those, and you isolate the left side; subtract them, and you isolate the right side.

EZ Wider

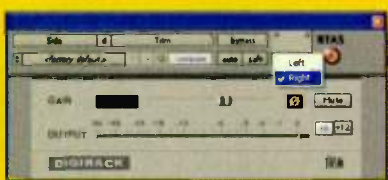
To manipulate stereo width in mixing or mastering, you need to reverse the process and derive mid and side from left and right. Sum the left and right channels in your mixer, and you have the mid channel (L + R); sum them after flipping the polarity of the right channel, and you have the side channel (L – R).

To do that, copy the left and right signals, and invert the polarity (often incorrectly called the "phase") of one copy of the right channel. This is easily done on many analog mixers, but DAWs often lack phase-invert buttons, in which case you'll need to use a plug-in such as the free Sonalksis FreeG (sonalksis.com). There are dedicated hardware and software processors that offer integrated M-S processing. To

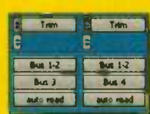
STEP-BY-STEP INSTRUCTIONS



Step 1: Return the stereo mix on Bus 1–2 to the two stereo aux inputs called Mid and Side.



Step 2: Use multimono Trim plug-ins to trim both auxes by 6 dB and to invert the right channel of the Side track. You'll need to unlink the left and right channels of the plug-in to do this.



Step 3: Assign the outputs of Mid and Side to Bus 3 and Bus 4 to accomplish the summing (L + R) and subtracting (L – R).

create an M-S matrix in Digidesign Pro Tools, see "Step-by-Step Instructions."

Because you're creating parallel signal paths, delay compensation may be required, depending on your DAW. In Pro Tools LE and M-Powered, there is no intelligent automatic delay compensation, but buffering at the CPU keeps things aligned perfectly under many circumstances. If you use a plug-in with its own look-ahead buffer on the mid or side track, however, you'll need to use Time Adjuster to delay the other track by the same number of samples. In DAWs that offer it, such as Pro Tools HD, simply enable automatic delay compensation.

Note that in the example project (see **Web Clip 1**), the mid and side tracks are trimmed down by 6 dB to prevent clipping when the left and right channels are summed to a mono bus. Feel free to hide the left and right aux tracks in the example when you're done, as you won't need to manipulate them further.

ONLINE BONUS MATERIAL

Squeeze from the Middle

Once you've derived mid and side, controlling width is only half the fun. Create master faders for the buses carrying the mid (bus 5) and side (bus 6), and use their inserts to process them separately. A favorite mastering trick is to use the mid channel to EQ the lead vocal; because it's usually panned to center, you won't affect the rest of the mix. You can also use your favorite multiband compressor to rein in the bass and kick on the mid channel without affecting the higher-frequency lead vocal in the mid channel and without altering any of the low sounds in the side channel.

If you want, you can go back and ride the fader on the lead vocal. To do that, split the mid channel to a pair of mono aux tracks, then use complementary highpass and lowpass filters on the two tracks to isolate the vocal on one of them.

Compress the side channel, which usually contains most of the ambience in a track, to correct a mix that's too dry without adding another layer of reverb. Because the bass and kick are usually panned to center, rolling off some of the low end of the side channel helps clear space for those two critical components.

Use a tube-emulation plug-in to warm up the mid channel without clouding the rest of the mix. Sonic dissection opens up all sorts of imaginative possibilities. For example, try the technique on stereo drum overheads, room mics, or any other stereo track or submix. Once you've mastered the mid-side matrix, anything is possible. em

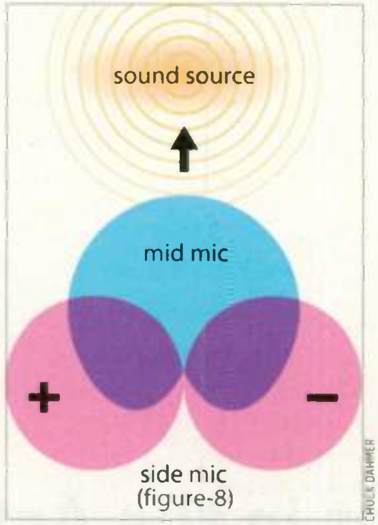
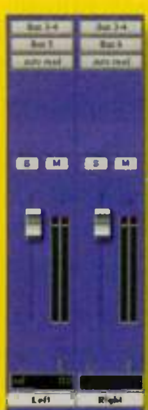


FIG. 1: M-S miking combines a cardioid mic with a coincident figure-8 mic aimed to the left. A simple matrix combines the two signals to derive discrete left and right channels.

Brian Smithers, author of Mixing in Pro Tools: Skill Pack *(Cengage Learning, 2006), teaches at Full Sail University, where he is chair of the Workstations Department.*



4

Step 4: Return Mid and Side (Bus 3–4) to the two stereo auxes called Left and Right.

5

Step 5: Use a multimono Trim plug-in to invert the right channel of the right aux input. Assign their outputs to Bus 5 and Bus 6 to accomplish the summing (M + S) and subtracting (M – S).

6

Step 6: Return Left and Right (Bus 5–6) to a stereo aux. Adjust the level of the Mid aux to vary the stereo width.

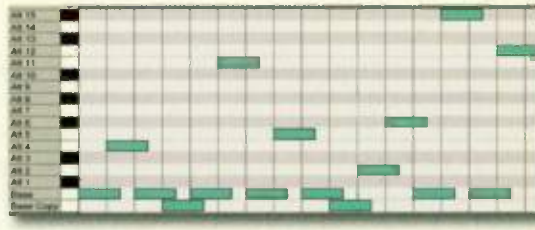


FIG. 1: This MIDI clip triggers captured slices at 1-bar intervals, but each note extends an extra two beats to accommodate a ping-pong-delay tail.

Caught in the Act

Multisample random processes and harvest the keepers. | By Len Sasso

Add a random element to note or automation sequencing, and you get varied results. If you simply loop the process, you'll get good takes interspersed with bad. Furthermore, the track will sound random—every take will be different (see **Web Clip 1**). One solution is to capture many takes, combine them in a multisampled instrument, and then create a composite track using the best ones. Here, I'll describe how to do that in your DAW, and then I'll discuss the potential pitfalls. If your computer is a Mac, you can automate the process and avoid some of the tedium by using Redmatica AutoSampler (redmatica.com).

ONLINE BONUS MATERIAL

In my examples, I've used a drum-sequencing script for Native Instruments Kontakt called Mobile Drums from Soniccouture Scriptorium (soniccouture.com). It is a 6-track, 16-step sequencer, with randomizers that shift the step positions, change the triggered drum sound and its level, and play drills (rapidly repeated notes). For each of the six tracks, you set how likely to occur each of these transformations is, and how radical each one should be. Any incoming note starts the sequence playing at the host DAW's tempo, and the sequence continues until the note is released. The incoming note's pitch doesn't affect the sequence. You can use the Mobile Drums script with any Kontakt 2 or 3 instrument, but it's intended primarily for percussion.

In Your DAW

Once you've set the tempo of your project and decided on the length of individual takes, create a looping MIDI clip to trigger the synth or sampler you're using for the source material (Kontakt, in my case). If the sound has a tail (as created by reverb, echo, or delay effects), the MIDI clip's loop should be long enough to capture the tail, but the trigger note it contains should be the length of a take. For instance, if you want to capture 1-bar drum patterns, the trigger note should be one bar long—but you might loop it every two bars. If you do not extend the loop in this way, the tails will be cut off when the MIDI clip loops (see **Web Clip 2**).

Repeat the trigger loop for the number of takes you want to capture, then render the track as audio. Slice the rendered audio clip into pieces that match the loop size (two bars in the above example), and use them to create a multisample map in your sampler. The length of the note you use to play a slice in your sampler determines how much of the slice plays, so the trigger notes should be long enough to include the tails. On the other hand, they should be spaced at intervals the size of the take length. That will cause consecutive trigger notes to overlap, which will not be a problem unless they trigger the same multisample zone. If you want to play the same zone twice in a row, copy it to a new multisample zone and trigger the two zones alternately (see **Fig. 1** and **Web Clip 3**).

As an alternative to sequencing the slices, consider using a step sequencer or triggering the slices manually in real time. In the latter case, you might want to quantize the trigger notes. And if your DAW doesn't support real-time quantizing, you may be able to rig up a transposing step sequencer or an arpeggiator to do the trick.

In AutoSampler

AutoSampler automates the process, including creating a multisampled EXS24 instrument. If your sampler doesn't import EXS24 instruments (most do), it's a simple matter to create your own multisample map from the WAV files that AutoSampler creates. The program does much more than this, including creating layered instruments with controller crossfades, and capturing multiple presets using MIDI Program Changes.

You start by setting Scan parameters to encompass a range of notes (pitch doesn't matter in this example) corresponding to the number of takes you want to capture. Then you set the take size in milliseconds (2,000 ms for 4 beats at 120 bpm). AutoSampler automatically takes care of capturing the tail by continuing to record until the audio drops below a threshold that you set on the Samples tab. Click on the Start button, and grab a cup of coffee.

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

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External SATA (eSATA) drives can provide better speed than either USB 2.0 or FireWire hard drives. The Seagate FreeAgent Xtreme (above) includes all three types of connectors on the rear panel (right).

eSATA Hard Drives

New storage solutions provide faster speeds than ever. | By Brian Smithers

Modern musicians are always looking for larger, faster, and more-convenient data storage. The current trend in mobile storage is *eSATA*, the external version of the SATA (Serial AT Attachment) standard. As its name suggests, this standard is related to the ATA/IDE specification, under which most internal drives have operated since the late 1980s. SATA drives are increasingly popular with PC users and have been standard on Mac desktops since the first G5s.

External Influence

Almost as soon as SATA drives first appeared, people started adapting them for external use, but the eSATA specification wasn't formalized until 2004. The official spec includes several optimizations for external use. A proper eSATA cable has an extra layer of shielding for greater protection from electromagnetic interference. In an effort to guard against electrostatic discharge upon insertion, the metal contacts are set deeper into the connector. To be sure that internal and external devices won't be mistaken for one another, the eSATA connector doesn't have the L-shaped insertion guide that characterizes the internal SATA connector.

The performance of eSATA outstrips its competition by a wide margin. USB 2.0 offers maximum data transfer at 480 Mbps (megabits per second), and FireWire 800 (IEEE-1394b) goes up to 800 Mbps. eSATA comes in two flavors: 1.5 Gbps and 3 Gbps. If you see a reference to SATA II, it probably means the 3 Gbps variety, but the name SATA II is incorrect.

If you divide 1,500 Mbps by 8 bits per byte, you would expect to get 187.5 MBps (megabytes per second). However, since SATA uses 8-bit-to-10-bit encoding (meaning that each set of 8 bits is mapped to 10 bits for technical reasons), SATA 1.5 Gbps provides "only" 150 MBps. If you compare that with USB 2.0's 60 MBps and FireWire 800's 100 MBps, eSATA's appeal to electronic musicians is immediately apparent.

Serial Killer

In the demanding world of external audio drives, USB has met with mixed success. Some people report great results using them with USB 2.0, but Digidesign specifically recommends against external audio drives for its Pro Tools systems. Although no other DAW manufacturer has issued a similar warning, that still represents a significant downside for USB. Though Digidesign has not yet officially qualified eSATA drives, the widespread consensus of Pro Tools users is that the drives are perfectly compatible.

FireWire drives are more universally accepted than USB, but there have been enough reports of fried controller chips when hot-swapping that several informed sources have been advising against the practice. Having had friends lose hardware this way, I always power down FireWire drives (if not the computer itself) before disconnecting them. To date, eSATA's hot-swap capabilities seem to be more robust.

Because few, if any, computers currently ship with eSATA ports, various devices are available to provide such connections. CardBus and ExpressCard solutions are available for notebooks, and PCI cards are available for desktops. SATA-to-eSATA extenders are available from a couple of sources (for example, from Sonnet [sonnettech.com]) that take advantage of two unused SATA connections on MacPro motherboards. But because they are merely adapters and not host bus interfaces, they are not fully eSATA compliant. Most significantly, they don't support hot-swapping.

When using a SATA controller, make sure your BIOS is running in AHCI (Advanced Host Controller Interface) mode instead of PATA (Parallel ATA) compatibility mode. AHCI is necessary to support hot-swapping, and it enables a feature called Native Command Queuing (NCQ). NCQ allows the drive to decide the best order in which to execute incoming requests, reducing inefficiencies in drives that support it.

Short Shrift

Two aspects of the eSATA specification come up short. First, the recommended maximum cable length is only 2 meters. Though this is fine for a single drive sitting on a desk, it complicates matters in a studio, where drives are commonly placed farther away to keep them quiet. Second, eSATA does not provide power to drives, making it slightly less convenient for remote recording.

The only constant in technology is change, and SATA-IO (sata-io.org), which oversees the SATA spec, has recently approved a 6 Gbps standard. With USB 3.0 promising 4.8 Gbps, and with 3.2 Gbps and 6.4 Gbps FireWire standards being developed, we're sure to revisit this subject periodically. em

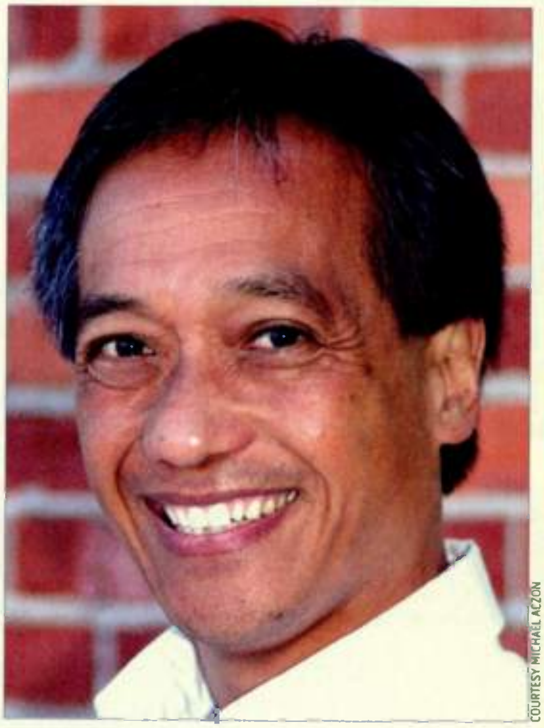
Brian Smithers is department chair of workstations at Full Sail University in Winter Park, Florida.

Q&A: Michael Aczon

The changing legal landscape for musicians.

It's always been critical for musicians to understand the legal issues they face, no matter which segment of the music business they're involved in. A legally informed musician is more likely to be a successful one. As technology has turned the music business on its head in recent years, the legal issues musicians must deal with have evolved. To get a sense for some of these new concerns, I spoke with Michael Aczon (www.aczon.org), who is the author of *The Musician's Legal Companion* (Cengage, 2008) and who has spent the past 25 years practicing entertainment law (see Fig. 1).

By Mike Levine



COURTESY MICHAEL ACZON

FIG. 1: Entertainment attorney Michael Aczon says technological developments have changed the legal issues musicians face.

What, in recent years, would you say is the most important issue that has changed for musicians legally?

Just as they have to do more themselves technologically and artistically, musicians now have to deal more with legal issues *before* they interact with other people. They have to cover their legal rear ends much the same way as they have to do their own artist development.

Why is that?

During the past couple of decades on the business, technological and social sides of the music industry, you've seen consolidation of record companies and of media in general—and radio and television specifically—and a business that has focused more on distribution and promotion than on artist development.

What do musicians have to do these days legally that they didn't have to do previously?

The first is to secure their own trademarks. We've turned into a world of getting your name out there and establishing yourself as a trademark, or transferring the rights to establishing your trademark to somebody else.

You mean trademarking the band name?

Yes. People are putting their work on social networks, selling their stuff on CD Baby, [and so on]. And the next thing you know, you find out that you're infringing on somebody else's trademark because they were using the same band name as you for the last ten years. A lot of people don't do basic research to find out if there's some other band with their name.

Let's say there's a hypothetical band called the Dodos. They make a MySpace page and put their music up there, and then they find out there's another band called the Dodos that has already trademarked the name. Is that trademark infringement?

Yes, it is.

Just the act of putting up a page with that name would be considered infringement?

Yes. The presumption is that if you're going to use a trade name, you've researched that name before using it. Now, let's go to the legal and the practical. The legal side is that if somebody has gone through the steps to register that name—either federally or even internationally—and they continue to use it, then clearly they're not going to give that right up. On the practical side, here's some independent band that's spent a couple of thousand of their own dollars recording, and that's put their music on a Web site, or started selling it on, for example, CD Baby. The legal and practical side of this scenario is that the person who already owns that trademark can make the second band shut it all down, take the name off of their products, and change their name. To a baby band, that couple of thousand dollars and the time, effort, and energy to rebrand their band means something.

Can you register your own trademark, or is that something that has to be done through an attorney?

[You can do it] yourself, and the USPTO—the United States Patent and Trademark Office [uspto.gov]—has relatively easy-to-follow instructions. I like to say if you can install Pro Tools software on your computer, you can probably read through this well enough to do it on your own. Keep in mind, though, that just like installing Pro Tools, if you're doing it for the first time, it may be a trial-and-error process. So it's important to weigh if the possibility of doing it incorrectly makes you want to have an expert do it for you.

It requires doing a search first?

Exactly. And there are search engines. I'm not going to recommend any, but you can go online and find search companies.

Why has the trademark issue changed? Didn't you always have to trademark your name? Is it because of the online aspect of it?

Right. I think that because of the ability to go out there and almost automatically cross state lines with your work [due to the Internet], a lot of acts are trademarking—I wouldn't say prematurely, but they're pulling the trigger a little bit quicker than they used to. Certainly more quickly than 20 years ago, when you really weren't thinking about "Oh, do I get a trademark because I'm going to go outside my city and try to sell CDs?" I'm giving away my age here, but when I started out, if I was going from, say, California to Utah to sell my recordings, I had to think, "Now I'm crossing state lines. Maybe I should get a few things in order." These days, I could be recording something this morning and it could be available for sale internationally by this afternoon.

What other legal issues have changed for musicians?

Copyrights. And again, this is a shift with technology. Before, musicians would think solely about two copyrights: their compositions and their recordings. Now, if you're putting up a Web site, you have to think about copyright ownership of photographs, text, and videos; as well as all the audio.

So let's say that a band puts up a Web site and they post photos that a photographer friend gave them. Would the band have the right to use those photos?

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What's your advice for musicians in terms of dealing with these new issues? It's pretty expensive to put an attorney on retainer.

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Like music conferences?

Exactly. We're seeing more and more of those. And I think it's also a good idea to pay an attorney for a basic consultation. It lets them know you mean business.

You're saying to go to the attorney, pay that person for a half hour or an hour of time, and see if you have any legal issues you need to deal with?

Exactly.

Do you think there will be more and more legal issues confronting musicians as technology progresses?

We've been moving so fast because of technology, particularly in the last ten years, that the legal issues have had to follow it. I think those are shaking out, and my personal prediction is that it's going to slow down a little bit now that some legal and business models are being accepted by the creative, business, and consumer sides of the industry. em

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Mike Levine is EM's executive editor and senior media producer. He hosts the twice-monthly Podcast "EM Cast" (emusician.com/podcasts).

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the steps to register that name—either federally or even internationally—and they continue to use it, then clearly they're not going to give that right up. On the practical side, here's some independent band that's spent a couple of thousand of their own dollars recording, and that's put their music on a Web site, or started selling it on, for example, CD Baby. The legal and practical side of this scenario is that the person who already owns that trademark can make the second band shut it all down, take the name off of their products, and change their name. To a baby band, that couple of thousand dollars and the time, effort, and energy to rebrand their band means something.

Can you register your own trademark, or is that something that has to be done through an attorney?
[You can do it] yourself, and the USPTO—the United States Patent and Trademark Office [uspto.gov]—has relatively easy-to-follow instructions. I like to say if you can install Pro Tools software on your computer, you can probably read through this well enough to do it on your own. Keep in mind, though, that just like installing Pro Tools, if you're doing it for the first time, it may be a trial-and-error process. So it's important to weigh if the possibility of doing it incorrectly makes you want to have an expert do it for you.

It requires doing a search first?
Exactly. And there are search engines. I'm not going to recommend any, but you can go online and find search companies.

Why has the trademark issue changed? Didn't you always have to trademark your name? Is it because of the online aspect of it?
Right. I think that because of the ability to go out there and almost automatically cross state lines with your work [due to the Internet], a lot of acts are trademarking—I wouldn't say prematurely, but they're pulling the trigger a little bit quicker than they used to. Certainly more quickly than 20 years ago, when you really weren't thinking about "Oh, do I get a trademark because I'm going to go outside my city and try to sell CDs?" I'm giving away my age here, but when I started out, if I was going from, say, California to Utah to sell my recordings, I had to think, "Now I'm crossing state lines. Maybe I should get a few things in order." These days, I could be recording something this morning and it could be available for sale internationally by this afternoon.

What other legal issues have changed for musicians?
Copyrights. And again, this is a shift with technology. Before, musicians would think solely about two copyrights: their compositions and their recordings. Now, if you're putting up a Web site, you have to think about copyright ownership of photographs, text, and videos; as well as all the audio.

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BRISBANE::BYRON BAY::AUCKLAND::MELBOURNE::TOKYO

Moog Music

Minimoog Voyager Old School

Reeducate yourself with this performance mono synth.

By Jason Scott Alexander

>> PRODUCT SUMMARY

monophonic analog synthesizer $2,395

PROS: Timeless, earthy sound. Three analog VCOs. Dual multimode filters. Enhanced modulation. CV inputs and outputs. Durable, handmade construction. Inspiring to touch and play. Major cool factor.

CONS: No presets. No MIDI.

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

Moog Music
moogmusic.com

>> In our reviews, prices are MAP or street unless otherwise noted.

ONLINE BONUS MATERIAL

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

It's been six years since Moog Music revitalized the most successful analog synthesizer in history. The Minimoog Voyager updated the classic Minimoog Model D by adding a software-based operating system, MIDI control, preset memory, an LCD, a three-dimensional touch pad controller, and many other improvements (see the October 2003 review at emusician.com).

Whereas the Voyager provided distinct advantages of digital technology, the Old School (OS) takes them all away. The OS's aesthetic, therefore, more closely resembles the Model D's, but the underlying analog sound engine remains strictly Voyager. The OS furnishes a patching and modulation schema that prog rockers could only have wished for in their heyday.

Playing Inside the Box

Weighing a hefty 40 pounds, the OS features a solid hardwood cabinet with a beautiful furniture-grade finish and a 44-note (F to C) Velocity- and Aftertouch-sensitive keyboard (see Fig. 1). Glide and Release switches are located above the pitch and modulation wheels—just as on the original—and the multiposition hinged design of the synthesizer module allows it to be adjusted to a number of angles.

The pitch-bend range is fixed at ±5 semitones, but Moog Music's Web site outlines a procedure to reconfigure an internal jumper to widen or narrow this range. Although the fixed range is a price you must pay for all-analog construction, at least Moog offers a means to change the default. The keyboard priority is last note and the trigger mode is legato, but you can configure the OS for multitrigger mode (in which each note played on the keyboard retriggers the gate) by holding down the keyboard's top two keys as you power up the synth. It will revert to single-trigger mode the next time you power up.

The rear panel is chock-full of connections, with a pair of audio outputs, an analog audio input, an effects insert, and an assortment of control voltage (CV) and gate inputs that allow the OS to act as a semimodular synth (see Fig. 2). But the big news is that it has keyboard CV and gate outputs, which were not present on the original Voyager—very handy. You can access many

FIG. 1: Although the Old School strips away the previous Voyager's patch memory and digital connectivity, it delivers hands-on programmability and control far beyond the original Minimoog's.

more control outputs using the optional VX-351 CV Expander ($295), which connects to the synthesizer's DB-25 multipin accessory port.

Class Is In

The Minimoog has always been a textbook lesson in subtractive synthesis. Its front-panel layout couldn't be more straightforward, and for the most part, things are just where you'd expect them on the OS, too.

Beginning at far left are the familiar Fine Tune and Glide Rate controls. Adjacent are the dedicated LFO and dual Modulation Bus modules, none of which were available on the original Minimoog. The LFO produces triangle and square waves as well as stepped and smoothed sample-and-hold (S&H) patterns. Though typically you'd use the LFO's square wave as the S&H trigger and the noise generator as the sample source, you can override them by patching external signals into the S&H inputs, making all sorts of cool user patterns possible. The Mod 1 and Mod 2 buses are identical, each capable of selecting from six sources, six destinations (including waveshape), and six controllers with adjustable amounts (I explore these modules in the online bonus material at emusician.com).

Three analog VCOs (voltage-controlled oscillators) follow, each with a range marked in standard organ-stop measurements of 32 feet (lowest setting) to 1 foot—a full octave higher than the original Minimoog. Waveforms are continuously variable from triangle through sawtooth to square and pulse. You adjust Oscillator 1 using the global Fine Tune control, which is necessary because the OS doesn't have an autotune function. Oscillators 2 and 3 have separate Frequency controls, each capable of fine-tuning seven semitones up or down relative to Oscillator 1. Two switches allow for syncing Oscillators 1 and 2 and for the linear frequency modulation of Oscillator 1 by Oscillator 3.

Within the Mixer section are five rotary knobs and five rocker switches for blending and toggling the three oscillators, external audio input, and internal noise generator on or off. Whereas the original Minimoog gave you a choice between pink and white noise, here you're provided with a single hybrid mix of the two noise colors.

[The underlying analog sound engine remains strictly Voyager.]

The Old School's Filters module comprises two classic Moog 4-pole (24 dB-per-octave) self-oscillating multimode filters instead of the original Minimoog's single lowpass. By flicking a toggle switch, you can configure them as parallel lowpass filters routed to the left and right outputs, respectively, or as a serial highpass-lowpass combination resulting in bandpass across both outputs equally. A single set of Cutoff, Resonance, and keyboard-tracking knobs acts on both filters simultaneously, while the Spacing control shifts or spreads the cutoff frequencies, depending on which filter mode the synth is in.

»» Use the multipin connector to access control outputs with the optional VX-351 CV Expander.

»» Keyboard gate and CV outputs allow you to interface with external instruments.

»» Process external sounds through a pair of 4-pole (24 dB-per-octave) filters.

»» **FIG. 2:** The Voyager OS's jack panel offers impressive analog connectivity for control-voltage and audio signals.

As on the analog-digital hybrid Voyager, the OS has reoriented some classic controls while making room for new ones. The Oscillators bank, for example, is now aligned into columns rather than rows as they were on the classic Minimoog. Similarly, the Filters section is now vertical, and the dual ADSR envelopes are broken away into their own module. Being a Minimoog veteran, it took me a minute to get used to the changes, but I appreciate that the layout more intuitively follows modern concepts of organizing synthesis functions.

Kickin' It Old School

I love that waveshape is a voltage-controllable parameter, because it allows you to dial in combinations of abstract tones and perform pseudowavetable synthesis using modulators (see **Web Clip 1**). Thanks to the incredible stability of the oscillators, the linear FM control lets you conjure up all sorts of clangorous and metallic-sounding goodies containing unmistakably warm, analog undertones. Hard sync is something every Minimoog owner has longed for, and depending on how you apply it here, the sound can be aggressive with extreme overdrive or warm, richly harmonic, and quite vocal (see **Web Clip 2**). Obviously, having Velocity and Aftertouch to integrate with up-to-date modulation offerings is a big thrill for Minimoog aficionados like myself, as are the exquisite-sounding multimode filters.

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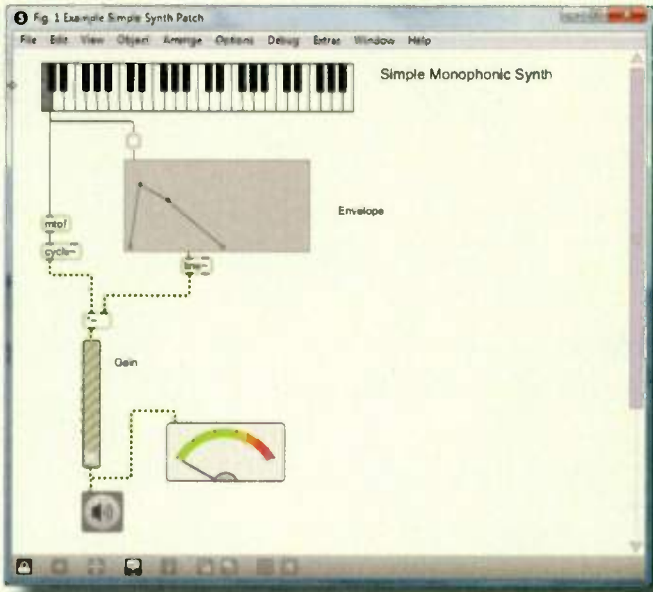

FIG. 1: This is an example of a simple monophonic synth patch. In Max, Objects are wired together like this to produce any multimedia configuration you can imagine.

Cycling '74

Max 5 (Mac/Win)

A new face for an old classic.

By Peter Hamlin

PRODUCT SUMMARY

sound-programming environment

Max	**$250 (MSRP)**
Max/MSP	**$495 (MSRP)**
Max/MSP/Jitter	**$699 (MSRP)**

(educational pricing also available)

PROS: Much-improved user interface and documentation. Huge collection of Objects. Unlimited potential for creating multimedia interactive tools.

CONS: A program of this nature takes some effort to master.

FEATURES	5
EASE OF USE	3
AUDIO QUALITY	5
VALUE	4

Cycling 74
cycling74.com

ONLINE BONUS MATERIAL

Cycling '74 Max has been around in some form for 20 years, and over that time it has become a particularly important and widely used tool for electronic musicians, especially those who are more experimentally inclined. Max was first commercially available in 1990. MSP, the audio-processing component, was added in the late 1990s, and a suite of video Objects called Jitter was added in 2003. (The previous major upgrade, Max 4.0/MSP 2.0, was reviewed in the April 2002 issue, available at emusician.com.)

In working with Max, you create programs (called patches) in a Patcher window by choosing from a huge library of Max, MSP, and Jitter Objects and then hooking them together with patch cords. Typically, you hook the outlet of one Object to the inlet of another. The patches you produce might be software synthesizers, audio or MIDI processors, algorithmic-music machines, interactive controls, animations that respond to music, music that responds to live video input, or any other multimedia project you can dream up. Your patches could be simple utilities that you create in a short time or complex programs that take months to design.

Fig. 1 shows a simple example of a patch: a monophonic synthesizer that you play with an onscreen keyboard.

Max 5 makes very few changes in the Objects themselves—for the most part, you still have the same massive collection of more than 600 (see the **online bonus material** "Object Oriented" at emusician.com for a list of some new and enhanced Objects). The improvements in Max 5 come in the documentation and the user interface.

What's Up, Docs?

Previous versions of Max offered contextual help: you right-clicked on an Object to get a help document that summarized the features of that Object. One of the coolest aspects of the contextual help system was that the help page was a working version of an example patch, not just a graphic diagram of an example. So while reading the information, you could also experiment with the working example, and even copy all or part of it to your own patch for reuse.

In previous versions, there was also a more detailed reference manual, but you had

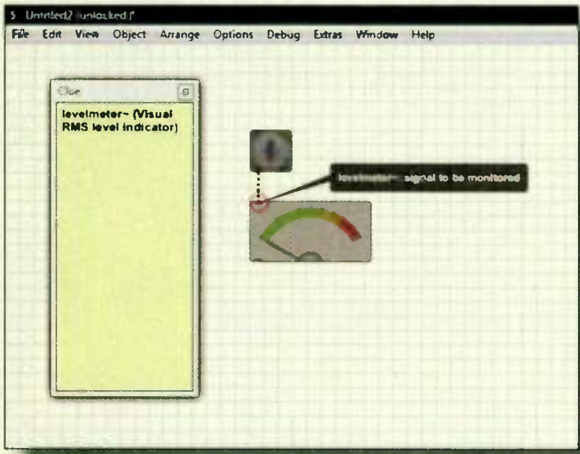

FIG. 2: A balloon with an arrow gives a brief explanation when the mouse moves over the inlet or outlet of an Object (in this case, a Viewmeter inlet). A Clue window gives the name of the Object and a short description.

to open that document separately to search for the information you wanted. In Max 5, there is now a link from the shorter contextual help page to the larger and more complete reference manual page for a particular Object. In fact, in Max 5, all of the documents are linked together, making it much easier to navigate through this extensive collection of online reference materials. A comprehensive Search feature also lets you look throughout the documents for any keywords.

When you place the mouse over an Object's inlet or outlet, a large balloon with a big arrow appears, describing what that inlet or outlet is used for. (The previous version showed a small and much-harder-to-read notation in the lower left of the screen.) There is also a new floating Clue window that gives a brief but helpful description of any Object the mouse moves over (see Fig. 2). With these improvements in the documentation, now you can spend most of your time building patches and less time figuring out how to do that.

Interface Enhancements

Many features of the user interface have been improved. I'll cover the ones that seem the most significant.

Presentation mode. In previous versions of Max, it was possible to create a front panel for your patches that showed only the Objects that were essential to operation (for instance, the knobs, displays, and other controls that the user changes and views in operating the patch) while hiding the underlying circuitry from view. But it was a little unwieldy to do this. Max 5 has improved greatly in this area with the addition of a Presentation mode that lets you effortlessly create an attractive and uncluttered front panel for your patch.

Patcher palette. Most Objects in Max perform simple functions (such as adding two numbers together), and these Objects are created simply by typing the name of the Object into a blank Object box. Max 5 has added a handy Auto-completion feature to simplify the process as you type. There are also User Interface (UI) Objects such as buttons, LEDs, level meters, number displays, and many others. In previous versions, the UI Objects were in a toolbar at the top of the screen. Now they appear in a nicely organized floating palette that is created any time you double-click in the Patcher window (see Fig. 3). The new Patcher palette is organized by Object type, the icons are larger and easier to identify, and you can add your own customized Objects (called Prototypes). Work is also speedier in Max 5 because there are new shortcut keys to quickly create the most common Objects.

Performance. Max handles polyphony with an Object called poly~. In Max 5, poly~ can take advantage of multiple processors to allow faster processing time.

There are many other changes, including Multiple Undo; a Zoom feature; the ability to maintain multiple views of your patch; a Grid and a Snap To Grid option that makes it easier to create tidy and clear patches; and a more uniform and powerful Inspector window that improves consistency when customizing the appearance and functionality of Objects. A new File Browser makes it much easier to keep track of the many files you'll be producing with Max.

On Time

Other new additions worth mentioning relate to timing increments and tempo. In previous versions of Max, time was referenced in milliseconds. But in Max 5, you can also work with traditional musical time values using ticks (1/480 of a quarter note) or note values (such as 1n for a whole note, 4n for a quarter note, 16n for a 16th note, 4nt for a quarter triplet, and so on). You can also use a bar:beat:tick notation (bar 1, beat 1, tick 240 would be notated 1:1:240). A new Transport Object and a Global Transport window let you start, stop, and pause the music, change tempo, and otherwise control the flow of sound in your patch.

The changes in Max 5 don't alter the program in a substantive way, but they do make it significantly easier to learn and use the program, organize and customize patches and Objects, and share or collaborate with others.

FIG. 3: A new Patcher palette organizes UI Objects in a much clearer and more convenient way.

There are many large changes and countless small ones, all of which are welcome. Your patches will be clearer and better-looking, and you'll have a much easier and funner time building applications that do what you want them to do (see the online bonus material "A Max Example" for a patch I created in version 5). I strongly recommend the upgrade (starting at $129) for existing Max users. And for anyone considering a modular toolkit of this sort, be sure to download the Max 5 demo and give it a try.

Peter Hamlin teaches electronic music, theory, and composition at Middlebury College and plays in the live electronic improv band Data Stream.

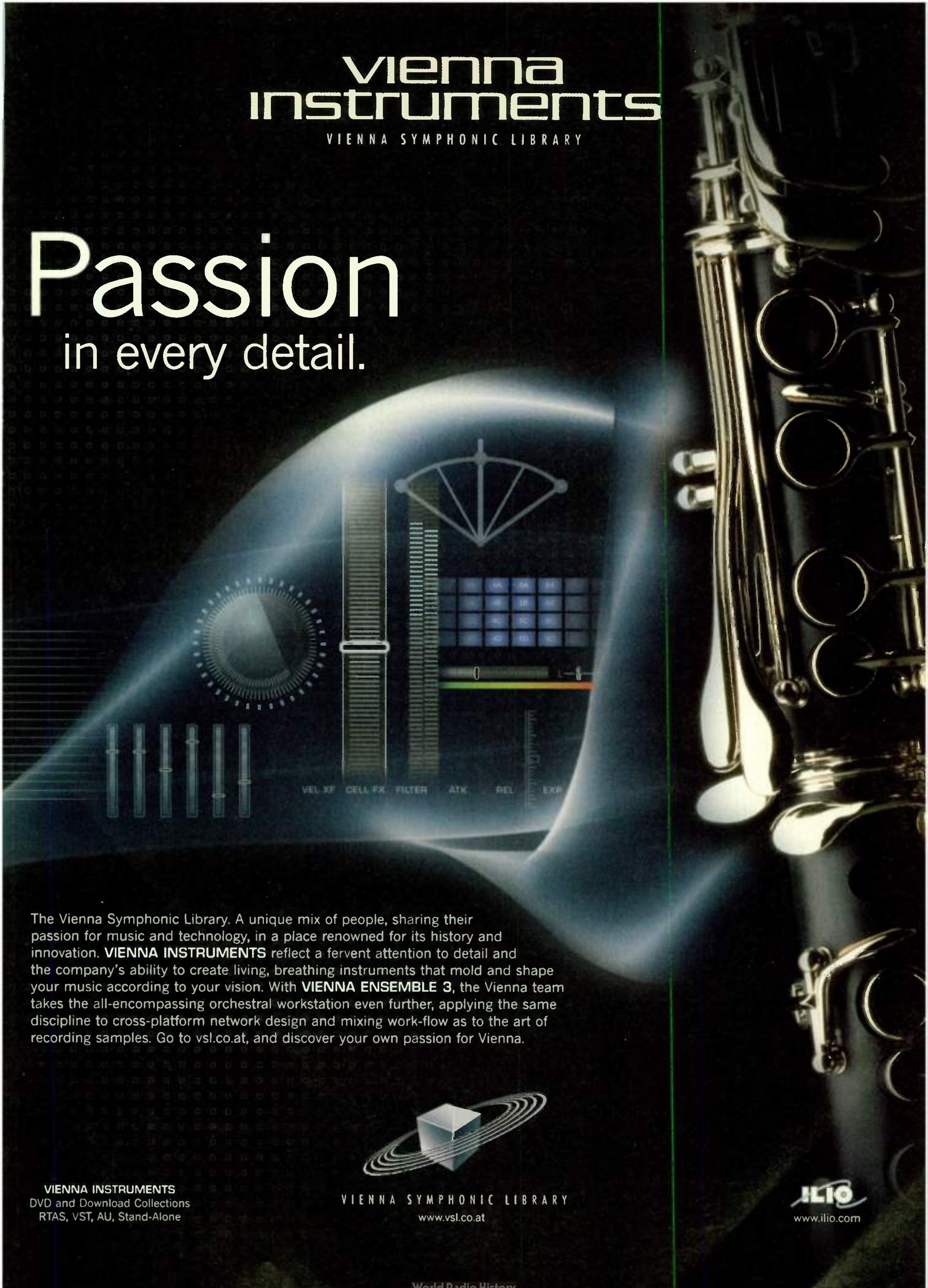
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FIG. 1: The ARC plug-in EQ's your playback system to give it a flat response based on the curve generated by the measurement application and the optional Target Curve.

IK Multimedia

ARC System (Mac/Win)

A software-based system to correct your studio's acoustics.

By Mike Levine

PRODUCT SUMMARY

room correction system **$599.99**

PROS: Can substantially improve playback in an untreated room. Good documentation. Target Curves allow further sound tailoring. Measurement process easy to complete. Allows for wide sweet spot.

CONS: Must turn off ARC processing before bouncing on some systems. Playing back bounced mix through ARC processing requires extra steps. Must switch EQ curves when switching monitors.

	Rating
FEATURES	3
EASE OF USE	3
AUDIO QUALITY	3
VALUE	4

IK Multimedia
ikmultimedia.com

ONLINE BONUS MATERIAL

Wouldn't it be great to trust the acoustics in your studio? To be able to do a mix where you didn't have to take a copy of your song, burn it to a CD, and then listen to it in your car, on your boombox, and even on your living room stereo to see how it translates? The reason why so many personal-studio owners often must do so much referencing during the mix process—even if they have excellent monitors—is that their studios have not been acoustically treated and tuned. As a result, sonic gremlins such as first-order reflections and standing waves wreak havoc with the sound and make it impossible to mix with confidence.

A big impediment to studios' getting treated is an economic one: custom acoustic treatments can be expensive. And many recordists would rather allocate their equipment budget to something sexier, like a new plug-in or microphone. But what if there were a relatively inexpensive way to use computer-based DSP to compensate for those acoustic anomalies? IK Multimedia's ARC System offers just such a solution, using Audyssey's MultEQ technology (see "Tech Page: Got Modes?" in the September issue, available at emusician.com), which was originally invented for the high-end home-audio and home-theater markets. The system is designed to correct both frequency- and time-based (phase) problems.

Process It

Can EQ alone cure your acoustic problems? I asked acoustic consultant Bob Hodas about it, and he told me that EQ is typically the icing on the cake of a treated room, used for final finishing-touch tweaks after all the absorbers, diffusers, and bass traps have been installed.

Other products on the market besides ARC use DSP as the primary tool for room tuning, including JBL's LSR-series speakers and KRK's ERGO system. One way in which these three products differ is that they all put their processing in different parts of the signal chain. For the LSR system, it's in the monitor cabinets themselves. For ERGO, it's in a dedicated box that's placed between the audio interface and the monitors. ARC does its processing as a VST, AU, or RTAS plug-in within your DAW (see Fig. 1).

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FIG. 2: The room measurement application features step-by-step onscreen instructions that guide you painlessly through the process of measuring your room's acoustics.

Checking—One, Two

ARC is composed of a standalone measurement application (see Fig. 2), a plug-in for implementing the EQ curves, and a measurement microphone. This omni-pattern condenser (see Fig. 3) is specifically tailored for ARC, and the system won't work correctly if you substitute another measurement mic. The mic comes in a padded black plastic carrying case, and a windscreen and custom clip are included. An IK spokesperson told me that it's of good-enough quality to also use as a recording mic. I tested it out on acoustic guitar, resonator guitar, and shaker, and the results were solid if not inspiring. (To hear it in action as a recording mic, see Web Clip 1.)

After installing the ARC software, the next step is to take measurements in your studio. You start by putting the measurement mic on a stand, pointing it straight up at the ceiling, and placing it at ear height at your listening position, centered between the monitors. You then launch the measurement software, and it prompts you to press the Test button to make sure you're getting enough level through your monitors (it emits a series of test tones). Once your levels are set, you're ready to start measuring.

The software guides you through the testing process and for each measurement emits a series of ten tones each for the left and right speaker. (Currently the ARC System doesn't support surround.) The manual recommends taking at least 13 measurements, preferably 14 to 16 for every set of monitors in your studio.

After each measurement, you move the mic to a different spot in and around your listening position. Suggested measurement positions are indicated graphically in the manual. (Kudos to IK for including a printed manual, not just a PDF.) According to IK, the Audyssey MultEQ technology is capable of correcting your room for multiple listening positions (for example, a main position with a client couch behind it).

If you have more than one pair of monitors, you need to take separate measurements for each type. That's because they have different frequency responses and interact differently with the room.

I began the measurement process for my studio by setting ARC up for my primary monitors with a single listening position. For each measurement, I followed the well-written manual's instructions and moved the mic to various symmetrical positions within an oval shape surrounding where I sit. Although the manual shows you a diagram of the various spots, it doesn't give you precise distances to move the mic. I tried to be ultra-accurate by placing a piece of tape on the ground at position 1, which is in the center, and using a tape measure to make sure the positions were symmetrical. I later found out from IK that you don't have to be that precise. The company says you can eyeball your mic positioning and the software will still take accurate measurements.

The measurement process goes fairly quickly, and once you've completed it, the software calculates an EQ curve and prompts you to name the preset. It then lets you select a speaker icon to graphically represent your speaker type when you call up a preset. (The graphic reminder is helpful if you have multiple monitors in your studio, because you have to change ARC presets each time you switch speakers.)

Open the DAW

To put your measurements to work, you open the ARC System plug-in within your DAW and insert it either on the master bus or, preferably, on a dedicated monitor bus, assuming you have a multioutput interface and your DAW allows it. I'll explain why in a moment.

The plug-in opens in a large window featuring a level meter, a trim control, a correction on/off switch, and several pull-down menus. The Measurement menu lets you choose the EQ preset that resulted from your measurement process. The Target Curve menu provides several additional presets to further tweak the room response if desired. Besides the default Flat setting, you can choose high-frequency rolloff, midrange compensation, and a curve that offers both.

The plug-in also displays a graph that shows helpful comparative data from the measurements: the original EQ curve of your monitors, the new curve with the ARC processing applied, and the frequency response after the Target Curve is enabled.

Once you've found the best setting for your room, the idea is to do your mixing while monitoring through the ARC plug-in with the correction on. When you're ready to bounce your mix down, you turn the correction off (assuming you have the ARC plug-in on the master bus; if you have it on a monitor bus, you can leave it on because it won't affect your mix).

Although turning off ARC before bouncing seems counterintuitive, it actually makes sense. By mixing while listening through ARC's processing, you're setting levels, EQ'ing, and adding effects with the corrected acoustics as your guide. You're making mixing decisions with ARC's processing compensating for the acoustic inaccuracies of your space. In a sense, ARC's EQ curves are fooling you into mixing as if you were in a "flat" room. That's why, if you're using ARC on your master bus, you must remember to bypass the plug-in before bouncing. Otherwise, you'd be redundantly

[The software guides you through the testing process.]

adding the ARC processing, and you'd totally skew your mix.

One problem I ran into is that it's easy to forget to turn the ARC processing off when it's time to bounce. That's because it's basically a set-and-forget system. You don't need to interact with it except at the beginning of a session when you're opening the plug-in and getting levels set, unless you're switching between monitor pairs. When I first began mixing with ARC, I found that I often left it on by mistake and had to rerun my mix. My life was made easier after I inserted the ARC plug-in on an aux bus that fed a separate output pair—other than the main L and R—for monitoring.

Another awkward aspect of the ARC System's architecture—specifically the fact that it does its processing from within your DAW—is that once you've bounced your mix and you want to listen back to it in your studio, you need to import your mixed track back into your DAW (or into a 2-track editor that lets you monitor through plug-ins) with the ARC plug-in turned on. This is necessary in order to hear the music as you mixed it; that is, with ARC compensating for your room's acoustics. (According to IK, a solution to this problem is in development.)

Into the ARC

Those issues aside, the most important question about the ARC System is: how well does it work at flattening the response of your studio? To help answer this, I tested the system both in my studio, which has acoustic treatment (both absorber panels and bass traps), and in another room in my house that's completely untreated.

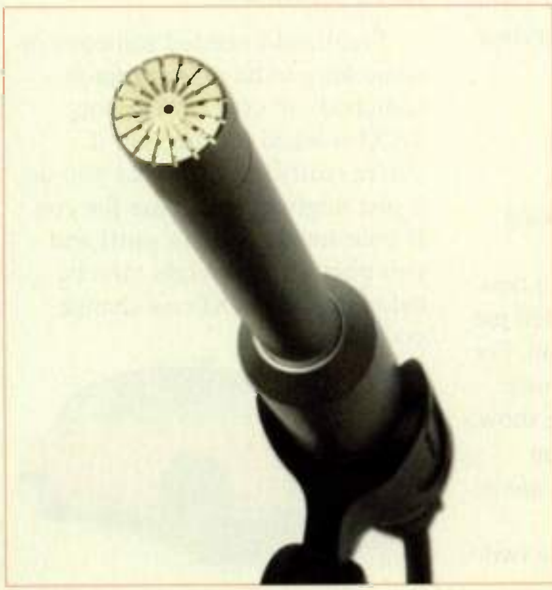

FIG. 3: Included is this omni measurement microphone, which has a frequency response that's custom-tailored for the ARC System.

Not surprisingly, the differences in my treated studio were mostly subtle ones. There was a little less tubbiness in the monitors, and slightly more present highs when listening with the ARC processing on. The stereo image seemed to shrink a tiny bit when the processing was engaged.

In the untreated room, the sonic differences were much more dramatic. That room is of medium size and has a rectangular shape, and the speaker system in it is an inexpensive consumer 2.1 setup.

In that room, the comparative graph in the ARC plug-in showed that the room had a significant bump in the low end and lower midrange. With the ARC processing switched on,

that bump went away. To compare the difference, I listened to mixes of songs that I was very familiar with. Sure enough, with it on, everything sounded clearer, less muddy, and more open. According to IK, one of ARC's strong points is its ability to make a bad-sounding room usable, and this was born out in my testing.

Mix and Match

I did quite a bit of mixing through the ARC System in both the treated and untreated rooms, and I was a little surprised to find that I didn't hear a major change in how those mixes translated to other speaker systems. Oddly, in my treated space, ARC's reduction of bass, which aimed to flatten the response, actually helped cause me to put too much bass into several mixes. Even in the untreated room, the differences between mixes done with and without ARC were less dramatic than I had expected.

Although I was surprised that ARC wasn't immediately improving my mixes, my contact at IK wasn't. He said that my ears were accustomed to my room's normal acoustics, and that it would take a while for me to get used to ARC's corrected sound.

Magic Bullet?

Overall, my opinion of the ARC System is mixed. I was impressed with its ability to apparently clean up the sound in a room, but I was disappointed when that didn't translate into substantial mix improvement. To be fair, there are other factors that affect how a mix turns out (mixing skill, ear fatigue, lack of objectivity, and so on), which certainly could have come into play. Still, I mixed close to ten full-length songs with ARC, over a number of weeks, and never saw significant changes in how my mixes translated.

A key question you have to ask yourself is: would you rather put the money you'd spend on the ARC System toward traditional acoustic treatment? If you're on a relatively tight budget and have a really bad-sounding room, ARC might be a cost-effective way to quickly improve the sound in your studio. But if you can afford physical acoustic treatment, I still think that's the best way to go.

Mike Levine is EM's executive editor and senior media producer and the host of the twice-monthly Podcast "EM Cast" (emusician.com/podcasts).

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JZ MICROPHONES

Black Hole

By Eli Crews

Latvian designer Juris Zarins is responsible for some of the most unique-looking microphones on the market, and the Black Hole large-diaphragm, multipattern condenser mic ($2,395 [MSRP]) is the flagship of his new brand, JZ Microphones. The most striking feature of the Black Hole is its design: it's shaped like a rectangular doughnut. The top of the mic has a compact black cage with two 27 mm capsules, while the Class A discrete amplifier electronics somehow fit into the thin sides of the body. In the hole is a switch for choosing the polar pattern (cardioid, figure-8, or omnidirectional), as well as two nubs, which are used to connect the mic to its innovative stand mount. A thumbscrew (or, alternatively, a coin-friendly screw) at the base of the mount allows you to easily position the mic.

The Black Hole is a large-diaphragm condenser mic offering three polar patterns.

Two other versions of the Black Hole are also available. The Black Hole SE ($1,895 [MSRP]) is cardioid only, and the Black Hole PE ($1,995 [MSRP]) is cardioid only but with an added pad switch (–5 and –10 dB). Although the pad option was added to meet consumers' requests, I never had a need for one—despite having the mic in front of loud brass instruments and screaming vocalists.

The Black Hole's accessories are heavy on design and light on practicality. For example, the little rubber feet isolating the nubs from the mount arms pop off easily (JZ says it's working on the problem), and the mount won't fit in the included wooden box unless you disassemble the thumbscrew. In addition, the magnets holding the box closed are so strong that a considerable amount of force is needed to open it, which could easily send the mic flying when the box pops open (the company says it's updating the box design).

JZ also sells a unique shockmount and pop-filter combo ($360 [MSRP]) that fits all three mics. I like it quite a bit, but it's surprisingly pricey.

A STAR IS BORN

I received two Black Hole mics shortly before I started a film-scoring project, and I was able to use them on a number of different instruments over several months: drums and percussion; various reeds and brass instruments; male and female voices; electric, steel-string, and nylon-string guitar; and upright and electric bass. The mic's standout application was as a drum overhead. The two mics captured the detail and clarity of the cymbals and drums in an extremely open, natural fashion. On at least one session, I was perfectly happy using the Black Holes almost exclusively for the final drum sound (augmented only by an additional mic on the bass drum).

Vocals and reed instruments were crisp and bright through the Black Hole, sounding clearer (though a bit thinner) than through my '60s Neumann U 87. On both nylon- and steel-string acoustic guitar, the Black Hole had much more depth and presence than the Blue Bluebird I had been using, although it should be noted that that mic retails for about one-third the price of the Black Hole. On brass instruments, the Black Hole had a brilliant clarity that worked well for accentuating the upper harmonics on tuba and trombone, although it was a little too bright for trumpet.

EM HOT PICK

When experimenting with the three polar patterns, I found that the cardioid position had a much brighter, more forward sound, whereas the omni pattern sounded darker but more natural, with the added roominess one would expect. Figure-8 was tonally in between, and both sides of the diaphragm sounded identical to me. The cardioid pattern seemed a little tighter than on most of my large-diaphragm condensers, allowing slightly less of the room characteristics to affect the sound of the miked instrument.

A SINGULARITY

I absolutely love the sound of these mics. They have a very open, natural quality, which is achieved by a boost in the high mids and the top end, with a dip in the boxy low midrange. Somehow they manage to do that without getting a brittle or harsh quality, although in a few rare cases I found the mic's sound to be a little thin.

There are many good large-diaphragm condenser mics on the market for considerably less money, but if you have golden ears and the pocketbook to match, give the Black Hole a listen. You may hear what you've been missing.

Value (1 through 5): 4
JZ Microphones
jzmic.com

SONIC REALITY

Ocean Way Drums Gold (Mac/Win)

By Mike Levine

If I told you that there was a drum sample library recorded in a top-quality studio (Ocean Way Recording) by two highly respected producer-engineers (Steven Miller and Grammy Award winner Allen Sides) using a tried-and-true software platform (Native Instruments Kontakt Player 2) and the expertise of a major sound developer (Sonic Reality), you would expect a superior product. In the case of Ocean Way

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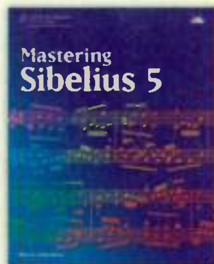

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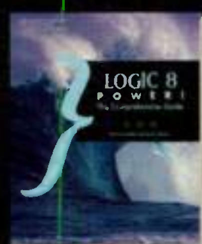

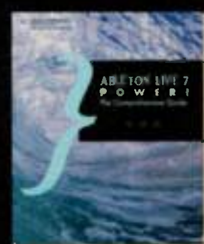

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»» Ocean Way Drums Gold gives you a library of huge-sounding multimiked drums with mix presets from Allen Sides and extensive user tweakability.

Drums (OWD), you would not be disappointed. OWD features numerous and varied multisampled kits, meticulously recorded by Sides and Miller using classic microphones in Ocean Way's vast Studio B. You get a library of drum sounds that are fat, punchy, and extremely realistic. These may be the best-sounding sampled drums I've ever heard.

OWD Gold ($895) includes a whopping 40 GB of sample data, which comes on six DVDs. Kontakt Player 2 (standalone and plug-in) is bundled with the library. (Also available is OWD Platinum [$1,795], which comes on its own hard drive with an astounding 120 GB of samples.) For both Mac and Windows users, a CPU with a minimum 2 GHz clock speed is recommended.

DRUM SOUND HEAVEN

The OWD Gold library consists of 19 kits, all of which offer samples derived from close, overhead, and room mics on each cymbal or drum (hi-hat, ride, and crash; kick; snare; and high, medium, and low toms). Each kit element features numerous Velocity layers for additional realism.

OWD's user interface lets you adjust the levels of the various mics used to record each element. On many of the snares, you also get a knob called RMX, which dials in a gated reverb sound. Many of the kicks have both RMX and Thwack controls, the latter being a heavily compressed signal. For instant gratification, there are six key-switchable mix presets for each kit, mixed by Sides himself (see **Web Clip 1**), which give you progressively more room sound as you step through them. These presets can also be adjusted for individual elements.

ONLINE BONUS MATERIAL

The kits range in sound from clean and punchy to big and fat. Each comes in two flavors: the C 12A version, for which the snare was miked with a stereo pair of AKG C 12As on top, and the 57 version, which has a mono Shure SM57 on the snare. (You can get 57s and C 12As on the snare in both kit types through the knobs in the OWD interface.) Both kit configurations offer stereo Sony 55P mics under the snare. Each kit is also available in either a Snares On or a Snares Off version, which refers to whether the snares were on (and thus rattling) or off when the other drums and cymbals were recorded. Overall, an immense amount of mixing control is available here.

> [These may be the best-sounding sampled drums I've ever heard.]

OFF THE MAP

All the kits are offered with keymapping for Roland V-Drums (featuring the TD-20 brain) and for Sonic Reality's proprietary I-MAP scheme. I-MAP is designed to make drum programming easier and more expressive from a keyboard (see **Web Clip 2**). Once you get used to it, it works extremely well. The version of OWD I reviewed did not offer GM keymaps, so it wasn't usable with preprogrammed MIDI drum sequences (or previously existing MIDI drum parts). However, the next update of OWD, which may be out by the time you read this, will add GM-mapped kits and a collection of MIDI files. Even before that comes out, a software patch that adds the GM-mapping feature will be available for download from the OWD site.

The PDF manual offers tips for using the library with Drumagog software (you need to download a free helper program before doing so). The idea of using OWD's sounds for drum replacement is tantalizing indeed.

DETAILS, DETAILS

I do have a few minor issues with OWD. There are no brush or Blastick samples, only stick hits. Depending on the musical style you're recording, that could be a limiting factor, although for straight-ahead rock and pop and a lot of contemporary country, these drum sounds are spot-on.

Another quibble is that the kits don't have descriptive names, only numerical ones (Kit 1, Kit 2). That fact made it difficult to recall what each kit sounded like; when auditioning the kits at random, it was tough to remember which ones I'd already listened to. Miller told me that he and Sides decided against descriptive names (especially brand names of drums in the kit) because they didn't want users to have false preconceptions of what the kits (which are all custom setups) sound like. Finally, I would have liked a printed manual, not just a PDF.

Overall, OWD is an amazing product. Yes, it's relatively expensive and requires a lot of disk space, but it offers world-class drum sounds and an incredible amount of mixing control.

Value (1 through 5): 5
Sonic Reality (distributor)
oceanwaydrums.com

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PRIMACOUSTIC

Recoil Stabilizer

By Michael Cooper

>> The Recoil Stabilizer is available in different sizes to accommodate many popular studio monitors.

In control rooms big and small, ergonomic necessities often impose damaging sonic penalties. We buy the best close-field monitors we can afford, but place them on top of shelves that dramatically degrade their performance.

Energy from a speaker causes the shelf it's mounted on to resonate, causing a boost in upper-bass and low-midrange frequencies and marring stereo imaging. In addition, as a monitor's woofer goes through its positive excursion (moving toward the listener), the speaker cabinet will tend to recoil slightly in the opposite direction. The backward recoil robs energy from all the drivers in the cabinet, rounding off transients and compromising stereo imaging and depth.

The solution to these problems is to acoustically decouple each monitor from the shelf—blocking energy transference to the shelf—while providing a solid anchor to prevent recoil. This is precisely what the Primacoustic Recoil Stabilizer ($84.99 to $279.99 each) does. Simply place each of your monitors on top of a Recoil Stabilizer to reap the benefits.

SUPER MODEL

Primacoustic currently offers ten different models of Recoil Stabilizers featuring various dimensions and load limits (from 32 to 144 pounds) to accommodate different monitors. (Visit primacoustic.com/recoil-home.htm for specs and to view a chart that recommends which model to use with various studio monitors.)

All models are composed of three layers of acoustic materials bonded together. A high-density, open-cell urethane foam (measuring 1.9 to 2.6 inches high, depending on the model) forms the bottom layer that comes in direct contact with the shelf. A laser-cut, ¼-inch-thick steel deck is fitted to the top of the foam. On top of the deck is an ⅛-inch-thick neoprene friction pad, which prevents a mounted speaker from slipping.

It's the heavy mass (provided primarily by the metal deck) and the antislip surface that distinguish the Recoil Stabilizer from more modestly priced decoupling products such as the Auralex MoPad, which is virtually an all-foam product. Generally speaking, the greater the mass, the more effective a decoupler will be. Heavy mass and an antislip surface also provide the firm anchor needed to thwart monitor recoil. Layering three acoustic materials—each having a different density and, therefore, different resonance characteristics—also helps prevent speaker cabinet resonances from transferring to the shelf the stabilizer sits on.

Most of the models can be ordered to have a top surface that either is flat or has a 5-degree downslope from back to front. The latter design is useful for angling your monitors slightly downward so they're aimed directly at your ears. Model RX5-UF features a 10-degree up-firing slope for use with small monitors placed on a tabletop. Two models are also available for use with midfield monitors and subwoofers, respectively.

MASS EFFECT

I reviewed the RX7a, a 10.3 × 13 × 2.6-inch model weighing 10.8 pounds and featuring a 5-degree downslope. I placed one RX7a under each of my two Yamaha NS-10M Studio monitors, which were situated on shelves of my Omnirax MixStation/02R.

I immediately recognized the improvement in the sound field my monitors produced when used with the Recoil Stabilizers. Bass reproduction was tighter and exhibited none of the upper-bass blurriness that typically plagues shelf-mounted monitors. Localization in the stereo field of different elements of the mix was much more precise. Detail and depth improved significantly. Surprisingly, hard-panned electric guitars also sounded more present. Taken together with improvements in the bottom end, however, that was a plus and did not compromise spectral balance. In comparison tests, the Recoil Stabilizers handily outperformed my Auralex MoPads in all ways except price.

Make sure that you choose a Recoil Stabilizer model whose dimensions match the real-world footprint of your monitors as closely as possible. This will vary depending on whether you align the monitors vertically or horizontally. For example, the RX7a was too narrow in width for optimal use with a horizontally aligned NS-10M, but Primacoustic offers the RX9, which is designed for using the monitor in a horizontal position. To get the least amount of resonance effects, the entire cabinet's base must be damped by the Recoil Stabilizer's mass.

ON SOLID GROUND

Adding Recoil Stabilizers to a setup using high-end monitors is a no-brainer; the expense will be only a fraction of what the monitors cost. The relative cost becomes more of an issue, however, when using them with budget-priced close-fields. But whatever the monitor's capabilities are, the Recoil Stabilizer will let you hear its full potential. This is by far the best-performing product of its kind I've reviewed to date. em

Value (1 through 5): 4
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The MOTU Desktop Studio

Powerful. Stylish. Compatible. Expandable.

CueMix FX

Apple 3GHz iMac

Beauty. Brains. And now more Brawn.

Setting the highest standards for elegance and simplicity, the all-in-one iMac now includes an Intel Core 2 Duo processor running at speeds up to 3.06 GHz, offering dazzling performance for **DP6**, **MachFive 2**, and all your 3rd-party instruments and plug-in processing.

MOTU UltraLite-mk3

Compact FireWire I/O with effects & mixing

The UltraLite-mk3 isn't just a 10 x 14 FireWire interface. It's a full-blown digital mixer with effects, including modeled analog EQ and compression on every channel, plus reverb — all accessed via the elegant **CueMix FX** on-screen mixer.

Waves Jack Joseph Puig Collection

Add his vintage sound to your MOTU rig

When JJP requires the perfect sound, he turns to three studio legends: the Fairchild 670, Pultec EQP-1A, & Pultec MEQ-5. Now, Waves has precision-modeled these rare pieces in JJP's studio to capture every detail of their rich harmonic complexity.

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Ocean Way Drums from Sonic Reality
The premiere virtual drum instrument

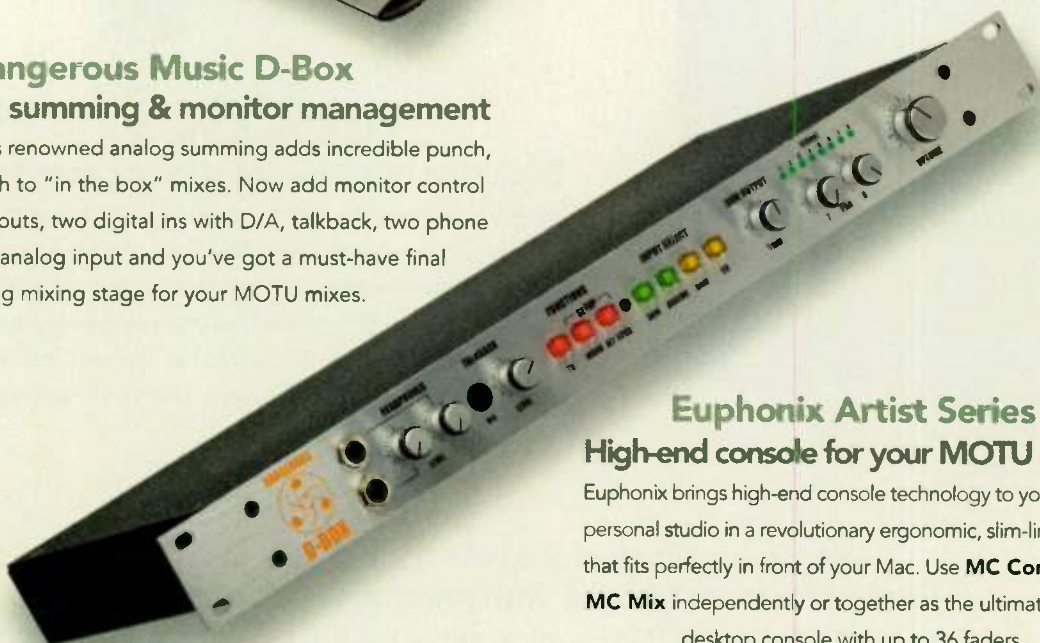
Put the power of the world's most awarded studio complex in your MOTU desktop studio. Ocean Way Drums delivers 19 drum kits immaculately recorded in legendary Ocean Way Studio B where artists like Radiohead, Green Day, Eric Clapton, Kanye West and Paul McCartney create hit records.

NI KOMPLETE 5 and KORE 2
Legendary virtual instruments with hands-on control

For Digital Performer users who want it all: 11 legendary instruments including KONTAKT 3 and the award-winning MASSIVE, combined with instant hands-on control. Choose from 7,500 presets in seconds and instantly tweak with real knobs.

Dangerous Music D-Box
Killer analog summing & monitor management

Dangerous Music's renowned analog summing adds incredible punch, depth and warmth to "in the box" mixes. Now add monitor control with two speaker outs, two digital ins with D/A, talkback, two phone outs and aux analog input and you've got a must-have final analog mixing stage for your MOTU mixes.



Euphonix Artist Series
High-end console for your MOTU studio

Euphonix brings high-end console technology to your MOTU personal studio in a revolutionary ergonomic, slim-line design that fits perfectly in front of your Mac. Use **MC Control** and **MC Mix** independently or together as the ultimate unified desktop console with up to 36 faders.

MC Mix

MC Control

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Monster Pro PowerCenters

Power conditioning and surge protection

Protect your MOTU studio investment and achieve optimal sound quality with the Monster Pro 2500 or Pro 3500 PowerCenters. Exceptional power conditioning, combined with Monster's patented Clean Power™ filter circuitry, provides separate noise isolation filtered outlets for digital, analog and high-current audio components for the cleanest sound possible.

RØDE NT2-A

Professional large capsule studio mic

Born from the legendary RØDE NT2, the NT2-A incorporates three pick-up patterns, three-position PAD and a three-stage high-pass filter conveniently located on the microphone body. The transducer has been voiced to complement today's modern recording techniques for the flexibility and superlative audio characteristics that make the NT2-A one of the most versatile condenser mics available. Backed by RØDE's industry-leading ten-year warranty.

BIAS Peak Pro 6

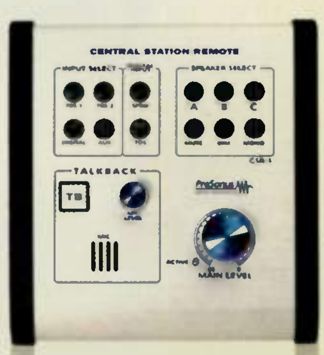
Evolution of an award-winning standard

Whether you're a musician, sound designer, audio editor, multimedia producer, or mastering engineer, Peak Pro 6 offers more creative potential than ever before. Used side-by-side or launched directly from within DP6, Peak Pro 6 streamlines your workflow with industry-renowned sonic quality and precision. For additional mastering, restoration and DDP 2.0 delivery power, step up to Peak Pro XT 6.

PreSonus Central Station

Control room monitoring with remote

The missing link between your MOTU recording interface, studio monitors, input sources and the artist. Monitor from among five sets of stereo inputs (three analog and two digital) and manage your sessions with hands-on control room features like talkback and listenback.



The MOTU Desktop Studio

Powerful. Stylish. Compatible. Expandable.

Mackie Control Universal Pro

The most complete control surface for Digital Performer

Nine motorized, touch-sensitive faders effortlessly control bankable channels while eight assignable V-Pots and over 50 master buttons provide unparalleled DAW control. A massive transport section with weighted jog wheel and robust build quality offers a true console feel. The 8-channel Expander Pro and C4 Pro virtual instrument controller allow seamless expansion.

Mackie HR824mk2 Active Studio Monitors

Premium performance to perfect your mix

A mainstay of professional studios worldwide, the HR824mk2 high-resolution monitors employ the new Zero Edge Baffle, which minimizes diffraction for a crystal clear image and an evenly dispersed ultra-wide sweet spot. Acoustic Space, LF roll-off and HF controls allow custom configurations, which are sure to suit your MOTU studio space… and your taste.

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Why Can't I Stop Listening to This?

By Nathaniel Kunkel

I have just spent a couple of weeks mixing a show I recorded last year of Guy Clark, Joe Ely, John Hiatt, and Lyle Lovett. They do the Songwriters Tour together once in a while, and I was lucky enough to be asked to record their Redwood City performances in Northern California. It's really an amazing show: they are all onstage at the same time, each one performs a song, and then he passes the baton to the next guy. One of the benefits of this arrangement is that there is no setlist. They play what they feel like playing after the previous guy has played. They might play a solo on another performer's song, and they usually talk to each other about their respective performances and/or what the song was about. It is amazing to watch and very educational.

After I mixed the show, however, I learned some amazing stuff.

Lyle, who spearheads the project, is one of the greatest champions of integrity I have ever met. So when I was mixing the show, I decided to leave in all the talking, all the tuning, all the audience chatter, everything. It was long, but it was the performance in its entirety. I figured that approach would be the most beneficial for the performers to evaluate the show before release.

The other thing that differed in my normal approach to mixing on this project was my abandonment of an elevated-level master. I am so over it (no pun intended). This product was going to have dynamic range.

So I printed my mixes of real performances with dynamic range (which should not be the anomaly that it is). Then I burned a couple of discs to reference in my car. However, I was not prepared for my reaction: I was fully engaged and unable to pull myself away from listening to this concert. That was unusual—I normally want to get as far away from a project as possible after listening to it for weeks. But this was different. There was flow. There were dynamics. There was humanity. There were jokes and discussions. There was, right there, something I don't hear much anymore: people entertaining me. Not machines. Not reverb. Not Auto-Tune. *People.*

It was also a complete listening experience—the dialog was as important as the songs. These are not just amazing songwriters; these are amazing people, with a story and a reason for everything they do. And they were sharing it with me.

I was floored. It was captivating, and my desire to keep listening over and over was unrelenting. Like a man who has come out of the desert and found water, I could not get enough.

So that got me thinking about some of the music by musicians I love—Buddy Holly, Arlo Guthrie, James Taylor, Led Zeppelin, David Bowie, Eric Taylor, Karla Bonoff, Sting, Stevie Ray Vaughn, Nat King Cole, and Frank Zappa. You know, the good stuff. The stuff that could, and still does, enthrall me.

I also thought about the first time I recorded James Taylor. I was so worried. Would I be able to record his voice right? It's one of the most recognizable voices in popular music. Then I heard him sing right in front of me, and my worries evaporated. Not because I was unnecessarily confident or thought I had the right mic. My worries evaporated because you could put a $49 Realistic mic from Radio Shack in front of him and he would still sound like James Taylor. You would have to try to bung it up. Really, you would.

Why am I telling you this?

Because like this show with Guy, Joe, John, and Lyle, and like my experience with James, when you have a truly great performance of a truly great song, sometimes you just need to get out of the way.

If you are in front of an artist and they don't floor you, I'm not sure adding a loop is going to make them a star. We should stop trying to make hit records for any artist and start making good records for hit artists. Not only is it easier, but if I were to judge an artwork's success by the enjoyment of the person experiencing it over time, I think it's more likely that a good record from a talented artist will result in great art. Only time will tell. em

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Nathaniel Kunkel (studiowithoutwalls.com) is a Grammy and Emmy Award–winning producer, engineer, and mixer who has worked with Sting, B.B. King, Insane Clown Posse, I-Nine, and comedian Robin Williams.


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